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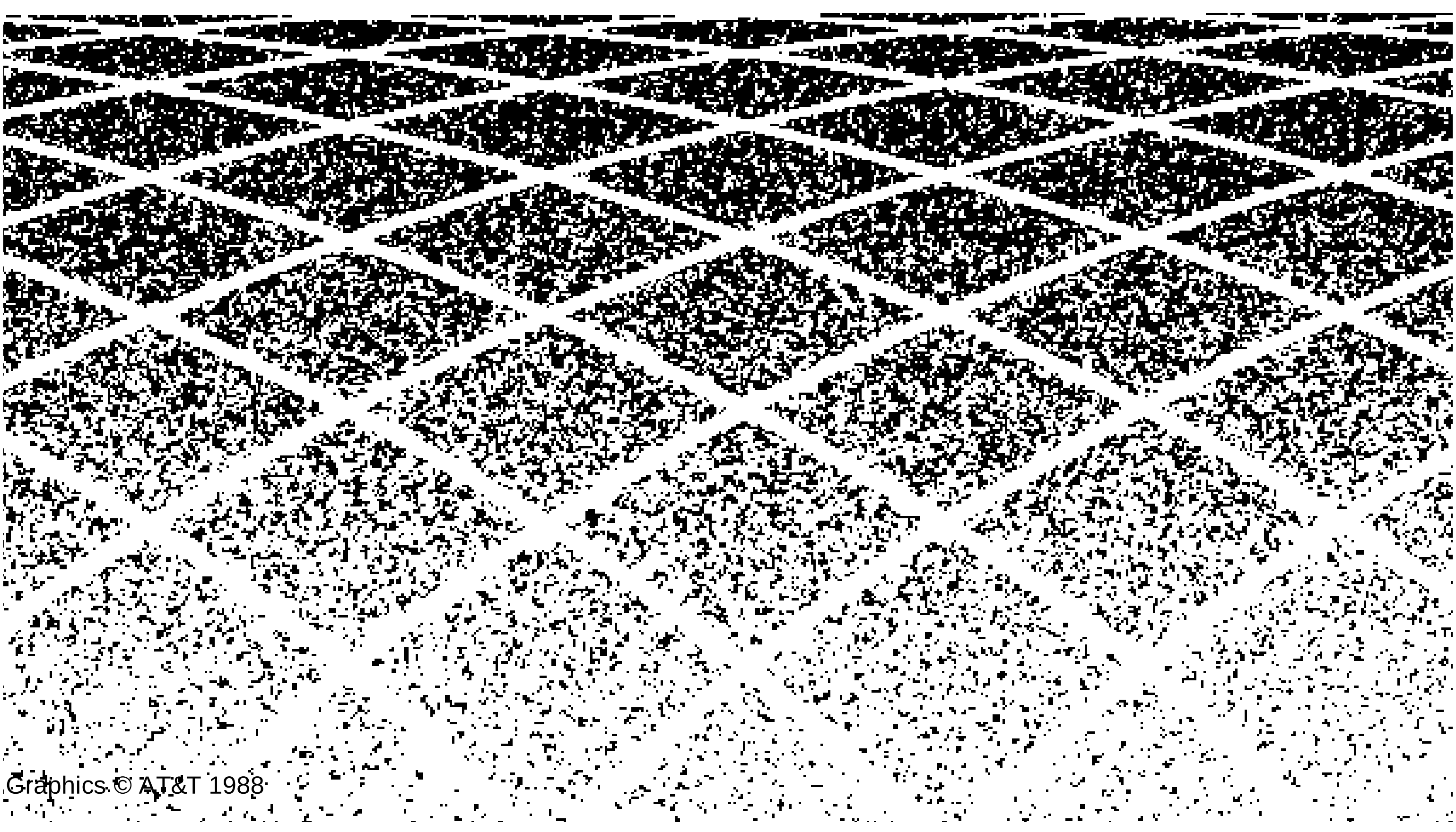
DEFINITY Communications System Generic 3 Feature Description



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About This Document

This document describes DEFINITY® Communications System Generic 3 features. Differences between the various Generic 3 (G3) releases, are specifically identified throughout this document.

Purpose

This document helps switch administrators and managers select communication features to include in their initial system or to add after the system is in service. It provides feature descriptions, administration and hardware information, and other considerations applicable for each available feature.

Along with the *DEFINITY Communications System Generic 3 System Description and Specifications*, 555-230-206, this document provides an overall reference for the planning, operation, and administration of your DEFINITY switch. It is also a tool for answering questions about the interactions between specific features. Feature capacities and limitations are described in the System Capacity Limits table in Appendix A.

This document is used in conjunction with the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, for software initialization and subsequent changes in feature assignments for systems G3sV1, 555-230-650 and 555-230-650ADD; G3iV1, 555-230-650; G3rV1, 555-230-651; G3i-Global, 555-230-652; G3V2, 555-230-653; G3V3, 555-230-653ADD1; G3V4, 653ADD1.

Intended Audiences

This document is intended for the DEFINITY switch administrators and managers in your company, end-users interested in information about specific features, and AT&T support personnel responsible for planning, designing, configuring, and selling the system.

Reason For Reissue

This document is updated to include Generic 3 Version 4 information.

How to Use This Document

This document is designed to be used as a reference document. If you are interested in information about a particular feature, use the table of contents or index to locate the page number where the feature is described.

Organization

This document is organized as follows:

- **Chapter 1, "Introduction"** provides an introduction to this document and provides a general description of the functions and services provided with the system. Chapter 1 also recommends various security measures to protect your system from unauthorized use.
- **Chapter 2, "Functional Description"** provides descriptions of the DEFINITY switch features, functions, and services. This chapter also presents these capabilities classified into six categories: Voice Management, Data Management, Network Services (including World Class Routing), System Management, Hospitality Services, and Call Center Services. Each group of functions and services, including a listing of associated features, is described separately in a section of this chapter.

  See Chapter 3, "Feature Descriptions" for detailed information on individual features.
- **Chapter 3, "Feature Descriptions"** provides detailed information of the system features. The features are arranged in alphabetical order, regardless of the functional area to which they apply. The information for each feature is presented under six headings: Feature Availability, Description, Considerations, Interactions, Administration, and Hardware and Requirements.
- **Appendix A, "System Parameters"** provides information relating to overall system characteristics and capacities. This chapter includes items that must be considered when planning for system implementation.

- **Appendix B, "References"** provides a list and brief descriptions of reference documents.
- **Appendix C, "Generic 3 V3 to Generic 3 V4 Transition Reference"** provides a list indicating new and enhanced features for G3V4 and briefly describes G3V4 feature enhancements.
- **Abbreviations** provides a list of abbreviations for the entire document.
- **Glossary** provides a glossary for the entire document.
- **Index** provides an index for the entire document.

Security Requirements

Chapter 1, "Introduction" contains a section describing security practices you should follow and list the features requiring special measures to secure them from unauthorized use. (You can also consult the index under “Security Measures.’’)

The feature descriptions for these features have an extra section, entitled “Security Measures,’’ describing the security steps you should perform.

Conventions Used in This Document

This document uses the following conventions:

- *Italic* typeface is used to emphasize key words in text.
- Command names, file names, and parameters are shown in **Helvetica Bold** typeface.
- Variables and information you type are shown in ***Helvetica Bold Italic***.
- System messages, screens, and responses are shown using `Constant Width` typeface.
- Variables are shown in *Helvetica Light Oblique* typeface.
- Keyboard keys are shown inside an oval (for example, RETURN).
- Telephone dial-pad buttons are shown inside square boxes (for example, 2 or *) and may refer to either rotary or touch-tone telephones.

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Introduction

1

Overview

This document describes the DEFINITY Generic 3 family of cost-effective digital communication systems. These systems:

- Route voice and data information between various endpoints (telephones, terminals, computers, etc.)
- Provide highly robust networking capabilities
- Include an extensive set of standard features (for example, Attendant Consoles, Voice Processing Interface, Call Coverage, DS1/E1 Connectivity, Hospitality Support, Recorded Announcement, and Trunk-to-Trunk Transfer)
- Provide flexibility and allow for the addition of optional features and/or upgrades to the system as business needs change

The term ‘‘DEFINITY Generic 3’’ refers to four distinct hardware platforms:

- DEFINITY Generic 3vs [A wall mounted unit] (G3vs)
- DEFINITY Generic 3s [A small floor or table unit] (G3s)
- DEFINITY Generic 3i [A medium size i386-based switch available in several configurations] (G3i)
- DEFINITY Generic 3r [A large MIPS-based switch also available in multiple configurations] (G3r)

These switches meet and exceed the needs of a wide variety of businesses. For example, G3s is a cost-effective switch for businesses with less than 200 stations and 100 trunks. G3i, on the other hand, meets the requirements of larger businesses that require more than 200 stations and 100 trunks. G3r, the largest switch, provides even greater capacities.

The current release of Generic 3 introduces Version 4 of the G3 switches. These phone switches are now more powerful and full-featured than ever.

This document describes versions of Generic 3. This chapter briefly describes the differences between the various Generic 3 switches and is organized into the following sections:

- **Generic 3 Version 1**

  This section describes the Generic 3 Version 1 offering; it compares the G3i, G3i-Global, and G3r Version 1 switches to the earlier G1 switch; and it compares the G3s and G3vs Version 1 switch to the G3i Version 1 switch.

- **Generic 3 Version 2**

  This section describes the Generic 3 enhancements introduced in Version 2.

- **Generic 3 Version 3**

  This section describes the Generic 3 enhancements introduced in Version 3.

- **Generic 3 Version 4**

  This section describes the Generic 3 enhancements introduced in Version 4.

- **Generic 3 Notation**

  This section explains notation used throughout this document to differentiate the various switches, and the different versions of the same switch.

For a list of the features available with each switch, see Table 1-6 in this chapter; for a summary of the system capacities for each switch, see Appendix A, "System Parameters".

Generic 3 Version 1

Generic 3 Version 1 includes the following switches:

- Generic 3i Version 1
- Generic 3i-Global (Contains features for international requirements)
- Generic 3r Version 1
- Generic 3s Version 1
- Generic 3vs Version 1

Each switch is described next.

G3i Version 1

G3i Version 1 is an enhanced version of earlier DEFINITY switches and combines many of the features of Generic 1 and Generic 2. G3i Version 1 addresses the needs of businesses in the United States and supports up to 1600 stations and 400 trunks.

G3i Version 1 introduced the following features and functions that are not part of G1 or G2:

- Access Endpoints (See the Administered Connections Feature)
- Administered Connections
- Administration Without Hardware
- Alphanumeric Dialing
- Attendant Serial Calling
- Attendant Intrusion (Call Offer)
- Attendant Override of Diversion Features
- Attendant Priority Queue
- Call Pickup Alerting
- Call Prompting
- Call Vectoring
- CallVisor™ ASAI (Adjunct/Switch Application Interface)
- DCS Over ISDN-PRI D-Channel
- Default Dialing
- Non-Facility Associated Signaling and D-Channel Backup
- Inbound Call Management
- Integrated Services Digital Network — Basic Rate Interface (ISDN-BRI)

- Look Ahead Interflow
- Security Violation Notification

Additional G1 features are significantly enhanced for G3i Version 1. These features include:

- ARS and AAR also known as “Ten-to-Seven Digit Conversion’’ are enhanced to support Multi-National Call Routing and Multi-National Toll Analysis
- Forced Entry of Account Codes, now known as CDR Account Code Dialing
- Agent Call Handling — Stroke Counts, Call Work Codes, and Forced Entry of Stroke Counts and Call Work Codes are added
- Automatic Call Distribution — Direct Agent Calling is added
- Call Detail Recording — New CDR record formats are added
- Class of Restriction — New restrictions are added
- Integrated Services Digital Network — Primary Rate Interface (ISDN-PRI) — Software Defined Data Network (SDDN) was added
- Toll Restriction

G3i-Global Version 1

G3i Version 1 is an enhanced version of earlier DEFINITY switches and combines many of the features of Generic 1 and Generic 2. G3i-Global Version 1 addresses business needs in the global market and supports up to 2400 stations and 400 trunks.

The number of differences between G1 and G3i-Global is predicated on whether you had an international version of G1 or the United States (US) version. Some international features were built into the G1 switches produced for locations outside this country.

G3r Version 1

G3r Version 1 is an enhanced version of earlier DEFINITY switches and combines many of the features of Generic 1 and Generic 2. It is the DEFINITY switch required for large applications or large offices. G3r Version 1 addresses the needs of businesses in the United States and supports up to 10,000 stations and 4,000 trunks.

G3r Version 1 introduced the following features and functions that are not part of G1. It also contains most of the G3iV1 features. New features are shown below:

- Alternate Facilities Restriction Levels
- Attendant Serial Calling

- Attendant Intrusion (Call Offer)
- Attendant Override of Diversion Features
- Attendant Priority Queue
- ACD — Auto-Available Split
- Audible Message Waiting
- Auto-Start/Don’t Split
- Automatic Transmission Measurement System
- DCS Over ISDN-PRI D-Channel
- Extension Number Portability
- Extended Trunk Access
- Hold — Automatic
- Malicious Call Trace
- Multiple User Access — Maintenance and Administration
- Restriction — Fully Restricted Service
- Terminal Translation Initialization
- Transfer — Outgoing Trunk to Outgoing Trunk

Other G1 features are significantly enhanced for G3r Version 1. These features include:

- Administration Without Hardware
- Attendant Dial Access
- Attendant Timers and Timed Reminders
- Automatic Alternate Routing (AAR)
- Automatic Route Selection (ARS)
- AUDIX
- Call Detail Recording (CDR) [Formerly called SMDR]
- CallVisor ASAI (Adjunct/Switch Application Interface)
- Class of Restrictions (COR)
- Conference — Attendant
- Conference — Voice Terminal
- Dial Plan
- Direct Inward Dialing (DID)
- Distinctive Ringing
- DS1 Tie Trunk Service

- Emergency Access to Attendant
- Facilities Test Calls
- Leave Word Calling
- Line Lockout
- Trunk Group Hunting

G3s Version 1

DEFINITY Generic 3s Version 1 is designed to provide businesses with up to 200 stations with greater flexibility in choosing their telecommunication systems. Not only does G3s carry the same strength of architecture, price performance, and investment protection provided by Generic 3i and Generic 3r, DEFINITY Generic 3s also includes many of the Generic 3i features and applications in packages that serve a variety of business needs.

DEFINITY Generic 3s offers two distinct packages: the Premier Business Package (PBP) and the Advantage Business Package (ABP). Each package contains a subset of the DEFINITY Generic 3i features and options that meet specific business needs and provide flexible communication solutions.

The Premier Business Package is designed for customers with up to 200 stations and 100 trunks, while the Advantage Business Package is designed for customers with up to 200 stations and 50 trunks. The optional features for PBP are the same as for G3i; the optional features for ABP includes a subset of the G3i optional features and are available through various option packages. See Table 1-6 for further details.

G3vs Version 1

DEFINITY Generic 3vs is a new member of the Generic 3 family of cost-effective digital communications systems designed to: provide businesses with 80 or fewer stations greater flexibility in choosing their telecommunication systems. It also provides satellite affiliates of multi-location businesses with the same telecommunication services available at larger locations using the G3s, G3i, and G3r switches.

G3vs is a convenient compact single-carrier cabinet switch that is wall-mounted. It has a FLASH memory cartridge that supports all G3s features and options except, Integrated Services Digital Network — Basic Rate Interface (ISDN-BRI) and CallVisor Adjunct/Switch Application Interface (ASAI).

As stated for G3s, the Advantage Business Package and the Premier Business Package are available.

Generic 3 Version 2

Version 2 combines almost all features available in Version 1 software across all hardware platforms into one software base. This base is available on all platforms. For example, the international features that were only available for DEFINITY G3i-Global are now available for G3r, G3i, and G3s/G3vs. Generic 3 Version 2 includes the following switches:

- Generic 3i Version 2 (G3iV2)

  G3i Version 2 also runs on two distinct processor platforms: an i286 and i386. To distinguish between these products this document will sometimes specify G3iV1-286 and G3iV1-386 or G3iV2-286 and G3iV2-386. The major differences between these packages are that G3i-386 has larger capacity limits than the G3i-286 (see Appendix A, "System Parameters" for details).

- Generic 3r Version 2 (G3rV2)
- Generic 3s Version 2 (G3sV2)

  G3s Version 2 has two packages: the Premier Business Package (PBP) and the Advantage Business Package (ABP).

- Generic 3vs Version 2 (G3vsV2)

  G3vs Version 2 has two packages: the Premier Business Package (PBP) and the Advantage Business Package (ABP).

New Features for Version 2

The following new features are supported in Generic 3 Version 2:

- EAS (Expert Agent Selection)
- ECMA (European Computer Manufactures Association) ISDN-PRI Basic Call
- Japan NTT ISDN-PRI Basic Call
- Redirection on No Answer (RONA)
- Spain Telephone System Compatibility
- Voice Response Integration (VRI)
- Wideband Switching

Generic 3 Version 3

Generic 3 Version 3 includes the following switches:

- Generic 3i Version 3 (G3iV3)

  G3i Version 3 runs on the i386 processor platform.

- Generic 3r Version 3 (G3rV3)
- Generic 3s Version 3 (G3sV3)

  G3s Version 3 has two packages: the Premier Business Package (PBP) and the Advantage Business Package (ABP).
- Generic 3vs Version 3 (G3vsV3)

  G3vs Version 3 has two packages: the Premier Business Package (PBP) and the Advantage Business Package (ABP).

New Features for Version 3

The following new features are supported in Generic 3 Version 3:

- Add/Remove Skills
- Administrable Logins
- Agent Sizing
- Attendant Room Status
- Constellation Voice/Data Terminal Support
- Forced Password Aging
- Multiple Call Handling
- Switch Based Bulletin Board
- VDN of Origin Announcements
- VuStats

Many Generic 3 features were also enhanced for G3V3. See *DEFINITY Communications System Generic 3V2 to Generic 3V3 Transition Reference*, 555-230-621, for a complete description of all V2 to V3 additions and enhancements.

Generic 3 Version 4

Generic 3 Version 4 includes the following switches:

- Generic 3i Version 4 (G3iV4)

  G3i Version 4 runs on the i386 processor platform.
- Generic 3r Version 4 (G3rV4)
- Generic 3s Version 4 (G3sV4)

  G3s Version 4 has two packages: the Premier Business Package (PBP) and the Advantage Business Package (ABP).
- Generic 3vs Version 4 (G3vsV4)

G3vs Version 4 has two packages: the Premier Business Package (PBP) and the Advantage Business Package (ABP).

New Features for Version 4

The following new features are supported in Generic 3 Version 4:

- Advice of Charge (ISDN-PRI)
- Call Forward Busy/Don't Answer
- DCS Call Coverage
- Flexible Billing
- Misoperation Handling (modified)
- PC Application Software Translation Exchange (PASTE)
- Ringing — Abbreviated and Delayed
- Tenant Partitioning
- World Class Core BRI

Many Generic 3 features were also enhanced for G3V4. See Appendix C, "Generic 3 V3 to Generic 3 V4 Transition Reference" for a complete description of all V3 to V4 additions and enhancements.

Generic 3 Notation

The Generic 3 notation refers to ALL G3 releases. To differentiate between the Generic 3 (G3) switches as a group, and the different implementations of the Generic 3 switches, the notations listed below are used in this document

Table 1-1. Generic 3 Version 1 Switches

Abbreviation	Refers to...
G3sV1	Generic 3s Version 1
G3sV1 ABP	Generic 3s Version 1 Advantage Business Package
G3sV1 PBP	Generic 3s Version 1 Premier Business Package
G3vsV1	Generic 3vs Version 1
G3vsV1 ABP	Generic 3vs Version 1 Advantage Business Package
G3vsV1 PBP	Generic 3vs Version 1 Premier Business Package
G3iV1	Generic 3i Version 1
G3i-Global	International Generic 3i Version 1
G3rV1	Generic 3r Version 1
G3V1	G3sV1, G3vsV1, G3iV1, G3i-Global, and G3rV1

.

Table 1-2. Generic 3 Version 2 Switches

Abbreviation	Refers to...
G3sV2	Generic 3s Version 2
G3sV2 ABP	Generic 3s Version 2 Advantage Business Package
G3sV2 PBP	Generic 3s Version 2 Premier Business Package
G3vsV2	Generic 3vs Version 2
G3vsV2 ABP	Generic 3vs Version 2 Advantage Business Package
G3vsV2 PBP	Generic 3vs Version 2 Premier Business Package
G3iV2	Generic 3i Version 2
G3iV2-286	Generic 3i Version 2 on 286 processor
G3iV2-386	Generic 3i Version 2 on 386 processor
G3rV2	Generic 3r Version 2
G3V2	G3sV2, G3vsV2, G3iV2, and G3rV2

Table 1-3. Generic 3 Version 3 Switches

Abbreviation	Refers to...
G3sV3	Generic 3s Version 3
G3sV3 ABP	Generic 3s Version 3 Advantage Business Package
G3sV3 PBP	Generic 3s Version 3 Premier Business Package
G3vsV3	Generic 3vs Version 3
G3vsV3 ABP	Generic 3vs Version 3 Advantage Business Package
G3vsV3 PBP	Generic 3vs Version 3 Premier Business Package
G3iV3	Generic 3i Version 3
G3rV3	Generic 3r Version 3
G3V3	G3sV3, G3vsV3, G3iV3 and G3rV3

Table 1-4. Generic 3 Version 4 Switches

Abbreviation	Refers to...
G3sV4	Generic 3s Version 4
G3sV4 ABP	Generic 3s Version 4 Advantage Business Package
G3sV4 PBP	Generic 3s Version 4 Premier Business Package
G3vsV4	Generic 3vs Version 4
G3vsV4 ABP	Generic 3vs Version 4 Advantage Business Package
G3vsV4 PBP	Generic 3vs Version 4 Premier Business Package
G3iV4	Generic 3i Version 4
G3rV4	Generic 3r Version 4
G3V4	G3sV4, G3vsV4, G3iV4 and G3rV4

The following table identifies switches independent of their versions.

Table 1-5. Generic 3 Switches

Abbreviation	Refers to...
G3s	G3sV1, G3sV2, G3sV3, G3sV4
G3vs	G3vsV1, G3vsV2, G3vsV3, G3vsV4
G3i	G3iV1, G3iV2, G3iV3, G3iV4
G3i-Global	G3V1-Global
G3r	G3rV1, G3rV2, G3rV3, G3rV4
G3	G3V1, G3V2, G3V3, G3V4

Security Measures

Security on the Remote Administration port cannot be overemphasized. If it is not secure, an unauthorized user can compromise the system in many ways, such as viewing Barrier and Authorization codes or viewing Feature Access Codes.

WARNING:
If you suspect that unauthorized access has occurred, the system administrator should change the barrier codes, authorization codes, passwords, and any other sensitive codes.

To help secure the system, the system administrator should follow the steps below:

- Change the factory default passwords after the system is powered up if you have a DEFINITY release prior to G3V3. With G3V3 and higher, the switch is equipped with init, inads, and craft passwords. These are the only default logins and are AT&T services logins. The first superuser customer login must be created during the installation process.
- If you are upgrading to V3 or V4, your logins and passwords will be preserved as part of the upgrade process. In this case, change logins and passwords following the upgrade.
- Change passwords quarterly.
- Administer the Security Violation Notification feature to report unsuccessful attempts to access the system. With G3V3 and later releases, the Security Violation Notification Feature has the added capability of disabling a valid login ID following a security violation involving that login ID and disabling remote access following a security violation involving a barrier code or authorization code.
- Use the Recent Change History feature to determine if unauthorized changes have been made to the system. To assist in identifying unauthorized use of the system, in G3V4, the Recent Change History Report lists each time a user logs in or off the system.

  NOTE:
  With the exception of log ins and log offs with G3V4 and later releases, the Recent Change History feature does not record and display administration commands that only display information.

Please pay special attention to the features and security measures discussed in the following sub-sections.

For assistance with toll fraud prevention, call either AT&T Corporate Security at 800 821-8235 or the DEFINITY Helpline at 800 225-7585.

If you have identified fraudulent calling in progress, and require assistance in stopping the fraud, call the AT&T Technical Service Center at 800 242-2121.

When you have dialed the service center number from a touch-tone phone, you will make selections on several menus to access toll fraud help. Follow the verbal instructions provided by the menu topics.

Logoff Notification

G3V4 and later releases provide for notification at logoff when the Remote Access or Facility Test Calls features remain active. Logoff Notification is administered on the Login Administration form and can be assigned to any login ID. The notification can be administered to force acknowledgment from the user.

Logoff Notification is primarily used to notify the system administrator when one of these security risk features is still active at logoff. Notification guards against inadvertently leaving a security risk feature active. It can also alert the system administrator to unauthorized feature activation. See the *GBCS Products Security Handbook*, 555-025-600, for more information.

Passwords

1. Change all default passwords in both the switch and adjuncts (CMS, CSM, AUDIX, Trouble Tracker, VMAAP, Manager II/III, Conversant, Intuity, and others)
2. The following customer logins are included in the G1 and G3 (pre-V3) product: cust, rcust, browse, bcms, and nms. The "cust" login is for superuser access. The remaining logins are non-superuser access. G3V3 and later releases have administrable logins.
3. AT&T will change the password quarterly on its login IDs.
4. Use the maximum number of digits, mixing both alpha and numeric
5. Change passwords at least quarterly

G3V3 and later releases enhanced switch security by adding Forced Password Aging and Administrable Logins. This security feature enables the user to define their own login IDs and specify a set of commands that each login has access to. In addition, the user is required to change the password for each login at specified time intervals when Forced Password Aging is administered.

Physical Security

1. Keep wiring closets and switch rooms secure
2. Keep any documentation pertaining to switch operation secure
3. Keep any reports that may reveal trunk access code, remote access barrier codes, authorization codes, or password information secure

Remote Administration Port

Security on the Initialization and Administration System (INADS) Remote Administration Port is critical. The optional Remote Port Security Device (RPSD) is a password encryption device that can be attached to the Remote Administration Port for increased security. It is virtually impenetrable and provides maximum protection.

To increase system security, G3V4 by default disables customer access to the INADS Remote Administration Port. For a customer to have access, AT&T must enable customer super-user login access. Once access is enabled, the customer must enable port access as needed per user login.

Trunks

1. If ARS is provided, do not allow Dial Access by Trunk Access Codes
2. Remove the Facility Test Call feature code (Feature Access Codes)
3. On the COR form disable the Facility Access Trunk Test (N)
4. If provided, administer a trk-ac-alm on the console to alert whenever the Facility Test Call feature is enabled
5. Disallow Trunk-To-Trunk Transfer if not needed
6. Disallow or limit outgoing calls from tie lines by assigning appropriate FRL's
7. Require Forced Entry of Account Codes or Authorization Codes on toll calls
8. Block International calls if no business is conducted overseas or use the ARS tables to limit calling to specific country codes or North American area codes.
9. Block area codes if calls to certain states are not permitted

AUDIX

1. Limit voice mail to internal calling only by Outward Restricting the COR of the voice ports and assigning the lowest possible FRL to the COR
2. Allow outbound calls to only certain numbers as defined by the Unrestricted Call List
3. Activate Enhanced Call Transfer if available in the AUDIX, DEFINITY AUDIX, or Intuity and if supported by the switch
4. Restrict the transfer capability of AUDIX from accessing trunks by assigning a COR that prohibits access.

For details on how to administer a DEFINITY AUDIX installation, refer to *Switch Administration for DEFINITY AUDIX System R1.0*, 555-300-509.

Automated Attendant

1. Restrict menu options to internal extensions only, by Outward Restricting the automated attendant ports
2. Restrict calls that transfer off-premises to specific numbers as defined in the Restricted Call List
3. Force disconnect or route to an attendant any menu option that is not valid

NOTE:
The above values may differ in different switches. Always try to use the maximum value for the feature as defined by the switch documentation. This list is to provide ideas in preventing fraud and does not list all possible ways fraud may occur. AT&T does not guarantee that the list above will stop all fraud.

Call Forwarding

Use the **list call-forwarding** command, available with G3V4 and later releases, to identify unauthorized Call Forwarding feature activation. The command can be used to list: all stations with Call Forwarding Active; a subset of stations with Call Forwarding active; or the Call Forwarding status of a specified station.

The list identifies stations with either Call Forwarding All Calls (on-net or off-net) or Call Forwarding Busy/Don't Answer active. It shows the station extension, the station name and the forwarded-to destination.

Call Vectoring

Call vectoring allows processing of incoming and internal calls according to a programmed set of commands. Vector commands can direct calls to on-premises or off-premises destinations, to a hunt group or split, or to a specific call treatment such as an announcement, forced disconnect or delay treatment. It is possible for the system to collect digits from the user and route calls to a destination specified by those digits, and/or do conditional processing according to the digits dialed (Call Prompting feature).

Calls access vectors using Vector Directory Numbers (VDNs). A VDN is a ‘‘soft’’ switch extension not assigned to a physical equipment location but having many of the properties of a normal extension number, including a Class Of Restriction (COR). The VDN, when dialed (or inferred), routes calls to the vector. Calls processed by the vector carry the permissions and restrictions associated with the COR of the Vector Directory Number.

Putting this all together, if a vector in the switch is written to collect digits, and then to route to the digits dialed, the restrictions on what calls can be placed are determined by the COR of the VDN. An incoming caller can access Trunk Access Codes, some Feature Access Codes, or most other sets of dialed digits.

In order to deny incoming callers access to outgoing facility paths, the COR of the Vector Directory Number must be configured to disallow outgoing access. This should include; lowering the Facility Restriction Level in the COR to the lowest acceptable value (FRL=0 provides the most restricted access to network routing preferences), assigning a Calling Party Restriction of ‘‘Toll’’ or ‘‘Outward’’, denying Facility Test Call capability, and blocking access to specific COR's assigned to outgoing Trunk Groups using the Calling Permissions section of the Class Of Restriction Screen.

Action

Review the Classes of Restriction assigned to your VDNs. If they are not restricted, consider assigning restrictions on the VDN to prevent callers exiting the system via the vector. For assistance, please contact the DEFINITY Helpline at 800 225-7585.

Enhanced Call Transfer

When using AUDIX, DEFINITY AUDIX, or Intuity, remember to activate the Enhanced Call Transfer (ECT) feature as part of your security plan. While this powerful feature limits transfers to valid extensions, there are some extension numbers that should NOT be made available to AUDIX transfers.

ECT allows callers to transfer out of AUDIX to valid extension numbers (as determined by the switch’s Dial Plan.) However, there are certain extension numbers, within the Dial Plan, that provide capabilities AUDIX transfers should be denied access to. Examples would include extension numbers used to dial-access system administration capabilities within the switch or extension numbers that are associated with features that provide second dial tone (Remote Access Extension).

To block AUDIX access to these extension numbers, they should be placed in special Class(es) of Restriction (COR). Using COR-To-COR calling permissions, the COR of the AUDIX analog ports should be denied access to the CORs containing these special extension numbers. To do this, access the Change - Class Of Restriction screen for the COR assigned to the analog ports that connect to AUDIX.

NOTE:
AUDIX ports should always reside in a separate COR designed specifically for them. In the COR-To-COR permissions area at the bottom of the screen, enter ‘‘no’’ in each COR field corresponding to the CORs assigned to the extensions you want protected.

Action

Review the Classes of Restriction assigned to your AUDIX analog ports and your Remote Access/Netcon (G3vs, G3s, G3i) or system port (G3r) data extensions. If they are not restricted, consider assigning restrictions that would prevent callers in AUDIX from being transferred to these extensions. For assistance, please contact the DEFINITY Helpline at 800 225-7585.

Remote Access

1. Use maximum Barrier Code Length (7)

   G3V3 and later releases provide a 7-digit barrier code with Remote Access Barrier Code Aging. Remote Access Barrier Code Aging limits the length of time a barrier code remains valid, and/or the number of times a barrier code can be used. The ability to define the life span and number of times a barrier code can be used reduces the opportunity for unauthorized use of the Remote Access feature.
2. Activate Authorization Code Required (Y)
3. Assign each barrier code a COR and COS that allow only necessary calls
4. Assign an appropriate Facility Restriction Level (FRL) and other restrictions for each COR (outward, toll, etc.)
5. Use maximum Authorization Code Length (7)
6. If providing attendant coverage, activate Timeout To Attendant (Y)
7. Change or remove authorization codes when authorized users leave the company
8. If Time of Day Routing is provided, raise the FRLs on route patterns during hours that remote access should not be used.
9. Suppress dial tone in the Remote Access Dial Tone field (Y)
10. If Remote Access is not going to be used, permanently disable the Remote Access feature. This feature is available on RV3, G1, and G3 V1.1 and higher.

Use the **status remote-access** command in G3V4 to check the status of the remote access feature and barrier codes. The command displays information that can help in determining why and when use of the remote access feature or a particular barrier code was denied.

If Remote Access is not permanently disabled, but is not administered, set Logoff Notification, available with G3V4 and later releases, to notify the system administrator at logoff when the Remote Access feature is enabled. Logoff Notification is administrable on a login ID basis.

For a detailed description of the status remote-access command and Logoff Notification, see the *GBCS Products Security Handbook*, 555-025-600.

Other Features Requiring Security Precautions

Follow the specific security measures recommended in this chapter when administering the following features:

— AUDIX Interface

— Call Vectoring

— Facility Test Calls

— Remote Access

— Remote Administration

— Transfer - Trunk to Trunk

— Terminal Translation Initialization

(See the index under ‘‘Security Measures’’ for the specific page numbers where the security measures are described for each of these features.)

Consult the *GBCS Products Security Handbook*, 555-025-600, for additional steps to secure your system and to find out how to regularly obtain information concerning security developments.

Organization of Features

The next few chapters describe the DEFINITY switch features, functions, and services. Chapter 2 presents these capabilities classified into six categories:

- Voice Management Overview
- Data Management
- Network Services (including World Class Routing)
- System Management
- Hospitality Services
- Call Center Services

NOTE:
It is useful to consider the features and functions from the perspective of the following functional views: System View, Call Center View, PBX-to-Host View, Voice Processing View, Network View (including World Class Routing), System Management View, Desktop View, Hospitality View, and Support View. Some features and functions support several of these views, so the views should not be seen as discrete categories for listing features and functions. For a description of the system organized from a functional perspective, see *An Introduction to DEFINITY Communications System Generic 3*, 555-230-020.

Organization of Each Feature Section

In Chapter 3 the features are arranged in alphabetical order. The information for each feature is generally presented under six headings with some differences among the sections:

- **Feature Availability**

  Defines in which release(s) the feature is available. (Also see the table later in this chapter for a listing of the G3 feature set.)

- **Description**

  Defines the feature, tells what it does for the user, or how it serves the system, and briefly describes how it is used.

- **Considerations**

  Discusses the applications and benefits of the feature, followed by the feature parameters and any other factors to be considered when the feature is used.

- **Interactions**

  Lists and briefly discusses other features that may significantly affect the feature being described. Interacting features are those that:

— Depend on each other — one of the features must be provided if the other one is.

— Cannot coexist — one of the features cannot be provided if the other one is.

— Affect each other — the normal operation of one feature modifies, or is modified by, the normal operation of the other feature.

— Enhance each other — the features, in combination, provide improved service to the user.

- **Administration**

  States whether or not administration is required, how the feature is administered, who administers the feature, and lists items requiring administration. See the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, for details.

- **Hardware and Software Requirements**

  Lists any additional hardware and/or software requirements needed for the feature.

Table 1-6 provides a complete list of the Generic G3 features and, for each feature, indicates whether the feature is standard (always part of the offering) or optional (can be purchased separately) with the Advantage Business Package, the Premier Business Package, Generic G3i, and Generic G3r.

The following notations indicate feature availability:

LEGEND:

S	Standard
O	Optional
N/A	Not Available
V2	Only available with G3V1.1, G3V2 and later releases. Not available with G3V1. Please note that feature availability between G3V1.1 is somewhat different. Any feature identified as V2 is NOT available with G3V1.1 except as an upgrade.
GD	Available with G3i-Global and G3iV2 (and later releases). Not available with G3iV1.
V3	Only available with G3V3 and later releases.
V4	Only available with G3V4 and later releases.

NOTE:
Customers outside of North America should refer to the column marked G3vs PBP for feature availability.

Standard features may require additional hardware.

Table 1-6. Feature Availability

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
AAR/ARS Partitioning	S	S	S	S	S	S
AAR/ARS Digit Conversion	O	O	O	O	S	S
Abandoned Call Search	S	S	S	S	S	S
Abbreviated Dialing	S[1]	S[2]	S[2]	S[2]	S[2]	S[2]
Abbreviated Dialing (Enhanced)	N/A	O	N/A	O	O	O
Add/Remove Skills	V3	V3	V3	V3	V3	V3
Administrable Language Displays	V2	V2	V2	V2	GD	V2
Administrable Logins	V3	V3	V3	V3	V3	V3
Administered Connections	N/A	S	N/A	S	S	S
Administration Without Hardware	S	S	S	S	S	S
Advice of Charge	V4	V4	V4	V4	V4	V4
Agent Call Handling	O[3]	O[4]	O[3]	O[4]	O[4]	O[4]
Agent Sizing	V3	V3	V3	V3	V3	V3
Alphanumeric Dialing	S	S	S	S	S	S
Alternate Facility Restriction Levels	V2	V2	V2	V2	V2	O
Answer Detection by Call Classifier	N/A	O	N/A	O	O	S
Attendant Auto-Manual Splitting	S	S	S	S	S	S
Attendant Call Waiting	S	S	S	S	S	S
Attendant Control of Trunk Group Access	S	S	S	S	S	S
Attendant Direct Extension Selection With Busy Lamp Field	S	S	S	S	S	S
Attendant Direct Trunk Group Selection	S	S	S	S	S	S
Attendant Display	S	S	S	S	S	S
Attendant Intrusion (Call Offer)	V2	V2	V2	V2	GD	S
Attendant Override of Diversion Features	V2	V2	V2	V2	GD	S
Attendant Priority Queue	V2	V2	V2	V2	GD	S
Attendant Recall	S	S	S	S	S	S
Attendant Release Loop Operation	S	S	S	S	S	S

1. Abbreviated Dialing is a standard feature; however, Enhanced Abbreviated Dialing is not available with the Advantage Business Package.
2. Abbreviated Dialing is a standard feature; however, Enhanced Abbreviated Dialing is a Premier Business Package, G3i, and G3r option.
3. Available when the Basic Call Center Option is active.
4. Available when ACD software is active.

Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
Attendant Room Status	V3	V3	V3	V3	V3	V3
Attendant Serial Calling	V2	V2	V2	V2	GD	V2
Audible Message Waiting	V2	V2	V2	V2	V2	S
Audio Information Exchange (AUDIX) Interface	S	S	S	S	S	S
Authorization Codes	O	O	O	O	O	O
Automatic Alternate Routing (AAR)	N/A	O[1]	N/A	O[1]	O[1]	O[1]
Automatic Callback	S	S	S	S	S	S
Automatic Call Distribution (ACD)	O[2]	O	O[2]	O	O	O
ACD Auto-Available Split (AAS)	V2	V2	V2	V2	V2	S
Automatic Circuit Assurance	S	S	S	S	S	S
Automatic Hold	V2	V2	V2	V2	GD	S
Automatic Incoming Call Display	S	S	S	S	S	S
Automatic Route Selection (ARS)	O	O	O	O	O	O
Automatic Transmission Measurement System	V2[1]	V2[1]	V2[1]	V2[1]	V2[1]	S
Automatic Wakeup	S	S	S	S	S	S
Auto-Start/Don't Split	V2	V2	V2	V2	GD	V2
Basic Call Management System (BCMS)	O[2]	O	O[2]	O	O	O
Bridged Call Appearance — Multi-Appearance Voice Terminal	S	S	S	S	S	S
Bridged Call Appearance — Single-Line Voice Terminal	S	S	S	S	S	S
Busy Verification of Terminals and Trunks	S	S	S	S	S	S

1. Available with Private Network Access (PNA) software.
2. Available with the Basic Call Center Option.

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Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
Call-By-Call Service Selection	N/A	O[1]	N/A	O[1]	0[1]	0[1]
Call Coverage	S[2]	S	S[2]	S	S	S
Call Detail Recording (CDR)	S	S	S	S	S	S
Call Forwarding All Calls	S	S	S	S	S	S
Call Forward Busy/Don't Answer	V4	V4	V4	V4	V4	V4
Call Management System (CMS)	O[3]	O[3]	O[3]	O[3]	O[3]	O[3]
Call Park	S	S	S	S	S	S
Call Pickup	S	S	S	S	S	S
Call Prompting	N/A	O	N/A	O	O	O
Call Vectoring	N/A	O	N/A	O	O	O
CallVisor Adjunct/Switch Application Interface (ASAI)	N/A	N/A	N/A	O	O	O
Call Waiting Termination	S	S	S	S	S	S
Centralized Attendant Service (CAS)	N/A	O	N/A	O	O	O
Class of Restriction (COR)	S	S	S	S	S	S
Class of Service (COS)	S	S	S	S	S	S
CDR Account Code Dialing	S	S	S	S	S	S
Code Calling Access	S	S	S	S	S	S
Conference — Attendant	S	S	S	S	S	S
Conference — Terminal	S	S	S	S	S	S
Constellation Voice/Data Terminal Support	V3	V3	V3	V3	V3	V3
Consult	S	S	S	S	S	S
Coverage Callback	S	S	S	S	S	S
Coverage Incoming Call Identification (ICI)	S	S	S	S	S	S
Customer-Provided Equipment (CPE) Alarm	S	S	S	S	S	S

1. Available as an option when ISDN-PRI software is purchased for public and private networking.
2. Linked Call Coverage Paths are not standard. However, Linked Call Coverage Paths are available as part of the Voice Mail Application Support Option package.
3. CMS is optionally available as an adjunct.

Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
Data Call Setup	S	S	S	S	S	S
Data Hot Line	S	S	S	S	S	S
Data-Only Off-Premises Extensions	S	S	S	S	S	S
Data Privacy	S	S	S	S	S	S
Data Restriction	S	S	S	S	S	S
DCS Alphanumeric Display for Terminals	N/A	O[1]	N/A	O[1]	O[1]	O[1]
DCS Attendant Control of Trunk Group Access	N/A	O[2]	N/A	O[2]	O[2]	O[2]
DCS Attendant Direct Trunk Group Selection	N/A	O[2]	N/A	O[2]	O[2]	O[2]
DCS Attendant Display	N/A	O[2]	N/A	O[1]	N/A	O[1]
DCS Automatic Callback	N/A	O[2]	N/A	O[1]	O[1]	O[1]
DCS Automatic Circuit Assurance (ACA)	N/A	O[1]	N/A	O[1]	O[1]	O[1]
DCS Busy Verification of Terminals and Trunks	N/A	O[2]	N/A	O[2]	O[2]	O[2]
DCS Call Coverage	N/A	V4/O[3]	N/A	V4/O[3]	V4/O[3]	V4/O[3]
DCS Call Forwarding All Calls	N/A	O[1]	N/A	O[1]	O[1]	O[1]
DCS Call Waiting	N/A	O[1]	N/A	O[1]	O[1]	O[1]
DCS Distinctive Ringing	N/A	O[1]	N/A	O[1]	O[1]	O[1]
DCS Leave Word Calling	N/A	O[1]	N/A	O[1]	O[1]	O[1]
DCS Multi-Appearance Conference/Transfer	N/A	O[1]	N/A	O[1]	O[1]	O[1]

1. Available when DCS software is purchased.
2. Available when DCS is purchased, but not available with DCS over PRI D-Channel when the PRI D-Channel is connected to the public network.
3. Optional feature with DCS.

Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
DCS Over ISDN-PRI D-Channel	N/A	O[1]	N/A	O[1]	O[1]	O[1]
DCS Trunk Group Busy/Warning Indication	N/A	O[1]	N/A	O[1]	O[1]	O[1]
Default Dialing	S	S	S	S	S	S
Dial Access to Attendant	S	S	S	S	S	S
Dial Plan	S	S	S	S	S	S
Digital Multiplexed Interface	S	S	S	S	S	S
Direct Department Calling (DDC) and Uniform Call Distribution (UCD)	S	S	S	S	S	S
Direct Inward Dialing (DID)	S	S	S	S	S	S
Direct Inward and Outward Dialing (DIOD) — International	S	S	S	S	S	S
Direct Outward Dialing (DOD)	S	S	S	S	S	S
Distinctive Ringing	S	S	S	S	S	S
Do Not Disturb	S	S	S	S	S	S
DS1 Trunk Service	S	S	S	S	S	S
E1 Trunk Service	V2	V2	V2	V2	GD	V2
EIA Interface	S	S	S	S	S	S
Emergency Access to the Attendant	S	S	S	S	S	S
Enhanced DCS (EDCS)	V2	V2	V2	V2	GD	V2
End-to-End Signaling	V2	V2	V2	V2	GD	V2
Expert Agent Selection	N/A	V2†	N/A	V2†	V2†	V2†
Extension Number Portability	V2	V2	V2	V2	V2	S
Extended Trunk Access	V2	V2	V2	V2	V2	S

1. Available when DCS software is purchased.

Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
Facility and Non-Facility Associated Signaling	N/A	O[1]	N/A	O[1]	O[1]	O[1]
Facility Busy Indication	S	S	S	S	S	S
Facility Restriction Levels (FRLs)	S	S	S	S	S	S
Facility Test Calls	S	S	S	S	S	S
Flexible Billing	V4/O	V4/O	V4/O	V4/O	V4/O	V4/O
Forced Entry of Account Codes	N/A	O	N/A	O	O	O
Forced Password Aging	V3	V3	V3	V3	V3	V3
Generalized Route Selection	N/A	O[1]	N/A	O[1]	O[1]	O[1]
Go to Cover	S	S	S	S	S	S
Hold	S	S	S	S	S	S
Hold—Automatic	V2	V2	V2	V2	GD	S
Hot Line Service	S	S	S	S	S	S
Hunting	S	S	S	S	S	S
Inbound Call Management	N/A	O	N/A	O	O	O
Individual Attendant Access	S	S	S	S	S	S
Information System Network (ISN) Interface	S	S	S	S	S	S
Integrated Directory	S	S	S	S	S	S
Integrated Services Digital Network — Basic Rate Interface	N/A	N/A	S	S	S	S
Integrated Services Digital Network — Primary Rate Interface	N/A	O	N/A	O	O	O
Intercept Treatment	S	S	S	S	S	S
Intercom — Automatic	S	S	S	S	S	S
Intercom — Dial	S	S	S	S	S	S
Internal Automatic Answer	S	S	S	S	S	V2
Inter-PBX Attendant Calls	S	S	S	S	S	S
Intraflow and Interflow	O[2]	O[3]	O[2]	O[3]	O[3]	O[3]
Last Number Dialed	S	S	S	S	S	S

1. Available when ISDN-PRI software is purchased for public and private networking.
2. Available when the Basic Call Center Option is purchased.
3. Available when ACD software is purchased.

Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
Leave Word Calling	O[1]	S	O[1]	S	S	S
Line Lockout	S	S	S	S	S	S
Look Ahead Interflow	N/A	O	N/A	O	O	O
Loudspeaker Paging Access	S	S	S	S	S	S
Loudspeaker Paging Access — Deluxe	S	S	S	S	S	S
Malicious Call Trace	V2	V2	V2	V2	V2	S
Manual Message Waiting	S	S	S	S	S	S
Manual Originating Line Service	S	S	S	S	S	S
Manual Signaling	S	S	S	S	S:S	
MERLIN./System 25 — Voice Terminal Support (731xH Series)	S	S	S	S	S	V2
Modem Pooling	S	S	S	S	S	S
Move Agents from CMS	O[2]	O[2]	O[2]	O[2]	O[2]	O[2]
Multi-Appearance Preselection and Preference	S	S	S	S	S	S
Multiple Call Handling	V3/O	V3/O	V3/O	V3/O	V3/O	V3/O
Multiple Listed Directory Numbers	S	S	S	S	S	S
Music-on-Hold Access	S	S	S	S	S	S
Names Registration	S	S	S	S	S	S
Network Access — Private	S	S	S	S	S	S
Network Access — Public	S	S	S	S	S	S
Night Service — Hunt Group	S	S	S	S	S	S
Night Service — Night Console Service	S	S	S	S	S	S
Night Service — Night Station Service	S	S	S	S	S	S
Night Service — Trunk Answer from Any Station	S	S	S	S	S	S

1. Available when the Voice Mail Application Support Option is purchased.
2. CMS is optionally available as an adjunct. Move Agents from CMS is only available if CMS is selected.

Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
Night Service — Trunk Group	S	S	S	S	S	S
Off-Premises Station	S	S	S	S	S	S
PC Application Software Translation Exchange (PASTE)	V4	V4	V4	V4	V4	V4
PC Interface	S	S	S	S	S	S
PC/PBX Connection	S	S	S	S	S	S
Personal Central Office Line (PCOL)	S	S	S	S	S	S
Personalized Ringing	S	S	S	S	S	S
Power Failure Transfer	S	S	S	S	S	S
Priority Calling	S	S	S	S	S	S
Privacy — Attendant Lockout	S	S	S	S	S	S
Privacy — Manual Exclusion	S	S	S	S	S	S
Property Management System Interface	S	S	S	S	S	S
Pull Transfer	V2	V2	V2	V2	GD	V2
QSIG Global Networking	V2	V2	V2	V2	V2	V2
Queue Status Indications	S	S	S	S	S	S
Recall Signaling	S	S	S	S	S	S
Recent Change History	S	S	S	S	S	S
Recorded Announcement	S	S	S	S	S	S
Recorded Telephone Dictation Access	S	S	S	S	S	S
Redirection On No Answer (RONA)	V2	V2	V2	V2	V2	V2
Remote Access	S	S	S	S	S	S
Report Scheduler and System Printer	S	S	S	S	S	S
Restriction — Controlled	S	S	S	S	S	S
Restriction — Fully Restricted Service	V2	V2	V2	V2	GD	S
Restriction — Miscellaneous Terminal	S	S	S	S	S	S
Restriction — Miscellaneous Trunk	S	S	S	S	S	S

Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
Restriction — Toll	S	S	S	S	S	S
Restriction — Voice Terminal — Inward	S	S	S	S	S	S
Restriction — Voice Terminal — Manual Terminating Line	S	S	S	S	S	S
Restriction — Voice Terminal — Origination	S	S	S	S	S	S
Restriction — Voice Terminal — Outward	S	S	S	S	S	S
Restriction — Voice Terminal — Public	V2	V2	V2	V2	GD	V2
Restriction — Voice Terminal — Termination	S	S	S	S	S	S
Ringback Queuing	S	S	S	S	S	S
Ringer Cutoff	S	S	S	S	S	S
Ringing — Abbreviated and Delayed	V4	V4	V4	V4	V4	V4
Rotary Dialing	S	S	S	S	S	S
R2-MFC	V2	V2	V2	V2	GD	V2
Security Violation Notification (SVN)	S	S	S	S	S	S
Send All Calls	S	S	S	S	S	S
Senderized Operation	S	S	S	S	S	S
Service Observing	S	S	S	S	S	S
Single-Digit Dialing and Mixed Station Numbering	S	S	S	S	S	S
Straightforward Outward Completion	S	S	S	S	S	S
Subnet Trunking	S	S	S	S	S	S
Switch Based Bulletin Board	V3	V3	V3	V3	V3	V3
System Measurements	O[1]	S	O[1]	S	S	S
System Status Report	S	S	S	S	S	S
Temporary Bridged Appearance	S	S	S	S	S	S
Tenant Partitioning	V4/O	V4/O	V4/O	V4/O	V4/O	V4/O
Terminal Translation Initiation	V2	V2	V2	V2	V2	S
Terminating Extension Group	S	S	S	S	S	S

1. Available when the System Measurements Option package or the Basic Call Center Option package is purchased.

Table 1-6. Feature Availability — *Continued*

Feature	G3vs ABP	G3vs PBP	G3s ABP	G3s PBP	G3i	G3r
Through Dialing	S	S	S	S	S	S
Time of Day Routing	O[1]	O[1]	O[1]	O[1]	O[1]	O[1]
Timed Reminder and Attendant Timers	S[2]	S[2]	S[2]	S[2]	S[2]	S[2]
Touch-Tone Dialing	S	S	S	S	S	S
Transfer	S	S	S	S	S	S
Transfer Outgoing Trunk to Outgoing Trunk	V2[3]	V2	V2[3]	V2	V2	S
Traveling Class Marks (TCMs)	N/A	O	N/A	O	0	0
Trunk Flash	S	S	S	S	S	S
Trunk Group Busy/Warning Indicators to Attendant	S	S	S	S	S	S
Trunk Identification By Attendant	S	S	S	S	S	S
Trunk-to-Trunk Transfer	S	S	S	S	S	S
Uniform Dial Plan (UDP)	N/A	O	N/A	O	O	O
Unrestricted Uniform Dial Plan (UDP)	NA	V2	NA	V2	V2	
VDN of Origin Announcements	V3	V3	V3	V3	V3	V3
Visually Impaired Attendant Services (VIAS)	V2	V2	V2	V2	GD	V2
Voice Message Retrieval	S	S	S	S	S	S
Voice Response Integration	V2†	V2†	V2†	V2†	V2†	V2†
Voice Terminal Alerting Options	V2	V2	V2	V2	V2	V2
Voice Terminal Display	S	S	S	S	S	S
VuStats	V3	V3	V3	V3	V3	V3
Wideband Switching	V2†	V2†	V2†	V2†	V2†	V2†
World Class Core BRI	V4	V4	V4	V4	V4	V4
World Class Tone Detection	V2	V2	V2	V2	GD	V2
World Class Tone Generation	V2	V2	V2	V2	GD	V2

1. Available with ARS or Private Network Access (PNA) software.
2. Timed Reminder and Attendant Timers held call timer transfer to ‘‘aatd.’’ Standard in G3vs/G3sABP, G3vs/G3sPBP, G3i, and G3r.
3. Transfer Outgoing Trunk to Outgoing Trunk is not available with G3vs/G3s ABP.

Functional Description

2

Overview

This chapter describes the DEFINITY switch features, functions, and services, and presents these capabilities classified into six categories: Voice Management, Data Management, Network Services (including World Class Routing), System Management, Hospitality Services, and Call Center Services. Each group of functions and services, including a listing of associated features, is described separately in a section of this chapter.

See Chapter 3 for detailed information on individual features. (The feature descriptions in Chapter 3 are arranged in alphabetical order.)

NOTE:
It is useful to consider the features and functions from the perspective of the following functional views: System View, Call Center View, PBX-to-Host View, Voice Processing View, Network View (including World Class Routing), System Management View, Desktop View, Hospitality View, and Support View. Some features and functions support several of these views, so the views should not be seen as discrete categories for listing features and functions. For a description of the system organized from a functional perspective, see *An Introduction to DEFINITY. Communications System Generic 3*, 555-230-020.

Voice Management Overview

The Voice Management features try to meet the individual communications needs of everyone in the system. As the individual needs change, the assigned features can also be changed. The Voice Management features provide many important services with benefits such as saving time and making calling more convenient.

Voice Management Features

The following features are associated with Voice Management:

- Abbreviated Dialing
- Alternate Facilities Restriction Levels
- Attendant Auto-Manual Splitting
- Attendant Call Waiting
- Attendant Control of Trunk Group Access
- Attendant Direct Extension Selection With Busy Lamp Field
- Attendant Direct Trunk Group Selection
- Attendant Display
- Attendant Intrusion (Call Offer)
- Attendant Override
- Attendant Priority Queue
- Attendant Recall
- Attendant Release Loop Operation
- Attendant Serial Calling
- Audio Information Exchange (AUDIX) Interface
- Audible Message Waiting
- ACD - Auto-Available Split
- Auto-Start/Don't Split
- Automatic Callback
- Automatic Incoming Call Display
- Automatic Transmission Measurement System (ATMS)
- Authorization Codes
- Automatic Callback
- Automatic Incoming Call Display
- Bridged Call Appearance — Multi-Appearance Voice Terminal

- Bridged Call Appearance — Single-Line Voice Terminal
- Busy Verification of Terminals and Trunks
- Call By Call Service Selection
- Call Coverage
- Call Forwarding All Calls
- Call Forward Busy/Don’t Answer
- Call Park
- Call Pickup
- Call Waiting Termination
- CDR Account Code Dialing
- Centralized Attendant Service
- Class of Restriction
- Class of Service
- Code Calling Access
- Conference — Attendant
- Conference — Terminal
- Consult
- Coverage Callback
- Coverage Incoming Call Identification
- Dial Access to Attendant
- Dial Plan
- Direct Department Calling and Uniform Call Distribution
- Direct Inward Dialing
- Direct Outward Dialing
- Distinctive Ringing
- Emergency Access to the Attendant
- Facility Busy Indication
- Go To Cover
- Hold
- Hold - Automatic
- Hot Line Service
- Hunting
- Individual Attendant Access

- Integrated Directory
- Integrated Services Digital Network — Basic Rate Interface (ISDN-BRI)
- Intercept Treatment
- Intercom — Automatic
- Intercom — Dial
- Internal Automatic Answer (G3i)
- Inter-PBX Attendant Calls
- Last Number Dialed
- Leave Word Calling
- Line Lockout
- Loudspeaker Paging Access
- Loudspeaker Paging Access — Deluxe
- Manual Message Waiting
- Manual Originating Line Service
- Manual Signaling
- MERLIN/System 25 Voice Terminal Support — 731xH Series
- Misoperation Handling
- Multi-Appearance Preselection and Preference
- Multiple Listed Directory Numbers
- Multiple Music-on-Hold
- Music-on-Hold Access
- Night Service — Hunt Group
- Night Service — Night Console Service
- Night Service — Night Station Service
- Night Service — Trunk Answer From Any Station
- Night Service — Trunk Group
- Personal Central Office Line
- Personalized Ringing
- Power Failure Transfer
- Priority Calling
- Privacy — Attendant Lockout
- Privacy — Manual Exclusion
- Recall Signaling

- Recorded Announcement
- Recorded Telephone Dictation Access
- Remote Access
- Restriction — Controlled
- Restriction — Fully Restricted Service
- Restriction — Miscellaneous Terminal
- Restriction — Miscellaneous Trunk
- Restriction — Toll
- Restriction — Toll/Code
- Restriction — Voice Terminal — Inward
- Restriction — Voice Terminal — Manual Terminating Line
- Restriction — Voice Terminal — Origination
- Restriction — Voice Terminal — Outward
- Restriction — Voice Terminal — Termination
- Ringback Queuing
- Ringer Cutoff
- Ringing Abbreviated and Delayed
- Rotary Dialing
- Send All Calls
- Senderized Operation
- Single-Digit Dialing and Mixed Station Numbering
- Straightforward Outward Completion
- Temporary Bridged Appearance
- Tenant Partitioning
- Terminal Translation Initialization
- Terminating Extension Group
- Through Dialing
- Timed Reminder
- Touch-Tone Dialing
- Transfer
- Transfer - Outgoing Trunk to Outgoing Trunk Trunk Flash
- Trunk Group Busy/Warning Indicators to Attendant
- Trunk Identification by Attendant

- Trunk-to-Trunk Transfer
- Voice Message Retrieval
- Voice Terminal Display
- Voice Terminal Alerting Options

Specific Attendant Features are listed below.

- DCS Attendant Control of Trunk Group Access
- DCS Attendant Display
- DCS Attendant Direct Trunk Group Selection
- Attendant Auto-Manual Splitting
- Attendant Control of Trunk Group Access
- Attendant Call Waiting
- Attendant Direct Extension Selection With Busy Lamp Field
- Attendant Direct Trunk Group Selection
- Attendant Intrusion (Call Offer)
- Attendant Override of Diversion Features
- Attendant Priority Queue
- Attendant Recall
- Attendant Release Loop Operation
- Attendant Serial Calling
- Attendant Display
- Conference — Attendant
- Centralized Attendant Service (CAS)
- Dial Access to Attendant
- Emergency Access to the Attendant
- Individual Attendant Access
- Inter-PBX Attendant Calls
- Privacy — Attendant Lockout
- Trunk Group Busy/Warning Indicators to Attendant
- Trunk Identification By Attendant
- Timed Reminder and Attendant Timers
- Visually Impaired Attendant Service (VIAS)

Data Management

The DEFINITY switch is a private digital switching system that permits connections with a variety of data equipment. Data terminals, printers, graphics, facsimile equipment, and computers can be connected to the switch through various protocols or interfaces. The physical connection can be through a digital data module, analog modem, or access endpoint.

For use outside the United States, modems that comply with the ITU-T 108.1 signaling procedures are supported. Administration forms are available to support combined (external) modem pools.

The system provides the ability to option data modules [or data-like devices such as a Data Line Circuit (DLC)] for Terminal Dialing. Also, data modules can be used without Terminal Dialing with host computers, printers, or other such applications. Computer file transfer at a rate of 64 kbps is possible with the Modular Processor Data Module (MPDM) and the Modular Trunk Data Module (MTDM).

The family of data modules includes a Wideband Data Module (WDM), Processor Data Module (PDM), a Digital Terminal Data Module (DTDM), a Trunk Data Module (TDM), a Z702AL1-DSU Data Module Base, a 7400A Data Module, a 7500B Data Module, an ISDN Asynchronous Data Module (ADM), and a 3270 Data Module. The data modules are generally more versatile than modems, operate at faster data rates, and provide additional features.

The WDM provides Wideband (128 kbps plus any multiple of 64 kbps, up to a total of 1984 kbps) communications between a Wideband data and dialing interfaces and an ISDN PRI interface. WDM provides for extremely high-speed data transmission and is used by applications ranging from video conferencing to data backup.

The DTDM provides synchronous or asynchronous data communications to 7403D and 7405D digital voice terminal users who have a terminal or personal computer. The DTDM and voice terminal integrate data and voice into the Digital Communications Protocol (DCP) to the digital switch.

The Z702AL1-DSU Data Module Base provides the Data Communications Equipment (DCE) interface connection for a 7407D voice terminal to data terminals. The module provides full-duplex asynchronous operation only. The module and 7407D voice terminal integrate data and voice into the DCP to the digital switch.

The MTDM provides an EIA RS-232D Data Terminal Equipment (DTE) interface for connection to off-premises (out of building) private-line trunk facilities, or a switched telecommunications network, and a DCP interface for connection to the digital switch. The MTDM may also serve as part of a conversion resource for modem pooling. The MTDM is also used to interface with DCE-type multiplexers.

The MPDM provides a DCE asynchronous or synchronous interface for connection to data terminals, Call Detail Recording (CDR) output devices, DEFINITY Communications System Generic 3 Management Terminals (G3-MT), Generic 3 Management Application Systems (G3-MA), on-premises (in building) administration terminals, and host computers. The MPDM can be preset in the factory to provide the following interfaces: EIA RS-232C, RS-449, V.35, and RS-366 to support Automatic Calling Unit (ACU) type dialing. The MPDM can be configured to support the Data Call Setup or Off-Premises Data-Only Extension feature. The MPDM also supports data rates of 56 and 64 kbps for downloading and other high-speed data transfer requirements.

The 7400A Data Module may be used instead of an MTDM when supporting the combined Modem Pooling feature. The 7400A Data Module supports asynchronous operation and provides a DCP interface to the switch and an EIA RS-232C interface to the associated modem. The 7400A Data Module also can be used with a data terminal and supports keyboard dialing in the same manner as the MPDM.

The 7500B Data Module is a stand-alone unit that supports asynchronous or synchronous DCE and asynchronous DTE on the ISDN Basic Rate Interface (BRI) switch interface (G3, DEFINITY Generic 2 (G2) and the 5ESS switch). In asynchronous mode, the 7500B supports packet or circuit-switched data communications, and can be controlled via the front panel or the keyboard of a connected terminal. In synchronous mode, the 7500B supports circuit-switched or nailed-up data communications, requires either the Multipurpose Enhancement Board or the High-Speed Synchronous Enhancement Board, and only can be controlled via the front panel.

When configured as an asynchronous DCE, the 7500B provides an EIA RS-232D interface and supports full-duplex data transmission at rates of 300, 1200, 2400, 4800, 9600, and 19200 bps. The following optional enhancements are available for the 7500B in an asynchronous DCE configuration: an RS-366 ACU interface and a second asynchronous EIA RS-232D interface. With an additional asynchronous EIA RS-232D interface, the 7500B can simultaneously support either two D-channel packet data calls or one D-channel packet call and one B-channel circuit call. However, the 7500B cannot simultaneously support two B-channel circuit-switched calls.

When configured as an asynchronous DTE, the 7500B provides an EIA RS-232D interface and supports full-duplex data transmission at rates of up to 19200 bps. This configuration is most commonly used for modem pooling applications.

In order to be configured as a synchronous DCE, the 7500B must have either the Multipurpose Enhancement Board or the High-Speed Synchronous Enhancement Board. With the Multipurpose Board, the 7500B provides an EIA RS-232D interface and an RS-366 ACU interface, and supports full-duplex data transmission at rates of 1200, 2400, 4800, 9600, 19200, 56000, and 64000 bps. The 7500B also supports half-duplex emulation at rates of 1200, 2400, 4800, 9600, 19200, and 56000 bps. With the High-Speed Synchronous Enhancement Board, the 7500B provides a V.35 interface and supports full-duplex data transmission at rates of 48000, 56000, and 64000 bps. The 7500B only provides half-duplex emulation at a rate of 56000 bps. Regardless of the configuration, the 7500B provides no voice functions and is not used with voice terminals.

The ISDN ADM may be used with asynchronous DTE as a data stand for 7500-series BRI voice terminals. Consisting of a board located inside the BRI voice terminal, the ISDN ADM allows simultaneous voice and data transmissions through one terminal. The ISDN ADM supports the standard Hayes command set for compatibility with existing PC communications packages and provides AT&T extensions to the standard Hayes command set to allow even greater flexibility in future applications. PC applications that use the ISDN ADM Applications Programming Interface (API) can simultaneously control, monitor, and process both voice and data calls.

The DLC, which provides eight ports to connect user's asynchronous EIA RS-232D interface to DTE, can be used as an alternative to DTDM or PDM.

Data modules support the following interfaces:

- All data modules (except the WDM, MPDM, and 3270) provide a modified EIA RS-232D interface.
- The WDM provides a ITU-T interface for Wideband transmissions.
- The MPDM provides either EIA RS-232D V.35 or RS-449 interface. The MPDM can also emulate an Automatic Call Unit (ACU) and supports the RS-366 interface. The ACU emulation and RS-366 interface are required for Keyboard Dialing and are discussed in the Data Call Setup feature description.
- The 3270 Data Module provides a Category A coaxial DCE interface for connection to 3270-type data terminals or a cluster controller. It also provides a DCP interface for connection to the digital switch.

The 3270 Data Module is available in the following three models:

- 3270T (Terminal) — connects to a Category A 3270-type terminal, such as the IBM. 3278 Information Delivery System. The 3270T Data Module must connect through the switch to a 3270C (Controller) Data Module.
- 3270A (Asynchronous) — provides the same function as the 3270T Data Module. It also allows the 3270-type terminal to emulate a Digital Equipment Corporation VT100 or an AT&T asynchronous terminal.
- 3270C (Controller) — connects an IBM 3274 or 3276 cluster controller to the switch. A 3270C Data Module can contain as many as eight ports.

Trunks or channels of a DS1/E1 can also be used as non-signaling data endpoints with the Access Endpoints function. An access endpoint is either a non-signaling channel on a DS1/E1 interface or a non-signaling port on an Analog Tie Trunk circuit pack that is assigned a unique extension. Since an access endpoint is non-signaling, it neither generates nor responds to signaling. As a result, an access endpoint cannot be used as a trunking facility (it cannot receive incoming calls or route outgoing calls). An access endpoint is used primarily to support devices, switches, or services that have a trunk interface but do not support signaling for the trunk. An access endpoint may be designated as the originating (local) endpoint or destination endpoint in an Administered Connection. The status of an access endpoint can be displayed by entering the **status access-endpoint** command from the G3-MT/G3-MA or a PC with a terminal emulator.

The system supports digital-to-digital, digital-to-analog, analog-to-digital, and analog-to-analog data calls. For data calls, the user can access the system through these digital or analog data endpoints. Digital data endpoints are data modules and associated data equipment, PCs, and data channels [used for remote G3-MT terminals, and CDR]. Analog data endpoints are modems (or acoustic coupled modems) and associated data equipment connected to the system through analog lines or trunks. Voice-band data calls using modems can be connected to the system through digital trunks.

The system supports DCP. This protocol provides framing, control, and signaling for each of two information channels. Only one channel is used for voice-only or data-only applications. Both channels are used for simultaneous voice and data transmission. Simultaneous voice and data information can be transmitted on calls to or from a 7403D or 7405D voice terminal with a DTDM, a 7404D with its built-in data module, any 7400-series with an optional data module base, and any 8400 series voice terminal with listed 7400 data module. Calls to or from other equipment are either voice-only or data-only.

ISDN-BRI provides one 16 kbps signaling channel (D-channel) and two 64 kbps information channels (B-channels), with a total information rate of 144 kbps. The primary purpose of the signaling channel is to convey Q.931 message-oriented signaling for the setup and tear down of calls carried by the B-Channels on the BRI. Since all the signaling is done on the D-channel, both B-channels are ‘‘clear.’’ As a result, the entire widths of the B-channels are used for simultaneously carrying voice and circuit-switched data. Voice and data information can be simultaneously transmitted on calls to or from a 7505, 7506, or 7507 voice terminal equipped with an optional ADM. Without the optional ADM, the 7505, 7506, and 7507 voice terminals can handle voice-only calls.

Data Networking

Data networking connects two or more data endpoints. The system is a highly reliable, centralized switch that provides switched access between endpoints. Typical data communications configurations for the system are shown in Figure 2-1.

Switched access allows one terminal to connect to one of any number of devices. Therefore, more effective use of data equipment is obtained than with dedicated (hard-wired) links. Switched access also reduces the need for duplicated (dedicated) equipment. Switched Access systems can emulate hardwired networks through use of the Administered Connections feature.

The system uses twisted-pair standard building wiring and eight-pin modular wall jacks. Each wall jack is a single outlet that can handle simultaneous voice and data information.

The digital switch, data modules, DCP, twisted-pair wiring, modular wall jacks, and switched data features give the system its unique capabilities. These capabilities merge the business office data processing and telecommunications functions into a single system.

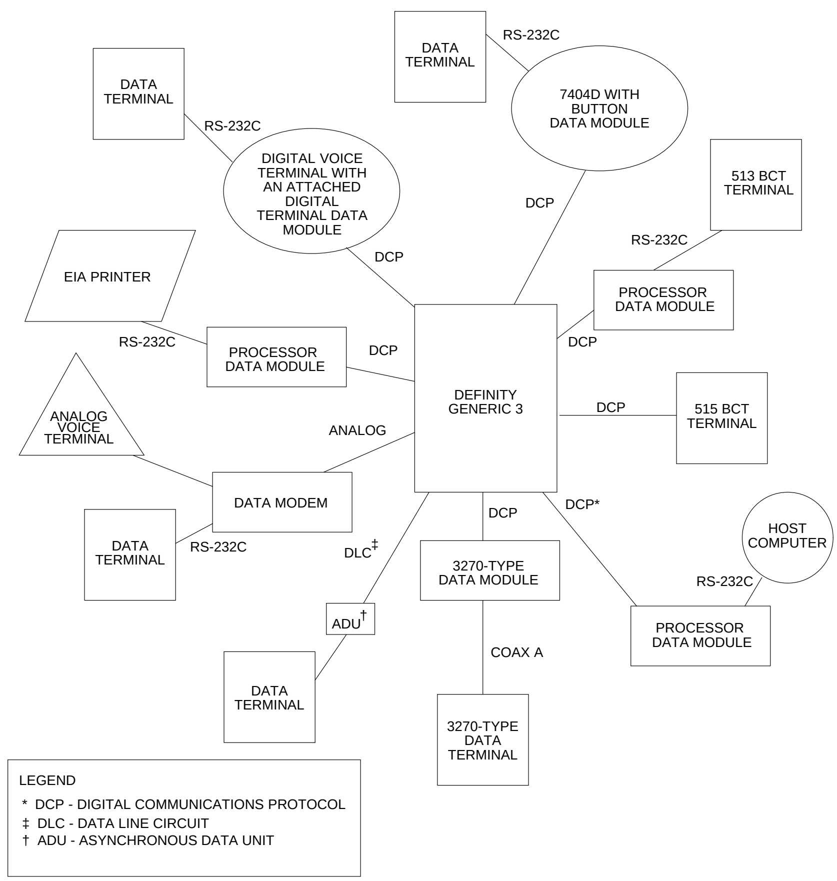


Figure 2-1. System Data Communications Configuration

Generally, data networks are either local area networks, extended networks, or combinations of both. The two networks and their implementation within the system are defined as follows:

- **Local Area Networks (LANs)**

  The system provides this capability by connecting communication devices physically located within a local-area or campus-like environment.

  These include conventional, semi-intelligent, and intelligent data terminals, personal computers, host computers, and virtually any device with the proper communications interface.

  The centralized network provides circuit-switched paths using twisted-pair building cable that extends to the endpoints. Since the business office equipment can access multiple data systems, the data equipment and applications can be used more productively. The system also provides several data-related features that are easy to use and contribute to expedient use of the system and its networking capabilities.

- **Extended Networks**

  Extended networks mainly provide connections between the system and other distant switches, including remote access facilities. With remote access facilities, a local terminal can access remote host computers. Also, remote terminals can access either local computer facilities or other remote computer facilities. Extended networks are constructed of analog or digital facilities and can be either public or private. Typical networking configurations are shown in Figure 2-2. Public networks available in the US include the following:

  — Local Central Office (CO) switching extended through direct distance dialing

  — Foreign exchange (FX) central office trunking

  — Wide Area Telecommunications Service (WATS)

  — MEGACOM Telecommunications Service

  — MEGACOM 800 Telecommunications Service

  — Software Defined Network (SDN)

  — Software Defined Data Network (SDDN) (G3i)

  — ACCUNET. Digital Service

- Private networks include:

  — AT&T DATAPHONE. Data Communications Service

  — Distributed Communications System (DCS)

  — Electronic Tandem Network (ETN)

  — Enhanced Private Switched Communications Service (EPSCS)

  — Private line (PL)

  — Software Defined Network (SDN)

  — Software Defined Data Network (SDDN)

  — Tandem tie trunk

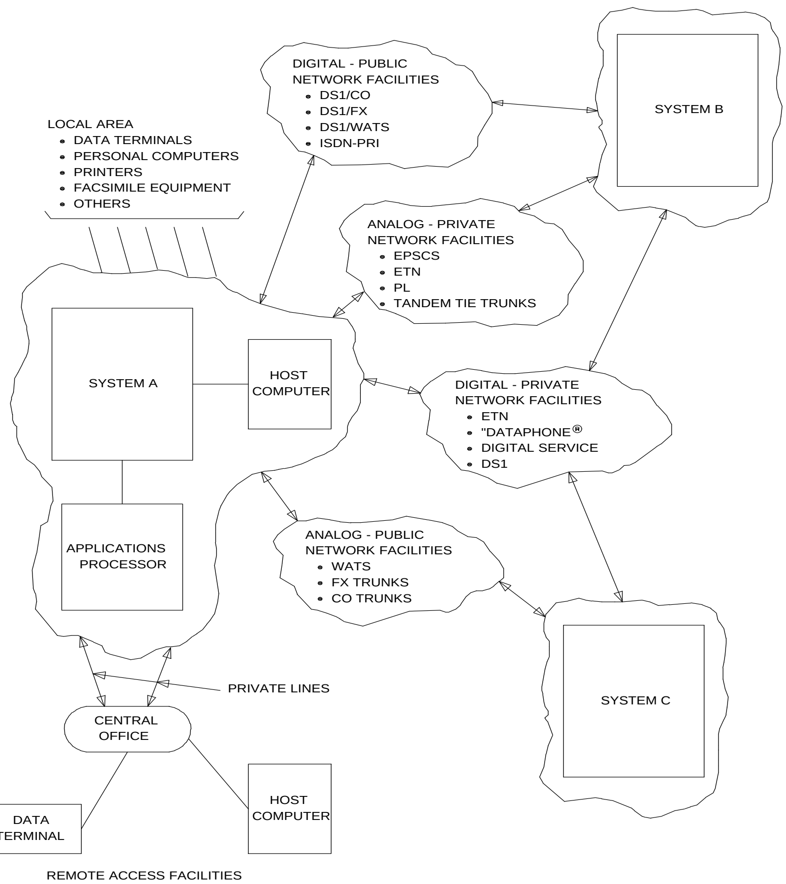


Figure 2-2. System Networking Configurations

Data Communications Protocols and Interfaces

A protocol is a set of conventions or rules that governs how data is transmitted and received. The rules generally cover the following:

- Physical interface
- Mechanical interface
- Electrical interface
- Framing

- Error detection and control

Communications protocols are designed to meet the transmission requirements for specific data exchange and data communications equipment. These communications protocols are sponsored by a national or international organization or a major corporation. The system equipment and communications processing software provide the following protocols:

- ISDN Protocols
- EIA RS-232C
- RS-449
- RS-366
- Standard Serial Interface (SSI)
- Teletypewriter (TTY) Modes
- Digital Communications Protocol (DCP)
- BX.25 Packet Switching
- ITU-T V.35
- Wideband Switching
- Binary Synchronous Communications (Bisync)

ISDN Protocols

The ISDN Q.931 Protocol is used to support Layer 3 call control signaling for both the network and user sides of an ISDN Primary Rate Interface (PRI). Both DS1 and E1 digital transmission standards are supported on a per-interface basis. This implementation provides call state transition, proper message content, and error recovery, as well as protocol support for other related features. For all switches (except G3vs/G3s ABP), the ISDN Q.932 Protocol is used to support the CallVisor Adjunct Switch Applications Interface (ASAI) required for the Inbound Call Management feature.

Electronic Industries Association (EIA)

EIA RS-232D

This protocol is widely used for short distance and low-speed applications such as data terminals and modems connecting data terminals. The data link consists of a 25-conductor cable. The conductors are used for data-link control and timing, as well as for transmitting and receiving signals. Data-link control is accomplished by handshake signaling between the transmit and receive devices. Data speeds are limited to 19.2 kbps or fewer.

The EIA RS-232D protocol provides two interface connectors. The female side connector is known as data communications equipment (DCE). The male side connector is known as data terminal equipment (DTE). Data equipment manufacturers design either the DCE or DTE interface into their products. Products such as modems, data service units (DSUs), Digital Terminal Data Modules (DTDMs), and Processor Data Modules (PDMs) have a built-in DCE interface. Products such as some types of multiplexers, data terminals, printers, computer ports, and Trunk Data Modules (TDMs) have a built-in DTE interface. Modular Data Modules (MDMs) can be configured as either DCE or DTE.

The maximum cable length recommended by EIA for the EIA RS-232D protocol is 25 feet (15 meters). However, practical applications have shown that the cable length can be much greater. Factors limiting cable length include transmission speed, cable capacitance, and nearness of noise sources such as fluorescent lights or electric generators. Each application should be considered separately.

RS-449

This protocol allows longer cables than the EIA RS-232D. Maximum cable lengths for various data speeds are as follows:

- 19.2 kbps — 200 feet (61 meters)
- 9.6 kbps — 400 feet (122 meters)
- 4.8 kbps — 800 feet (244 meters)
- 2.4 kbps — 1,600 feet (488 meters)

The RS-449 protocol is provided as a communications link interface on the Applications Processor (AP). This standard uses a 37-conductor cable. The AP RS-449 interface contains unbalanced driver/receivers that also permit interconnection to the EIA RS-232D interface when used with a 37- to 25-pin cable adapter. Since the AP RS-449 interface is compatible with the EIA RS-232D protocol, it also is limited to the same maximum 19.2 kbps data rate.

RS-366

The RS-366 communications protocol specifies the standards for interfacing computers to ACUs. This permits a computer to originate data calls over a switched telephone network. The AP provides one RS-366 interface for each six EIA RS-232D interface ports.

Digital Communications Protocol (DCP) Interface

The DCP is used by the system's digital switch, digital voice terminals, data modules, the 510D terminal, and the 515 BCT. This protocol permits simultaneous voice and data over the same communications link to the switch.

The DCP consists of a 160 kbps, four-wire serial data link that operates full-duplex over standard twisted-pair building cable. For data-only transmission, the maximum cable length is 5,000 feet (1,524 meters). When voice and data transmission is carried over the same data link, as when a 510D terminal, 515 BCT, or a DTDM is used, the cable length is limited by the voice transmission distance.

The DCP sends digitized voice and digital data in frames. Each frame consists of four fields or channels (see Figure 2-3). The first field is a unique three-bit framing pattern that defines the frame boundary. The second field is a one-bit control or signaling channel between the digital switch and digital data endpoint. The third and fourth fields are two independent information (I) channels. The information channels are eight bits each and are used to send digitized voice or digital data.

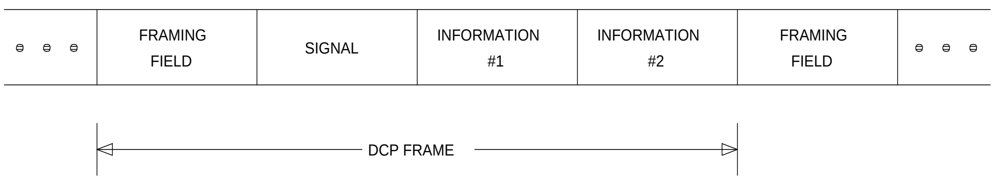


Figure 2-3. Digital Communications Protocol Frame Structure

There are 8,000 frames per second. Therefore, the bit rate available is eight for the signaling channel and 64 kbps for the information channel. The digital switch routes each information channel independently so that simultaneous voice and data can be completed to different destinations.

The full capacity of the information channels (64 kbps) is available for digitized voice. Data terminals typically operate at speeds from below 300 bps up to 19.2 kbps, asynchronous or synchronous. The DCP uses data modules to map the data terminal data into a 64 kbps information channel.

The framing rate of 8,000 per second and eight bits per information channel is consistent with other telecommunication systems such as the DS1/E1 carrier. This minimizes potential conversion problems when interfacing to different digital facilities.

BX.25 Packet Switching Protocol

The BX.25 protocol implements the international standard for packet switching. It is a multilayered protocol. [Layering is a structuring of specific protocol functions (for example, error detection and correction) that are grouped together as a unique layer or level.]

The BX.25 protocol is similar to the ITU-T X.25 protocol and, from a user perspective, is compatible with the standard. The BX.25 protocol has three layers that are not specified for the X.25 protocol. These layers are Application, Presentation, and Session. The Application and Presentation layers are defined in the Transaction-Oriented Protocol (TOP) of the BX.25.

The TOP is a high-level protocol, intended to standardize communications between transaction-oriented systems. Transaction-oriented communications involve communication of small messages or requests describing a single unit of work that may result in a reply being sent back to the originating system. The Session layer is intended to establish, manage, and terminate sessions for use by higher-level protocols or, in some cases, by user applications directly. Other differences between X.25 and BX.25 are as follows:

- The X.25 protocol specifies network standards only; the BX.25 protocol places requirements on the user interface as well.
- The X.25 protocol provides for datagram services while the BX.25 protocol does not. Datagram service has not been implemented within the continental United States.
- The X.25 protocol leaves the users in a point-to-point environment to develop their own solutions to the following areas of potential conflict, while the BX.25 protocol provides solutions:
  - — Link layer addressing
  - — Logical channel selection
  - — Call collision

Basic elements of the Application and Presentation layers must be user defined under both protocols. The following figures shows the relationship and similarity between the BX.25 and X.25 protocols.

The BX.25 protocol is used in the system to provide communications between the switch and the switch-related features. The BX.25 protocol is also used in the system to provide communications between the switch and the AUDIX and to provide communication between DCS switches

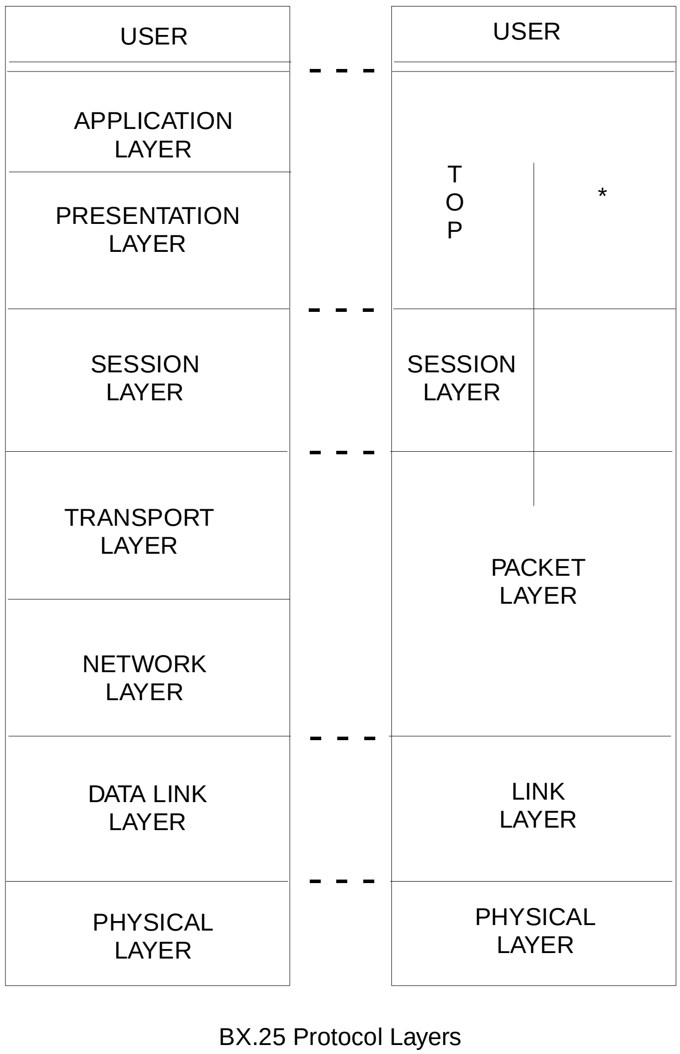


Figure 2-4. BX.25 Packet-Switching Protocol

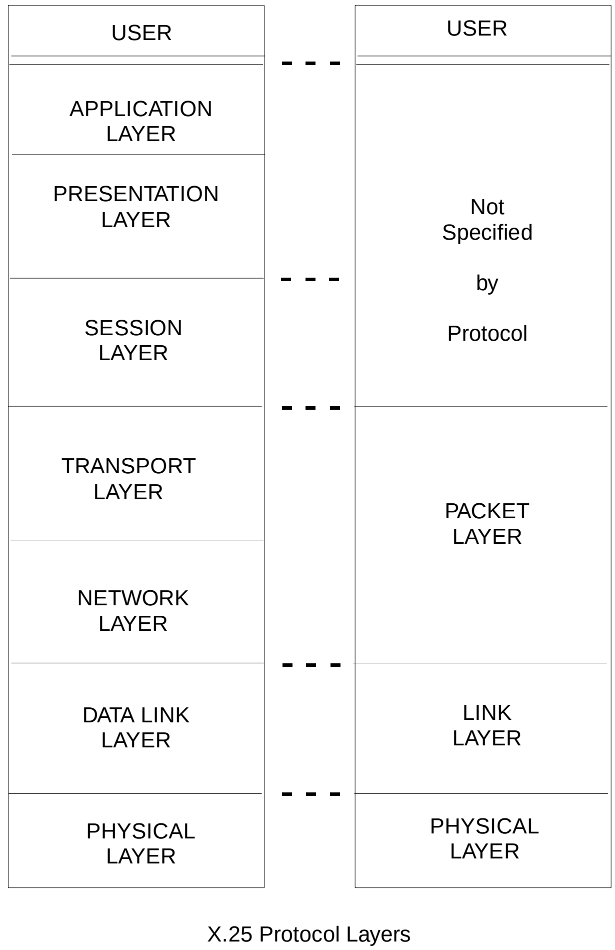


Figure 2-5. X.25 Packet-Switching Protocol

International Telecommunications Union - Telecommunications, Specifications Sector (ITU-T) Interface

X.25 Packet-Switching Protocol

The ITU-T (formerly CCITT) is one of three divisions of the International Telecommunications Union, an agency of the United Nations. The standards set by the ITU-T generally deal with public networks. Two series of standards or recommendations specifically deal with data transmission:

- The V-series provides recommendations for data transmission over analog or voice telephone networks.
- The X-series provides recommendations for data transmission over digital networks.

The V-series includes the V.10, V.11, V.24, V.28, and V.35. Also, V.26, V.27, and V.28 are modem recommendations for 2400, 4800, and 9600 bps, respectively.

V.10 and V.11 are the equivalent to the EIA RS-423 and RS-422.

V.24 provides definitions for all interchange circuits crossing the DTE/DCE interface.

V.28 defines a set of electrical characteristics compatible with EIA RS-232D.

V.35 provides the constant current interface for 48 kbps operation.

The X.25 protocol is the ITU-T recommendation for implementing International Standards Organizations Reference Model of Open Systems Interconnection. This is the international model for packet-switching networks and is a bit-oriented, layered-type protocol. The transport, network, data link, and physical layers (levels) are defined functionally by the ITU-T.

The X.25 protocol specifies network requirements and procedures to provide the user interface for a packet-switching network. Typically, users generate low-speed asynchronous data. The X.25 software segments this data into packets, adds framing and routing information, and queues the packets into a buffer memory. User data packets, along with the added framing bits, are then transmitted over high-speed carriers. This permits efficient and dynamic sharing of these high-speed data links.

The X.25 protocol provides the communications links between multiple APs.

Wideband Switching

Wideband Switching provides a range of data transmission speeds (from 128 to 1984 kbps for E1 services, 128 to 1536 kbps for T1 services). The higher transmission rates are needed for applications such as video conferencing, data backup, scheduled batch processing, and primary data connectivity.

Data Management Features

The following features are associated with Data Management:

- Administered Connections
- Alphanumeric Dialing
- Data Call Setup [including Default Dialing and Alphanumeric Dialing]
- Data Hot Line
- Data-Only Off-Premises Extensions
- Data Privacy
- Data Restriction
- Default Dialing
- Digital Multiplexed Interface
- DS1/E1 Tie Trunk Service
- EIA Interface
- Information System Network (ISN) Interface
- Modem Pooling
- PC Interface
- PC/PBX Connection
- Uniform Call Distribution (See Direct Department Dialing [a VOICE Management Feature])
- Wideband Switching
- World Class Core BRI

Network Services

Network Services allows a group of switches (consisting of DEFINITY Generic 1, Generic 2, Generic 3, System 75 and System 85, and/or other systems) to be configured to meet the communications needs of a medium- to large-size corporation. Possible arrangements include an "Electronic Tandem Network (ETN)", "Distributed Communications System (DCS)", and Main/Satellite/Tributary. Each is briefly described in this chapter.

Do not assume that the system has any capabilities other than those explicitly stated herein. Refer to the *System 75/85 AT&T Network and Data Services Reference* manual, 555-025-201, for differences between this system and other AT&T systems.

Network Services Features

The following features are associated with Network Services:

- AAR/ARS Partitioning
- ARS to AAR Conversion
- Automatic Alternate Routing
- Automatic Circuit Assurance
- Automatic Route Selection
- DCS Alphanumeric Display for Terminals
- DCS Attendant Control of Trunk Group Access
- DCS Attendant Direct Trunk Group Selection
- DCS Attendant Display
- DCS Automatic Callback
- DCS Automatic Circuit Assurance
- DCS Busy Verification of Terminals and Trunks
- DCS Call Coverage
- DCS Call Forwarding All Calls
- DCS Call Waiting
- DCS Distinctive Ringing
- DCS Leave Word Calling
- DCS Multi-Appearance Conference/Transfer
- DCS Trunk Group Busy/Warning Indication
- Non-Facility Associated Signaling and D-channel Backup (G3i)
- Facility Restriction Levels and Traveling Class Marks

- Forced Entry of CDR Account Codes
- Generalized Route Selection
- Integrated Services Digital Network — Primary Rate Interface
- Network Access — Private
- Network Access — Public
- Off-Premises Station
- QSIG Global Networking
- Restriction — Toll
- Subnet Trunking
- Time of Day Routing
- Uniform Dial Plan

Private Network Configurations

A private network is a configuration of trunk and switching facilities dedicated to the use of a business or organization. It may have as few as two switches or it may have hundreds of switches located throughout the world. Although they normally serve moderate to heavy calling between locations, the following configurations make it possible for organizations of all sizes to realize the benefits of a private network.

- "Electronic Tandem Network (ETN)" — Serves the needs of customers with many locations in a large geographic area. This configuration provides for calling between locations without accessing toll facilities.
- "Distributed Communications System (DCS)" — Serves the needs of customers with several locations in a small or large geographic area. A Distributed Communications System (DCS) appears as a single switch with respect to certain features.
- "Main/Satellite/Tributary" — Serves the needs of customers with a few locations in a small geographic area.

The system also can be used within a Tandem Tie Trunk Network (TTTN). A TTTN is a nonhierarchical network of tie trunks interconnecting three or more switches. User dialing into each switch in the call's path is required. That is, the user at one switch dials the trunk access code for a tie trunk group to another switch, receives dial tone from that switch, and then dials another trunk access code to reach another switch. When dial tone is received from the final (desired) switch, the user dials the desired extension number.

Electronic Tandem Network (ETN)

An Electronic Tandem Network (ETN) is a hierarchical network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations. An ETN consists of tandem switches, the intertandem tie trunks that interconnect them, the access or bypass tie trunks from a tandem switch to a main switch, and the capability to control call routing over these facilities. A Main/Satellite/Tributary configuration can be served by an ETN or a DCS can also be part of an ETN.

The system can serve as an ETN tandem switch.

Within an ETN each location is identified by a unique private network office code. With G3i, this private network office code may be of the form RN, RNX, RX, XX, RNX, RXX, XXX, and RNXX, depending on administration (R = digits 2 through 9, N = digits 2 through 9, and X = digits 0 through 9). After accessing the ETN, the user simply dials the private network office code plus the desired extension number, for a total of ten digits or less.

Private network office codes are unique within an ETN. Private network office codes are assigned when the ETN is established. When Direct Inward Dialing (DID) is provided by the local central office, the extension numbers (last four digits of the number) will match. Network Inward Dialing (NID) is the ETN equivalent of DID and can be provided without DID.

The software program that controls call routing over an ETN is called Automatic Alternate Routing (AAR). AAR not only determines the route for a call, but, through the Facilities Restriction Level (FRL) function, defines up to eight levels of calling privileges for users of the ETN. Another function of AAR, Subnet Trunking, can convert an on-network number to a public network or international number. This function is useful when all on-network routes are busy or are not provided. The details of Automatic Alternate Routing, Facilities Restriction Level, and Subnet Trunking are provided in this chapter.

AAR digit conversion can be used to convert private network numbers to other private network numbers or public network numbers. This allows the system to steer some AAR calls to other switches in the private network or, by changing specific dialed digits to a public network number, eventually route some calls via ARS. Also, unauthorized private network calls can be routed to an attendant or receive intercept treatment. Details on AAR digit conversion can be found in the Automatic Alternate Routing feature description.

Distributed Communications System (DCS)

A Distributed Communication System (DCS) is a cluster of private communications switches (nodes) interconnected among several geographic locations. These switches can be either a DEFINITY Generic 1, DEFINITY Generic 2, DEFINITY Generic 3, System 75, System 85, or DIMENSION PBX.

Refer to the System Parameters table In Appendix A for the node limitations of each system type.

An attribute of a DCS configuration that distinguishes it from other networks is that it appears as a single switch with respect to certain features. This provides simplified dialing procedures between locations, as well as the convenience of using some of the system's features between locations. DCS is particularly useful if there is frequent interlocation calling.

Each DCS node is connected with every other DCS node by tie trunks or ISDN-PRI trunks (DEFINITY Generic 3) for voice communications and data links that send and receive control and feature information. However, each DCS node does not have to be directly connected to every other node. Communication may be through a DCS tandem node. The data links and voice channels may be directly between nodes or may pass through a tandem node. Nodes that cannot serve as a tandem node (that is, those that cannot receive information from one node and pass it on to another node) are called endpoints (or endpoint nodes). Nodes that can pass information are simply referred to as nodes. DEFINITY Generic 3i can serve as either an endpoint node or a regular (tandem) node. Figure 2-6 shows a typical DCS configuration.

A DCS can consist of all endpoints. That is, each node in the DCS may be directly connected by data links and voice channels with every other node in the DCS.

Some of the applications of the DCS configuration are as follows:

- In a ‘‘campus environment’’ that has two or more separate buildings and the nodes are connected by local cable.
- In a larger area such as a city, several states, or even the entire country, where the nodes are separated by distances too great for local cable and may be connected to different central offices.

A DCS has the property of ‘‘transparency’’ with respect to inside calling and some features. Transparency is the ability of the system, from the user's standpoint, to operate across several nodes in the same way it does at the local node. This allows users to dial from any terminal to any other terminal within the DCS without regard for which nodes are involved. Likewise, transparency allows certain voice features to be used across nodes.

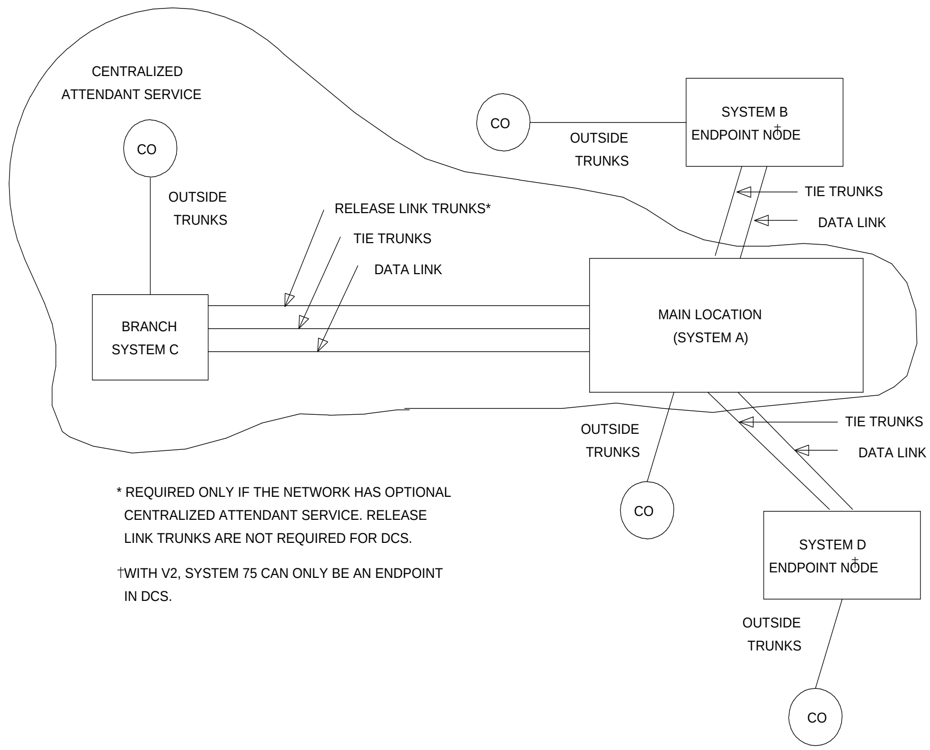


Figure 2-6. Typical Distributed Communications System

Some voice features have transparency in a DCS configuration. The following voice features have unique aspects in a DCS environment and are described in detail in this book.

- DCS Alphanumeric Display for Terminals
- DCS Attendant Call Waiting (described under DCS Call Waiting)
- DCS Attendant Control of Trunk Group Access
- DCS Attendant Direct Trunk Group Selection
- DCS Attendant Display
- DCS Automatic Callback
- DCS Automatic Circuit Assurance
- DCS Busy Verification of Terminals and Trunks
- DCS Call Coverage

- DCS Call Forwarding All Calls
- DCS Call Waiting — Termination (described under DCS Call Waiting)
- DCS Distinctive Ringing
- DCS Leave Word Calling
- DCS Multi-Appearance Conference/Transfer
- DCS Priority Calling (described under DCS Call Waiting)
- DCS Trunk Group Busy/Warning Indication.
- Enhanced DCS (described under Enhanced DCS).

Abbreviated Dialing and Last Number Dialed also have transparency in a DCS configuration. These features operate the same in a DCS as they do at a single switch.

A DCS cluster can consist of up to 64 nodes. Since AUDIX and the Call Management System (CMS) and ISDN-PRI each require the same data link facilities as a node, each of these included in the system reduces the number of available data links, which, depending on the system configuration, may reduce the maximum number of nodes.

Use of the DCS over the ISDN-PRI D-channel feature on the other hand, may increase the maximum number of nodes. (Note that on G3r, ISDN-PRI does not require the same data link facilities as a node.)

DCS Message Hopping lets a DCS message route through an intermediate node without tandeming an associated trunk call. This is accomplished through the use of hop channels. The system provides Message Hopping through up to two hops.

DCS transparency is more restricted when the tandem node is an Enhanced DIMENSION PBX or a System 85 Release 2 Version 1 than when it is a System 85 Release 2 Version 2, or later, or a DEFINITY Generic 2.1, or later. (See the DCS Alphanumeric Display for Terminals and DCS Leave Word Calling features.)

Certain feature capabilities are unique to a particular type of node (for example, a DEFINITY Generic 3 endpoint node). Therefore, a detailed feature description should be consulted for each type of node.

The Centralized Attendant Service (CAS) feature can be used as an advantage in DCS networks where all attendants are at one node. CAS reduces traffic volume on interconnecting tie trunks caused by incoming attendant-seeking calls at the endpoint nodes. DEFINITY Generic 3 can serve as the main location for CAS attendants. Centralized Attendant Service capabilities are given in detail in this manual.

With DCS Call Coverage (G3V4 and later releases), calls to an extension on one system can be covered by extensions administered as coverage points on remote systems. DCS Call Coverage provides transparency across systems for the Call Coverage feature.

Main/Satellite/Tributary

Figure 2-7 shows a Main/Satellite/Tributary configuration. It can function independently or serve as an ETN access arrangement. For a Main/Satellite configuration, attendant positions and public network trunk facilities are concentrated at the Main, and calls to or from satellite locations pass through the Main. To a caller outside the Main/Satellite complex, the system appears to be a single switch with one Listed Directory Number. This is accomplished with the optional Uniform Dial Plan software.

Tributary and Satellite locations are similar except that a Tributary has one or more attendant positions and its own Listed Directory Number.

DEFINITY Generic 3 can serve as a Main, Satellite, or Tributary.

A small business can start with a single Main/Satellite or Main/Tributary complex and add trunk and switching facilities as the business grows. In this situation, tie trunks connect the main locations within an urban area and intercity traffic is routed via the public network. This arrangement favors a medium-size organization or one that has small isolated locations where the intercity traffic is too small to justify the cost of tie trunks.

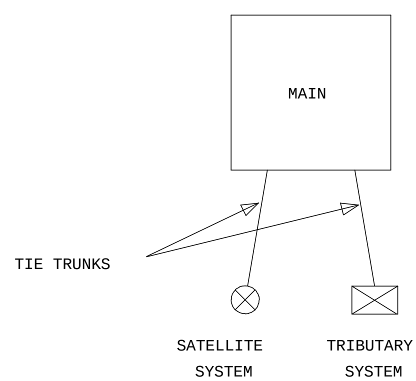


Figure 2-7. Main/Satellite/Tributary Configuration

Trunking

Trunking is the use of communications links to interconnect two switching systems, such as connecting the switch to a local central office or to another switch. These links, called trunks, can be grouped together in Trunk Groups when all the trunks in the group perform the same function. This grouping simplifies administration since the required service characteristics (parameters) are assigned to the group rather than to each trunk. Grouping also simplifies call processing. Calls requiring a trunk are routed to the appropriate trunk group and an idle trunk, if available, is selected from the group.

There are analog trunks and digital type trunks. The type of signal is specified for each trunk in the list below. The following types of trunk groups can be used with the system:

- Auxiliary — Provides internal trunk applications for features such as Loudspeaker Paging and Music-on-Hold. This is an analog signal.
- CO — Provides a link with the local Central Office (CO) for Direct Outward Dialing (DOD) and manually completed incoming calls, not Direct Inward Dialing (DID) calls. The signals may be analog or digital
- Direct Inward Dialing (DID) — Provides a link with the local CO that is only usable for incoming calls and allows the CO to send digits to the PBX so the call can be routed to a particular station. The signal can be analog or digital
- Direct Inward/Outward Dialing (DIOD) — Provides a link with the local CO that allows digits to be sent in either direction for incoming and outgoing calls. The signal can be either analog or digital
- DS1/E1 Tie Trunk — Provides for two types of digital tie trunk interfaces: Voice-Grade DS1/E1 and Alternate Voice/Data (AVD) DS1/E1 tie trunks. The Voice-Grade DS1 tie trunks are an alternative to four-wire analog E&M tie trunks and may be used to interface with other properly-equipped switching systems. AVD DS1/E1 tie trunks permit alternate voice and data calling between a System 75, DEFINITY Generic 1, DEFINITY Generic 3 and a System 85 or DEFINITY Generic 2. DS1/E1 tie trunks also can be used with Release Link trunks for Centralized Attendant Service, and can be used with AT&T unbanded telecommunications service. (digital signal)
- FX — Provides a link with a CO other than the local CO. (analog or digital signal)
- ISDN-PRI — Provides end-to-end digital connectivity and supports a wide range of voice and non-voice services. Calls to a variety of switched nodal services and calls destined for different inter-exchange carriers can be processed. (digital signal)
- Tie and Release Link — Provide a link with another private switching system for calls between the systems. Release link trunks are used only with Centralized Attendant Service. The signal can be analog or digital. Tie trunks are used on calls to or from the following:

- — a Private Branch Exchange (PBX)
- — an ETN switch
- — an EPSCS or Common Control Switching Arrangement (CCSA) office
- — MEGACOM Service (AT&T unbanded long distance service.)

- WATS —Provides access to a portion of the DDD network via dedicated trunks to a WATS serving office. Each WATS trunk terminates as a WATS line in the WATS serving office. The trunks used are always one-way outgoing. Outgoing WATS calls to various geographic areas are made on an attendant-handled basis or a station direct-outward-dial basis. Signals can be analog or digital.

Tie trunks used with the system are administered as either internal or external. The internal or external designation controls the type of ringing (which is administrable) received at a voice terminal when an incoming tie trunk call arrives and controls the routing of the call if it is redirected through the Call Coverage feature. The default ringing pattern is the following:

- Incoming internal tie trunk calls cause internal-call ringing and will redirect according to the redirection criteria administered for internal calls.
- Incoming external tie trunk calls cause external-call ringing and redirect according to the redirection criteria administered for external calls.

The number of bursts for internal/external/priority calls is administrable in G3i-Global and G3V2 and later releases. The "Call Coverage" feature interaction with Tie Trunks is described in detail in Chapter 3 of this manual.

Selection of the trunk group to be used for a given call is determined by digit translation on the trunk access code by the AAR/ARS routing tables. Assuming an idle trunk in the selected group is found, a seizure signal (service request) is sent to the distant switch. If the distant switch requires the called number, a start dial signal is normally returned to the calling switch, indicating readiness to accept digit transmission.

The start dial signal(s) used is dictated by the serving FX office, WATS office, or local CO. For interconnection with other private switching systems, the System Manager may select the start dial signal(s) to be used.

‘‘Trunk type’’ refers to the physical design of a trunk circuit. Trunk type and the start dial signal are often used interchangeably, although trunk type is a more accurate term. A brief description of the available trunk types follows:

- Ground Start — A ground signal is sent over the trunk ring lead and is received over the trunk tip lead.
- Loop Start — A closure signal is sent through the loop formed by the trunk leads.

- Immediate Start — No start dial signals are used. On outgoing calls, the system waits at least 80 milliseconds (a configurable amount of time) after sending the seizure signal before sending the digits required at the distant switch. This gives the distant switch enough time to attach a digit receiver to the call.
- Wink Start — A momentary signal (wink) is sent to the distant switch when the trunk is ready to receive digits.
- Delay Dial — A steady signal is sent to the distant switch and is removed when the trunk is ready to receive digits.
- Automatic — No dialing is performed. The seizure signal sent or received is sufficient to route the call. The call destination is specified when the trunk group is administered. The destination can be the attendant group or any extension number assigned in the system.

Trunk groups connecting with a WATS office, FX office, or local CO can be ground or loop start. DID trunk groups can be immediate or wink start. Tie trunk groups can be delay dial, wink start, immediate start, or automatic.

Trunk groups can be one-way incoming, one-way outgoing, or two-way. Whether the trunk group is available for incoming, outgoing, or two-way traffic is called direction. A two-way loop-start trunk is subject to glare. Glare occurs when the distant switch is trying to use a given trunk for a call to a DEFINITY PBX at the same time the DEFINITY system is trying to use the same trunk for a call to the distant switch. Incoming calls are not aborted because of glare. The incoming call will complete, and the outgoing call will receive reorder tone (G3i and G3vs/G3s). In G3r and G3V2 and later releases, glare retry handling for two-way trunks is administrable. Queuing at both ends of a two-way trunk group compounds the possibility of glare and is, therefore, not recommended.

Each non-DCS outgoing and two-way trunk group can have a queue. If all trunks in the group are busy, the call waits in the queue until a trunk becomes idle. The queue length, which is the number of calls waiting, may be from one to 100. A queue length of 0 (zero) indicates no queue has been established. This information is entered on the trunk group form when the trunk group is administered.

Dual Tone Multifrequency (DTMF) (touch-tone) signaling or rotary dial (dial-pulse) signaling can be used between switches. The system can send or receive either type of signaling required by the distant switch as well as ISDN-PRI and R2-MFC and others.

An incoming trunk call to the system can be connected to another trunk, a voice terminal, an attendant console, or an announcement. When the call is answered, ‘‘an answer supervision’’ signal is sent to the distant public network switching office. This signal initiates the recording of the call details normally used for charging. Any CO call routed outward is deemed ‘‘answered’’ 10 seconds (system default; however, this may be administered as higher or lower on the trunk group form) after the last digit is dialed. Tie trunk calls are deemed ‘‘answered’’ when answer supervision is returned from the far end or when answer supervision time-out expires. Also, if there is a trunk incoming from one of the previously listed offices on a call of this type, then answer supervision is sent to that office. An incoming call to a Direct Department Calling (DDC) or Uniform Call Distribution (UCD) recorded delay announcement is deemed ‘‘answered’’ when the calling party is connected to the announcement. Other types of announcements, such as unassigned number announcements, are treated as an unanswered call.

System Management

System Management provides the capabilities to control and maintain the system and also provides system usage reports to help determine if the system is being used as intended. In short, System Management allows the System Manager to establish the system, monitor its use, and make additions and/or changes as necessary. System management devices are the Generic 3 Management Terminal (G3-MT) or the Generic 3 Management Adjunct (G3-MA). The G3-MA system management device is an enhanced PC-based administration system.

WARNING:
Secure your system from unauthorized access by following the guidelines suggested in the GBCS Products Security Handbook, 555-025-600, and in the "Security Measures" section, Chapter 1.

System Management features and functions are described in this chapter. Functions are more fully described in the following documents.

- *DEFINITY Communications System Generic 3 V4 Implementation,* 555-230-655
- *DEFINITY Communications System Generic 3 V2 Traffic Reports,* 555-230-511
- *DEFINITY Communications System Generic 3r Maintenance*, 555-230-105

Changes made to system translations are effected only at the single system for which the changes were made. If a system is part of a network, changes may need to be made at more than one system to effect the desired changes to the network. Similarly, changes intended for only a single system could affect the network. Therefore, the System Manager must understand the effect on the network before making any changes.

System Management Features

The following features are associated with System Management:

- Administration
- Advice of Charge
- Call Detail Recording (CDR)
- Customer-Provided Equipment (CPE) Alarm
- Facility Test Calls
- Move Agent From CMS
- Recent Change History
- Report Scheduler and System Printer
- Security Violation Notification

- System Measurements
- System Status Report

System Administration

Allows the user to implement (initialize) and administer all the terminal and system features and system parameters. System Administration allows the following:

- Initializing the system
- Managing system, voice terminal, and data terminal features on a day-to-day basis
- Performing system back-up procedures
- Monitoring, detecting, and determining system performance
- Maintaining system security

System administration and maintenance are performed at the G3-MT or G3-MA, a Remote Administration terminal, or AT&T location. The G3-MT and G3-MA are referred to from here on as the administration terminal. The G3r multiple administration feature supports up to eight simultaneous system management sessions. Up to five of these sessions may be performing administration updates simultaneously. Up to five of these sessions may be executing a maintenance command simultaneously.

The administration terminal can be any of the following:

- 715 BCT
- G3-MA
- G3-MT
- MS-DOS compatible PC with 4410 emulation software

The administration terminal must be located within 50 feet of the system cabinet and must be connected directly to the "terminal" or "duplication option terminal" connected on the Switch Processing Element (SPE). The administration terminal consists of a video display and keyboard that allow a System Manager to input system commands and translations. The administration terminal is first used to initialize the system. After initialization, the administration terminal is used to reconfigure translations and to monitor system performance. Remote AT&T service locations have access to the same administration capabilities as the local administration terminal.

G3V3 and later releases provide enhanced login/password security by adding a security feature that allows the user to define their own logins/passwords and to specify a set of commands for each login. For complete instructions about logging in and password administration, see the AT&T document *DEFINITY Communications System Generic 3 V4 Implementation* 555-230-655.

Remote Administration

Allows the system to be administered from a remote terminal located either on or off the customer's premises. A terminal located more than 50 feet from the system cabinet is considered remote. A remote administration terminal can be on the same premises as the local administration terminal or it can be off-premises. The remote terminal performs the same functions as the local administration terminal.

The VT 220, 610 BCT, 615 MT BCT, 715 BCT, 4410 terminal, or 4425 terminal may be used as either an on-premises or off-premises remote terminal. If the remote terminal is a 4410 terminal, VT220, 513 BCT, 610 BCT, 615 MT, or 715 BCT, it must be connected to the system through a PDM, 7400A data module, 7400B data module, or Data Line circuit pack. If a 4425 terminal (models that include a built-in modem) is used as a remote terminal, a PDM, 7400A is not required. See the *DEFINITY Communications System Generic 1 and Generic 3 Installation and Test* manual, 555-230-104, for additional information.

Technical Service Center (TSC)

The TSC is an organization of AT&T service personnel who provide system administration and maintenance from a remote location.

Personnel at the TSC access the system and perform administrative tasks assigned to the System Manager. The administrative commands used by the System Manager are also available to the TSC personnel. TSC personnel can also execute maintenance routines.

During system access, the TSC personnel automatically receive major and minor alarm notifications from the system. When an alarm is received, TSC personnel can access the system and perform the following tasks:

- Clear errors
- Display alarms
- Display errors
- Download a copy of the system tape
- Perform any required administration
- Receive backup translations for the system
- Set time and date
- Test and busyout circuit packs, voice terminals, and trunks
- Dispatch field technicians when required
- Notify customer of disposition and status of alarms.

Hospitality Services

The Hospitality Services features meet the lodging industry's need to provide services for their guests. The basic feature set is included in the basic voice application software and is sometimes referred to as the hotel/motel feature software package.

Hospitality Services Features

The following features are associated with Hospitality Services:

- Automatic Wakeup
- Do Not Disturb
- Names Registration
- Property Management System Interface
  - Check-In/Check-Out
  - Housekeeping Status
  - Controlled Restriction
  - Guest Information Input/Change
  - Room Change/Room Swap
  - Message Waiting Notification

Call Center Services

The Call Center Services features support industries such as airlines, travel agencies, and catalogs that have a large number of similar incoming and/or outgoing calls. These features can provide balanced call distribution to a large group of voice terminals.

Call Center Services Features

The following features are associated with Call Center Services:

- Abandoned Call Search
- Agent Call Handling
  - — Stroke Counts
  - — Call Work Codes
  - — Forced Entry of Stroke Counts and Call Work Codes
- Automatic Call Distribution (ACD)
- Basic Call Management System (BCMS)
- Call Prompting
- Call Vectoring
- CallVisor (ASAI)
- Expert Agent Selection (EAS)
- Flexible Billing
- Inbound Call Management
- Intraflow and Interflow
- Look Ahead Interflow
- Malicious Call Trace (MCT)
- Move Agent From CMS (See Appendix A, "System Parameters")
- Multiple Call Handling
- PC Application Software Translation Exchange (PASTE)
- Queue Status Indications
- Redirection on No Answer (RONA)
- Service Observing
- Voice Response Integration (VRI)
- VuStats

Feature Descriptions

3

Overview

This chapter defines the DEFINITY Communications System features. The features are arranged in alphabetical order, regardless of the functional area to which they apply. The information for each feature is usually presented under six headings: Feature Availability, Description, Considerations, Interactions, Administration, and Hardware and Software Requirements.

- Feature Availability: Defines the release in which the feature is available.
- Description : Defines the feature, describes what it does for the user or how it serves the system, and briefly describes how it is used.
- Considerations: Discusses the applications and benefits of the feature and any other factors to be considered when the feature is used.
- Interactions: Lists and briefly discusses other features that may significantly affect the feature. Interacting features depend on each other; one of the features must be provided if the other one is. Other features may enhance each other; the features, in combination, provide improved service to the user. Some features cannot coexist; one of the features cannot be provided if the other one is. Finally, one feature can affect another; the normal operation of one feature modifies, or is modified by, the normal operation of the other feature.
- Administration: States whether or not administration is required, how the feature is administered, who administers the feature, and lists items requiring administration.
- Hardware and Software Requirements: Lists any additional hardware and/or software requirements for the feature.

AAR/ARS Partitioning

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides for the Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) services to be partitioned among as many as eight different groups of users within a single DEFINITY Generic 3 switch. This provides individual routing treatment for the different groups of users.

A partitioned user group consists of those users who are grouped together and share the same Partition Group Number (PGN). The PGN is not a restriction, but a means used to indicate the choice of routing tables to be used on a particular call. Each Class of Restriction (COR) is assigned a specific PGN or Time of Day specification. Different CORs may be assigned the same PGN. Therefore, it is possible for members of the same partitioned user group to have different CORs.

When the "AAR/ARS Partitioning" feature is used in a hotel/motel or a hospital environment, different facilities access is provided through ARS for guest/patient voice terminals and administrative staff member voice terminals. For example, within a hotel or motel, the guests and staff voice terminals might be partitioned into two user groups. When a guest places an interstate call, the guest user group’s ARS tables may specify that the call be routed using AT&T QUOTE Service, a telephone billing information system that is used to bill back or allocate long-distance charges. A similar call placed by a staff member might be routed over a Direct Distance Dialing (DDD) trunk.

All partitioned user groups share the same pool of Routing Patterns. (See the "Automatic Alternate Routing (AAR)" and "Automatic Route Selection (ARS)" features for further explanations on routing.) The translation tables that specify the Routing Pattern number are unique for each partitioned user group. Routing Patterns may be shared among the user groups or may be dedicated to a particular user group. Once a user activates the "Automatic Route Selection (ARS)" or "Automatic Alternate Routing (AAR)" feature and dials enough digits for the system to search for the Routing Pattern, the PGN of the originator’s COR is used to select the table to look up the Routing Pattern.

Users of AAR/ARS Partitioning include the following:

- Single-Line Voice Terminals
- Multi-Appearance Voice terminals
- Attendants
- Remote Access Users

- Data Endpoints
- Incoming Tie Trunks
- Other Trunks, used when calls are forwarded to an off-premises number

Considerations

With AAR/ARS Partitioning, different groups of users within the same system can receive individual routing treatment. For example, the following types of situations may require AAR/ARS Partitioning:

- Groups of users with different routing preferences for calls to a given area due to special billing needs
- Groups of users who wish to have dedicated use of a particular network facility
- Groups of users in different businesses in one or more buildings serviced by a single system
- Data users who require special facility types on outgoing calls

Partition user groups are only used with UDP, AAR and ARS. There is no capability to access the partitioned user groups directly. Operation of the groups is completely transparent.

Interactions

The following features interact with the "AAR/ARS Partitioning" feature.

- Bridged Call Appearance

  If a Bridged Call Appearance is used for an AAR or ARS call, the system will use the PGN of the bridged principal's extension instead of the PGN of the originating user's extension.

- Call Detail Recording (CDR)

  The PGN used to route the call is not recorded in CDR.

- Call Forwarding All Calls

  If a call terminates at a voice terminal that has Call Forwarding All Calls activated and the forwarded-to number uses AAR or ARS, the COR of the calling user is used to look up the PGN for the call.

- DCS

  The "AAR/ARS Partitioning" feature can cause different Routing Patterns to be used on DCS calls. For example, one user's Routing Pattern may specify a DCS trunk group as a member of the pattern. A user of a second PGN may use a different Routing Pattern that does not specify the DCS trunk group. In this case, one user has DCS feature transparency and the second user does not.

When a call routes over a DCS trunk, no PGN information is sent to the far-end PBX. Thus, the far-end PBX only will be capable of using the incoming trunk's PGN to route the call.

- Remote Access

  If a Remote Access user activates ARS, the COR assigned to the barrier code dialed (or the Authorization Code, if required) is used to select the PGN for the call.

- Straightforward Outward Completion and Through Dialing

  If the attendant assists or extends a call for a user and activates ARS, the attendant's COR is used to select the PGN for the call.

- Uniform Dial Plan (UDP)

  Since UDP calls expand the dialed digits into seven-digit numbers and then use AAR to route the call, these calls will make use of partitioning. Once the call begins to be handled by AAR, the user's active COR will be used to identify the proper PGN to handle the call.

Administration

AAR/ARS Partitioning is administered by the System Manager. The following items require administration:

- Different Digit Analysis tables must be administered for each partitioned user group.
- A PGN must be assigned to each COR table. Up to eight PGNs can be used. If the "Time of Day Routing" feature is assigned, a Time of Day Plan Number is assigned to the COR instead of the PGN.

Hardware and Software Requirements

No additional hardware or software is required.

Abandoned Call Search

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides identification of abandoned calls for CO offices that do not provide timely disconnect supervision.

Before an incoming Automatic Call Distribution (ACD) split rings the hunt group member or agent, the system checks to make sure the calling party has not abandoned the call (hung up). If the calling party has abandoned the call, the call does not ring the hunt group member or agent. Abandoned Call Search adds an overhead of up to one second to each call delivered to an agent.

To see if the calling party has abandoned the call, after the call has been abandoned by an announcement, the system must determine if the calling party is still connected to the ground-start trunk at the central office (CO). To do this, the system flashes (opens the tip-ring loop for 150 to 200 ms) the CO end of the trunk. If the calling party is still connected, the CO will not respond. If the calling party has hung up on the call, the CO will send a disconnect signal within 700 to 800 ms. The system interprets this as an abandoned call, releases the trunk, and the call does not ring the hunt group member or agent.

Outside of the US, a flash of this duration may be responded to differently. Please see the "Trunk Flash" feature for more information.

After it is administered for a trunk group, this feature is performed automatically by the system. No operation is required by system users.

Considerations

Abandoned Call Search is suitable only for older COs that do not provide timely answer supervision. Most COs provide timely disconnect supervision, and therefore do not require the "Abandoned Call Search" feature. Some older COs can take as long as two minutes to notify the PBX of a disconnect and, thus, require the PBX to determine, within one second, whether the call has been abandoned, prior to extending the call. Even with Abandoned Call Search or disconnect supervision, a small probability exists that a call will be extended to the destination hunt group after the caller has hung up. Abandoned Call Search and disconnect supervision significantly reduce that probability.

Abandoned Call Search works only with ground-start analog trunks.

Abandoned Call Search allows agents and hunt group members to answer more calls because time is not wasted on abandoned calls. In addition, call handling statistics generated by the CMS are more accurate because the CMS knows when a call is abandoned.

Interactions

None.

Administration

Abandoned Call Search is administered on a per trunk group basis by the System Manager. Each ground start CO, FX, and WATS trunk group is administered as either having Abandoned Call Search or not having it.

Hardware and Software Requirements

For G3i-Global, Abandoned Call Search requires the use of a TN747B CO Trunk circuit pack. The same requirement is in effect if the serving CO is a No.1 or No. 5 Crossbar switch.

No additional software is required.

Abbreviated Dialing (AD)

Feature Availability

Abbreviated Dialing is available with all Generic 3 releases. Enhanced Abbreviated Dialing (also known as Enhanced Number List) is optional with all releases except G3vs/G3s ABP. Enhanced Abbreviated Dialing is not available with G3vs/G3s ABP. Automatic Dialing Buttons and designated user programming of group lists are available with G3V4 and later releases.

Description

Provides lists of stored numbers that can be accessed to place local, long-distance, and international calls; to activate features; or to access remote computer equipment. Stored numbers can be accessed by voice terminal users and data terminal users. Certain stored numbers can also be accessed by attendants.

Automatic Dialing Buttons, available with G3V4 and later releases, allow users direct access to a designated number that is not stored on an Abbreviated Dialing list. See "Access Options" later in this section for more information about Automatic Dialing Buttons.

List Types

Desired numbers are stored in any of four types of lists. Each stored number is one list entry. To use Abbreviated Dialing, a user accesses the appropriate list by dialing an access code, and then dials the one-, two-, three or four-digit list entry number where the desired number is stored. The number is then dialed automatically by the system. For a frequently called number, the list and list entry number can be stored on an Abbreviated Dialing button. In this case, simply pressing the button places the call.

The following section describes the types of Abbreviated Dialing lists. The number of lists per system and the number of entries per list is determined by the type and version of the switch. See "System Hardware and Software Capacity Limits" on page A-12 for more information.

- Personal Number Lists

  Allow voice and data terminal users to have a personal set of stored numbers. A user can have up to three Personal Number Lists. The user, or the System Manager, programs the Personal Number Lists. The System Manager determines the users that are allowed to have a personal list and the size of each list.

  The Personal Number Lists are created automatically when the capability of accessing a personal list is administered for the individual station.

- Group Number Lists

  Allow access by a group of users, such as purchasing or personnel departments, who frequently dial the same numbers. An individual user can access up to three specific Group Number Lists, as set by the System Manager.

  The Group Number Lists are administered by the System Manager. G3V4 and later releases also provide for the administration of a designated user (extension number) who can program a Group Number List. The designated user for each list is specified on the “Abbreviated Dialing Group List” form.

- System Number List

  Can contain any number or dial access code. The System Manager programs the System Number List and sets which users can access the list. One System Number List is allowed per system.

- Enhanced Number List

  One Enhanced Number List is allowed per system in addition to the System Number List. The Enhanced Number List can contain any number or dial access code. The System Manager programs the Enhanced Number List and sets which users can access the list.

> **NOTE:**
> The enhanced number list capability is an optional feature which, if desired, must be purchased separately. The Enhanced Number List option is called the "Enhanced Abbreviated Dialing (EAD)" feature.

List Entries

The following sections describe the numbering schemes used to select a list entry once an Abbreviated dialing list is accessed.

Personal Number List Entries

For Personal Number Lists administered for 5 or 10 entries, the user dials a single digit to access a list entry number. Entries are numbered 1 through 9 and 0 (list entry 10). For example, to access entry 6 the user dials ***6***, to access entry 10 the user dials ***0***.

For Personal Number Lists administered for more than 10 entries (G3V4 and later releases only), the user dials a two-digit number to access a list entry. For example, to access entry 6 the user dials ***06***, to access entry 100 the user dials ***00***.

Group Number and System Number List Entries

G3V4 provides the option of selecting between two Group Number and System Number List entry numbering schemes. The type of numbering scheme used is administered on the “System-Parameters Customer-Options” form. The first numbering scheme corresponds to the one used in G3V3 and earlier releases. The second numbering scheme corresponds to the one used in G2 switches.

Table 3-1. Group Number and System Number List Entry Numbering Schemes

List Entry Number	User Dials (Scheme 1)	User Dials (Scheme 2)
1	11	01
2	12	02
.	..	..
.	..	..
.	..	..
10	20	10
11	21	11
..	..	..
89	99	89
90	00	90
91	01	91
..	..	..
..	..	..
100	10	00

Enhanced Number List

List entries for the Enhanced Number list are numbered 000 through 999 in G3V3 and earlier releases. With G3V4 and later releases, the system can be administered to have list entry numbers of either 000 through 999 or 0000 through 9999.

List Assignments and Designations

Each extension number can be assigned up to three Abbreviated Dialing Lists — List 1, List 2, and List 3. Each of these three lists is designated as being either Personal, Group, System, or Enhanced. The three lists may be any combination of the above as long as there is no more than one System and/or Enhanced List. When a list is designated as being a Group List, the particular number of the Group List is specified (for example, group list 42). Personal Lists must have a group number as well (Personal Lists are designated P1, P2, P3, etc.).

To access Abbreviated Dialing, the user accesses List 1, List 2, or List 3 either by dialing the access code or by using a button programmed with the access code. The access codes for List 1, List 2, and List 3 are the same systemwide. Therefore, it is possible for a System List or a particular Group List to have a different access code at different voice terminals. For example, suppose the feature access codes for List 1 and List 2 are 101 and 102, respectively.

One voice terminal may have List 2 administered as ‘‘group 42.’’ Another voice terminal may have List 1 administered as ‘‘group 42.’’ In this case, the access code for ‘‘group 42’’ is 102 for the first voice terminal and 101 for the second voice terminal.

Privileged Lists

All Group Number Lists, the System Number List, and the Enhanced Number List can be designated as Privileged by the System Manager. Calls automatically dialed from a Privileged List are completed without Class of Restriction or FRL checking. (FRLs are associated with the "Automatic Route Selection (ARS)" and "Automatic Alternate Routing (AAR)" features.) This allows access to selected numbers certain voice terminal users might otherwise be restricted from manually dialing. For example, a voice terminal user may be restricted from making long-distance calls. However, the number of another office location may be long distance. This number could be entered in a list designated as Privileged. The user could then call the office location using Abbreviated Dialing, while still being restricted from making other long-distance calls.

Special Characters

A number stored in an Abbreviated Dialing List can be a combination of numerical digits and special characters. A special character instructs the system to take a different action when dialing reaches the point where the character is stored. Each special character counts as two digits toward the maximum number of digits in a list entry. Refer to the AT&T document *DEFINITY Communications System Generic 3 V4 Implementation,* 555-230-655 for more information. The following special characters can be stored:

- Pause (~p)

  When a Pause is included in a string of stored digits to be outpulsed over a trunk, outpulsing of the digit(s) following the Pause is delayed 1.5 seconds. Outpulsing automatically resumes after expiration of the delay timing.

  The Pause is useful when there is good probability the far-end dial tone (provided by another switch) will be returned within 1.5 seconds. Typical applications include tandem switching through private networks and end-to-end signaling over the public or a private network.

  The pause characters may not operate properly if used on buttons pressed after the call is connected. Use of pause characters on calls where the total number of digits sent exceeds 36 may cause loss of digits.

- Wait (~w)

  When a Wait precedes, or is included in, a string of stored digits to be outpulsed over a trunk, outpulsing of the digit(s) following the Wait is delayed 5 to 25 seconds or until dial tone is detected, whichever occurs first. Outpulsing resumes after the End-Wait signal is received or when delay timing expires.

  The Wait is useful in cases where dial tone delays of variable length and/or network blocking outside the system are frequently experienced. Typical applications include tandem switching through private networks and end-to-end signaling over the public or a private network.

- Indefinite Wait (~W)

  When an Indefinite Wait precedes, or is included in a string, outpulsing of the digits following the Indefinite Wait is delayed until dial tone is detected no matter how long it takes or until the calling party goes on hook. The Indefinite Wait is useful in situations where Dial Tone is frequently delayed for many minutes.

- Mark (~m)

  When a Mark precedes, or is included in a string of stored digits, all digits following the Mark are treated as end-to-end signaling digits to be outpulsed over an outgoing trunk in “Touch-Tone Signal” form even if a dial pulse trunk was used to set up the call. As a typical application, a DTMF data call can be made over a dial pulse trunk (for example, retrieving messages from AUDIX). The mark character should be placed immediately after the extension, before any digits being sent to the answering station.

- Suppress (~s)

  When a Suppress precedes, or is included in a string of stored digits, the system treats all digits following the Suppress the same as any other digits for call setup and digit outpulsing. The Suppress character only affects the display of the stored number. Stored numbers are normally displayed when an alphanumeric display is provided through the "Voice Terminal Display" feature; however, the digits following the Suppress character are not displayed. The display shows the lowercase letter s instead of the stored digits.

The Pause and Wait special characters are needed to delay outpulsing of the initial digits following access of an outgoing trunk if the system does not know when to start outpulsing over a trunk (for example, in Europe). Use of these characters as the very first character could cause calls to be aborted. These characters are used when outpulsing should be delayed until dial tone is returned from a distant point reached through a switched connection outside the system.

Access Options

Abbreviated Dialing numbers can be accessed by any of the following options:

- Abbreviated Dialing-Code (AD Code)

  This option allows users to access a stored number by dialing the "Abbreviated Dialing (AD)" feature access code and a list entry number. Each AD code automatically dials the number stored in the list the user accessed.

- Abbreviated Dialing-Button (AD Button)

  This option allows multi-appearance voice terminal users and attendants to access stored numbers by pressing one or more buttons. Each AD button automatically dials the number stored in the list and the list entry number administered to the button.

  Access to any list and associated list entry number can be programmed in an AD button on a multi-appearance voice terminal. An AD button on an attendant console can be programmed to access a Group List, the System List, or the Enhanced List and associated list entry number.

  The System Manager administers the AD button. If the button is administered to access a number in the user's Personal Number List, the user can change the number assigned to the button. If the number assigned to the button accesses an entry on a Group List either the System Manager or a designated user (G3V4 and later releases) can make the change. If the number assigned to the button accesses an entry on the System List or the Enhanced List only the System Manager can make the change.

- Automatic Dialing Button

  Available with G3V4 and later releases, this option allows users to directly dial numbers by pressing one or more buttons. Each Automatic Dialing button is administered to automatically dial a number that is not stored on any of the Abbreviated Dialing lists. The number can be up to 16 characters long. The Automatic Dialing buttons are programmed by the user or the System Manager.

A separate list, called the 7103A Group Number List, is used only by 7103A Fixed Feature voice terminal users as a group. This list allows button access to stored numbers and can have eight list entries. Any number can be stored in the 7103A Group Number List; however, it is intended primarily for feature access codes. The System Manager programs the 7103A Group Number List.

All users can program their Personal Number List, any AD button that accesses a number from their Personal Number List, and their Automatic Dialing buttons. Only designated users can program an AD button that accesses a number from a Group Number List. Programming is done by dial access or by pressing the PROGRAM button, if assigned.

Programming Personal Lists, Abbreviated Dialing Buttons and Automatic Dialing Buttons

To program an entry in a Personal Number List, the user dials the Program access code or presses the PROGRAM button, then dials the personal list number, the Dial Code number, and the number to be stored (up to 24-digits), and then presses either the # key or the AD button. Confirmation tone is heard when the number is stored. While in the program mode, users can program all Personal Number List entries, if desired. To exit the program mode, the user simply hangs up.

To program an AD button administered to access a particular entry in the Personal Number List, the user dials the Program access code or presses the PROGRAM button, if assigned. The user presses the AD button, dials the desired number (up to 24-digits), and presses the #. Confirmation tone is heard when the number is stored. In the program mode, the user can program as many assigned AD buttons (as well as Automatic Dialing buttons) as desired. To exit the program mode, the user hangs up.

Only the System Manager and multi-appearance voice terminal users can program special characters. Voice terminal users need PAUSE, MARK, WAIT, INDEFINITE WAIT, and SUPPRESS buttons or a Function Entry button to program special characters. Pressing a PAUSE, MARK, WAIT, INDEFINITE WAIT, or SUPPRESS button programs the special character administered to the button. Pressing the AD FUNCTION ENTRY button and then dialing 1, 2, 3, or 4 programs Pause, Wait, Mark, or Suppress respectively. Special characters count as two toward the total number of digits that can be stored in a list entry or button. See "Special Characters" on page 3-10 for additional information.

Programming Group Lists

With G3V4 and later releases, voice terminal users can program the Group Lists to which they are assigned as the designated user. For example, a department secretary may be the designated user for a Group List created for the department. Designated users are assigned (as Program `Ext`) per Group List on the “Abbreviated Dialing Group List” form. Users who attempt to program group lists for which they are not the designated user receive intercept tone.

The method a designated user will use to program a Group List is dependent upon the access codes and buttons administered for the terminal and system. The designated user can:

- Program the Group List using the Program access code or AD Program button (ABR_PROG) in conjunction with an Abbreviated Dialing (ABRV_DIAL) button. In this case, the number to be programmed must be associated with that AD button.

- Program the Group List using the Program access code or AD Program button (ABR_PROG) when an Abbreviated Dialing (ABRV_DIAL) button is not available. In this case, the user can only program lists assigned (as List 1, 2 or 3) to the terminal he or she is using. Other lists for which the user is the designated user cannot be programmed in this way.

  Group Lists can only be programmed in this way if the “Abbr Dial Programming By Assigned List” field is set to ***y*** on the “Feature-Related System-Parameters” form.

- Program the Group List using the AD Group List Program FAC, which is available with G3V4 and later releases. When the AD Group List Program FAC is used to program a group list, the user can program any list for which they are specified as the designated user.

The following instructions describe how designated users program group lists from their voice terminals. Group lists can also be programmed on the switch by the System Administrator.

- Programming the Group List With the AD Program Access Code or AD Program Button With an Abbreviated Dialing Button

  An Abbreviated Dialing button can be used by the designated user to program the Group List number associated with that button. The user dials the AD Program access code or presses the AD Program button. The user then presses the AD button, dials the desired number, and then presses the #. Confirmation tone is heard when the number is stored.

- Programming the Group List With the AD Program Access Code or AD Program Button (No Abbreviated Dialing Button)

  The user dials the AD Program access code or presses the AD Program button. The user then dials 1, 2, or 3 to select a Group List (administered on their “Station” form) followed by the 2-digit List Entry Number of the entry to be programmed. The user then dials the desired number, and then presses the *#*. Confirmation tone is heard when the number is stored.

- Programming the Group List With the AD Group List Program FAC

  The user dials the AD Group List Program FAC. He or she then dials the three, or four-digit Group List Number to select a Group List followed by the two-digit List Entry Number of the entry to be programmed. The user then dials the desired number, and then presses the *#*. Confirmation tone is heard when the number is stored.

Considerations

Abbreviated Dialing provides easy access to selected numbers by decreasing the number of dialed digits required to place the call. Instead of dialing the entire number, the user merely dials a short code to access the desired number. The system then dials the stored number automatically. For frequently called numbers, an Abbreviated Dialing button or Automatic Dialing button can be assigned, allowing the call to be placed by merely pressing the button. By assigning a Privileged list of numbers, a user is allowed to place calls to selected numbers that might otherwise be restricted.

Users can be assigned access to three AD lists. The three lists can be made up of any combination of up to three Personal Lists, up to three Group Lists, the System List, and the Enhanced List. A Personal List cannot be administered to an attendant console.

A number stored in any list in the switch can contain up to 24 digits. A special character used for Pause, Wait, Indefinite Wait, Mark, or Suppress counts as 2-digits.

A terminal or attendant cannot be removed if it is designated as the extension number that is permitted to program a Group List.

Abbreviated dialing digits may be used after the call is connected to send digits from the originator to other connected parties. Other parties may not use abbreviated dialing buttons to send digits back to the originator. Pause characters in abbreviated dialing strings may be ignored if used after the call is connected, and may cause loss of digits when the total number of digits used exceeds 36.

Security Measures

Privileged group, system and enhanced lists give access to calls from stations that would otherwise be restricted.

Interactions

The following features interact with the "Abbreviated Dialing (AD)" feature.

- Attendant Consoles

  Automatic Dialing Buttons are not allowed on Attendant Consoles.

- AUDIX Interface

  When using an Abbreviated Dialing button to access AUDIX, the user’s login and password may be assigned to the button.

- Bridged Call Appearance

  A user, accessing Abbreviated Dialing while on a bridged call appearance, accesses his or her own Abbreviated Dialing lists. The user does not access the Abbreviated Dialing lists of the primary extension associated with the bridged call appearance.

A designated user permitted to program a Group List is not permitted to program the Group List from the bridged call appearance of the designated extension number.

- Last Number Dialed

  This feature places a call to the same number as called previously, even if Abbreviated Dialing was used on the previous call.

  If the previously called number was in an Abbreviated Dialing Privileged List, and if the user is not normally allowed to dial the number because of his or her Class of Restriction, Intercept Treatment is given when using Last Number Dialed. To redial the number, the user must again use the Abbreviated Dialing Privileged List.

- Remote Access

  Remote Access users cannot access Abbreviated Dialing.

Administration

Abbreviated Dialing is administered by the System Manager. However, an Abbreviated Dialing Personal List can be programmed by either the System Manager or the voice terminal user. With G3V4 and later releases, the Group List can be programmed by a designated user or the System Manager.

The System Manager must establish a Personal Number List. For example, during implementation, a voice terminal must first be assigned a Personal Number List on the “Individual Voice Terminal” form. The list can be established on the “Abbreviated Dialing Personal List” form or from the user’s voice terminal. In order to use all members of a Personal List for an individual voice terminal, the list must be modified to expand it to the maximum members for that list on the Abbreviated Dialing List form.

The following items, if required, are set by the System Manager:

- Feature Access Codes for List 1, List 2, and List 3, for accessing a personal list. Also FAC for programming a personal or group list
- Voice Terminal Assignments
  - AD buttons, if desired
  - AD Program button, if desired
  - Automatic Dialing buttons, if desired
  - Mark, Pause, Suppress, Wait and Function Entry buttons, if desired
  - Access to as many as three lists
- Data Module Assignments (Access to an Abbreviated Dialing list)
- Abbreviated Dialing Lists
  - Personal Number Lists

- — Group Number Lists
- — System Number List
- — Enhanced Number List
- — 7103A Group Number List

- Wait Delay Interval (5 to 25 seconds)
- Attendant Console Parameters

See *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655 or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for complete instructions for administering the "Abbreviated Dialing (AD)" feature.

Hardware and Software Requirements

Additional tone detectors such as TN744C-Tone Detector/Call Classifier, TN748C-Tone Detector, TN420C-Tone Detector, or TN2182-Tone Clock/Detector/Generator may be required if the special ***wait*** character is used frequently. See *DEFINITY Communications System Generic 3 System Description and Specifications,* 555-230-206, for more specifics.

Optional software is required for the Enhanced Abbreviated Dialing list.

Add/Remove Skills

Feature Availability

This feature is available with all G3V3 and later versions with the "Expert Agent Selection (EAS)" feature enabled.

Description

This feature allows agents using Expert Agent Selection (EAS) to add or remove skills.

A *skill* is a talent or ability of an agent and a requirement of a caller that is identified within the switch by a number. For example, if an agent has the ability to speak English and Spanish, that agent could be assigned a Spanish speaking skill that has a numerical identifier, such as ***50***. The agent can then add skill 50 to his/her set of working skills. If a customer requires a Spanish-speaking agent, the system can route the call to the agent(s) with that skill. Skills can be administered for a vector directory number (VDN) and an ACD agent login ID, and can be active for an ACD caller via vector commands. Each agent can have up to four skills active at any one time.

NOTE:
In the EAS environment, agents must have at least one skill assigned to them during a login session. Therefore, if an agent with only one skill tries to remove that skill, the system does not remove the skill. Also, the system does not allow an agent with four active skills to add a fifth skill.

This feature allows agents to add or remove a skill by dialing a feature access code (FAC). Agents use one FAC to add a skill and another FAC to remove a skill. Also, any voice terminal user with console permission can add or remove an agent’s skill on behalf of the agent by entering the agent’s login ID.

Agents and supervisors can use queue-status indications, VuStats, or Call Management System (CMS) or Basic Call Management System (BCMS) information to determine if they need to add or remove a skill. When adding a skill, the agent must specify if the skill is primary or secondary. A primary skill is one that the agent answers before answering any secondary skills, if there are calls in queue for both skills. Therefore, it is worth noting that adding or removing a primary or secondary skill does impact how calls are distributed to an agent. The system displays text to *prompt* the agent through the process of adding or removing a skill (assuming the terminal is equipped with a display).

When an agent adds or removes a skill, the system displays on the voice terminal (assuming the terminal is equipped with display) the updated set of skills. Each skill number is preceded by either ***p*** for primary or ***s*** for secondary.

If a supervisor with console permission adds or removes a skill for an agent, the agent does not receive any notification that the change has taken place.

Feature History

This feature was available in the DEFINITY G2 Version 2 release with EAS. Therefore, its addition to G3V3 and later releases fills a void in previous G3 releases. However, the feature has been enhanced in this release by the addition of agent ability to designate a skill as primary or secondary when they add the skill.

Considerations

A skill cannot be removed from an agent's skill set if the agent is on a call for that skill. A skill cannot be removed from an agent's skill set if the agent is in the After-Call-Work (ACW) state for that skill.

If a supervisor with console permission adds or removes a skill for an agent, the agent does not receive any notification that the change has taken place. An agent or other user does not need to be logged in for a skill to be added or removed. Also, agents and users cannot add or remove a skill while on a call or in ACW.

Interactions

- Auto-Available Splits (AAS)

  If an agent adds a skill that is administered as Auto-Available, the agent's login ID must have the "AAS" field set to ***y*** on the "Agent Login ID" form, or the agent cannot add the skill.

- BCMS

  BCMS begins tracking the new skill as soon as it is added. When an agent removes a skill, the real-time agent information specific to that skill is removed from the real-time reports. The data for the skill still appears on the historical reports.

- VuStats

  Because VuStats displays information gathered by BCMS (whether BCMS is enabled or not), the BCMS interaction mentioned above applies to VuStats also.

Administration

EAS must be enabled on the "Feature-Related System-Parameters" form before using the "Add/Remove Skills" feature.

Feature access codes (FACs) for adding and removing skills must be entered in the “Feature Access Code” form. Also, for each class of restriction (COR), the ability for users with that COR to add and remove skills must be enabled or disabled. This is done on the “Class of Restriction” form. Console permissions are administered on the “Class of Service” form.

Text for user-defined displays for the "Add/Remove Skills" feature are administered on page 5 of the “Display-Messages Miscellaneous” form.

NOTE:
Skills are defined on the “Hunt Group” form. Each skill can be administered on the ‘Vector Directory Number’ form or directly in vectors.

Hardware and Software Requirements

EAS is required.

Administered Connections

Feature Availability

Administered Connections is available with all Generic 3 releases except G3vs/G3s ABP and replaces the Permanent Switched Calls feature of previous System 75 releases and of DEFINITY Generic 1.

Description

Automatically establishes an end-to-end connection between two access/data endpoints. The "Administered Connections" feature provides the following enhanced capabilities.

- Support of both permanent and scheduled connections
- Auto Restoration (preserving the active session) for connections routed over Software Defined Data Network (SDDN) trunks
- Administrable retry interval (from 1 to 60 minutes) per Administered Connection
- Administrable alarm strategy per Administered Connection
- Establishment/retry/auto restoration order based on administered priority

The status of an Administered Connection (disabled, connected, and so on) can be displayed by entering the **status administered-connection** command from the Management Terminal.

The endpoints which can be connected via the "Administered Connections" feature are either access endpoints or data endpoints. Access endpoints are nonsignaling trunks and data endpoints are devices that connect the switch to data terminal/communication equipment. Throughout this section the term *endpoint* is used to mean either data endpoint or access endpoint.

Access Endpoints

An access endpoint is either a nonsignaling channel on a DS1 interface or a nonsignaling port on an Analog Tie Trunk circuit pack that is assigned a unique extension. Since an access endpoint is nonsignaling, it will neither generate nor respond to signaling. As a result, an access endpoint cannot be used as a trunking facility (it cannot receive incoming calls or route outgoing calls). An access endpoint is used primarily to support devices, switches, or services that have a trunk interface but do not support signaling for the trunk. An access endpoint may be designated as the originating (local) endpoint or destination endpoint in an Administered Connection. The status of an access endpoint can be displayed by entering the **status access-endpoint** command from the Management Terminal.

If a data call/connection between two access endpoints is set up from a voice station via the "Transfer" feature, the call can only be dropped (and the endpoints freed) by busying out either one of the Access Endpoints or a trunk over which the connection is routed (if one exists) from the Management Terminal. This is required since neither of the endpoints can initiate a drop (access endpoints are nonsignaling).

Typical Administered Connection Configurations

The "Administered Connections" feature allows a great amount of flexibility in assigning the destination address of the connection. As a result, many different configurations are possible with an Administered Connection. An Administered Connection can be established between two endpoints on the same switch; between two endpoints in the same private network, but on different switches; or between an endpoint on the controlling switch and another endpoint off the private network. In all configurations, the Administered Connection must be administered on the same switch as the originating endpoint.

If the two endpoints of the Administered Connection are on two different switches within a private network, normally, the connection will be routed through tie trunks (such as ISDN-PRI, DS1 or analog tie trunks) and possibly intermediate switches. However, if preferred, the connection can be routed through the public network.

The following are typical Administered Connection configurations and their application examples:

- A local data endpoint connects to a local or a remote access endpoint.

  One example of this is an MPDM connecting to a T1 Multiplexer via a DS0.
- A local access endpoint connects to a local/remote access endpoint.

  Two examples are a DS0 cross-connect and a four-wire leased line modem to a four-wire leased line modem connection via analog tie trunks.
- A local data endpoint connects to a local/remote data endpoint.

  One example is a connection between two 3270 data modules.

Establishment of Administered Connections

The originating switch will only attempt to establish an Administered Connection if the following conditions exist:

a. the Administered Connection is enabled

b. the Administered Connection is due to be active (either a permanent Administered Connection or the time of day requirements are satisfied if a scheduled Administered Connection)

c. the originating endpoint is in the in-service/idle state

If the originating endpoint is not in-service/idle, no activity will take place for the Administered Connection until the endpoint transitions to the desired state. The destination address is used by the originating switch to route the connection to the desired endpoint. When two or more Administered Connections are to be established at the same time, they are established in priority order.

Administered Connection Establishment Retries

Administered Connection establishment attempts can fail for the following reasons:

- resources are unavailable to route to the destination
- a required conversion resource is not available
- access is denied. COR, FRL, BCC, or an attempt is made to route voice-band-data over SDDN trunks in the 4ESS switch network (or other public switch network)
- incorrect destination address
- destination endpoint is busy
- other network or signaling failure

In the event of a failure, an error will be logged in the error log and an alarm will be generated, if it is warranted by the alarming strategy. The reason an Administered Connection has failed can be displayed by the System Manager via the **status administered-connection** command. This information is also contained in the error log.

As long as an Administered Connection is due to be active, continued attempts to establish an Administered Connection will be made by the originating switch unless the establishment attempt failed because of an administrative error (like a wrong number) or service blocking condition (like outgoing calls barred). Establishment attempts for Administered Connections that fail as a result of one of these conditions will resume when the problem is resolved (that is, Administered Connection administration has been changed). The frequency at which failed establishment attempts are retried is determined by the administered retry interval (1 to 60 minutes) of each Administered Connection. Retries will be made after the retry interval has elapsed regardless of the restorable attribute of the Administered Connection. If more than one Administered Connection is to be retried at the same time, they will be retried in priority order. When the customer changes the time of day on the switch, an attempt will be made to establish all Administered Connections in the 'waiting for retry' state.

Dropping an Administered Connection

Once established, an Administered Connection will remain active until one of the following events occurs:

- The Administered Connection is changed, disabled, or removed. (See the “Administration” section for identification of which attributes, when changed, will result in the dropping of an active Administered Connection.)
- The time of day requirements of a scheduled Administered Connection are no longer satisfied.
- One of the endpoints initiates dropping the connection. This could be a result of a user initiated drop (in the case of a data endpoint), maintenance activity resulting from an endpoint failure, or the busying out of the endpoint or handshake failure. If the endpoints involved in an Administered Connection are incompatible, the connection will successfully connect before the handshake failure occurs.

  NOTE:
  Administered Connections between access endpoints will remain connected even if the attached access equipment fails to handshake.

- An interruption (that is, facility failure) occurs in the path between the endpoints involved in the Administered Connection.

No action is taken if an Administered Connection drops because it was disabled/removed or is no longer due to be active. If an Administered Connection drops because of changed Administered Connection attributes, an immediate attempt will be made to establish the connection with the changed attributes if it is still due to be active. Existing entries in the error/alarm log are resolved if they no longer apply. If it can be determined that handshake failure resulted in the dropping of the connection, in the case of an Administered Connection involving at least one data endpoint, no action will be taken for that Administered Connection until the **change administered-connection** command has been executed.

Administered Connection Failure: Auto Restoration and Fast Retry

When an active (established) Administered Connection drops prematurely, either auto restoration or fast retry will be invoked. It can be determined whether or not auto restoration will be attempted for an active Administered Connection by observing the contents of the restorable field displayed on the Status Administered Connection screen.

Auto restoration will be attempted if the Administered Connection was optioned for auto restoration and the connection was routed over SDDN trunks. During restoration, connections are maintained between the switch and endpoint at both ends of the connection. In addition to allowing the active session to be maintained, this also provides a high level of security by prohibiting other connections from intervening in active sessions. The ‘Auto Restoration’ feature cannot guarantee restoration within a certain time period, but successful restorations (involving remote endpoints on a G3i switch) must be completed before the expiration of the 60-second endpoint holdover timer utilized during restoration. If auto restoration is successful, the session that was active when the failure occurred might be maintained (no guarantee). If the session is maintained, the restoration is transparent to the user with the exception of a temporary disruption of service while the restoration is in progress. A successful restoration is reflected by the *restored* state on the Status Administered Connection screen. The restored status will be displayed, even if the destination endpoint was idle (that is, already dropped) when the restoration attempt arrived at the destination node. (Although the restoration was successful, the data session may not have been preserved.)

If the auto restoration function is not optioned or the Administered Connection is not routed over SDDN trunk(s), the switch will immediately attempt to reestablish the connection (fast retry). Fast retry will also be attempted if the originating endpoint initiated the drop. In the event of a fast retry, connections are not maintained on both ends. Fast retry will not be attempted for an Administered Connection which was last established via fast retry, unless the Administered Connection has been active for at least two minutes.

If the auto restoration or fast retry attempt fails to restore/reestablish the connection, the connection will be dropped and the Administered Connection will go into retry mode. Retry attempts will continue, at the administered retry interval, as long as the Administered Connection is due to be active.

Interactions

The following features and functions interact with the "Administered Connections" feature:

- Abbreviated Dialing

  Abbreviated dialing entries can be used in the “Destination Address” field. Entries must comply with the restrictions of the dial plan.

- AAR/ARS/Generalized Routing Selection (GRS)

  These features may be used in the routing of an Administered Connection.

- Busy Verification of Stations and Trunks

  This feature does not apply to access endpoints because access endpoints are used only for data.

- Class of Restriction

  A COR should be reserved for Administered Connection endpoints and SDDN trunks. This would restrict endpoints, not involved in Administered Connections, from connecting to SDDN trunks or endpoints involved in Administered Connections.

- Class of Service/Call Forwarding

An Administered Connection endpoint should be assigned a Class of Service that will block call forwarding activation of the endpoint.

- Data Call Setup

  A Default Dialing destination should not be assigned to a data module that is used in an Administered Connection.

- Data Call Hotline

  A hotline destination should not be assigned to a data module that is used in an Administered Connection.

- Digital Multiplexed Interface (DMI)

  DMI endpoints can be used as the destination in an Administered Connection. DMI endpoints do not have associated extensions, so they cannot be used as the originator in an Administered Connection.

- Facility Test Calls

  The feature does not apply to access endpoints because an access endpoint acts as an endpoint rather than as a trunk.

- Hunting

  A hunt group extension is not allowed to be used as the origination extension of an Administered Connection.

- Modem Pooling

  If a conversion resource (pooled modem) is required in an Administered Connection, one will be inserted. If no conversion resource is available, the connection will be dropped.

- Non-Facility Associated Signaling (NFAS) and D-Channel Backup

  Auto Restoration for Administered Connections, initially routed over an NFAS facility, may fail if the only backup route is over the facility on which the backup D-channel is administered, since the backup D-channel may not come into service in time to handle the restoration attempt.

- Set Time Command

  When the System Manager changes the system time via the **set time** command, all scheduled Administered Connections are examined. If the system time change causes an active Administered Connection to be outside its scheduled period, the Administered Connection will be dropped. If the time change causes an inactive Administered Connection to now be within its scheduled period, the switch will attempt to establish the Administered Connection.

  Also, if any Administered Connection (scheduled or continuous) is in the retry mode and the system time changes, the switch will attempt to establish the Administered Connection immediately.

- CDR

For an Administered Connection that uses a trunk which has CDR enabled, the origination extension of the Administered Connection will be used as the originator of the call.

CDR is not available for access endpoints.

- System Measurements

  Access endpoints are not measured. All other trunks in an Administered Connection are measured as usual.

- Terminal Dialing

  It is recommended that the terminal dialing capability be turned off for data modules involved in an Administered Connection.

  NOTE:
  This will stop call processing related messages (INCOMING CALL,...) from being displayed on the terminal.

- Trunk Groups

  In order for auto restoration to be invoked, an Administered Connection must be routed over SDDN trunks. Since a successful restoration depends on there being an SDDN path over which to route the restoration attempt, some SDDN trunks should be kept idle to be used in the event of failure for restoration. SDDN trunk group usage should be restricted to Alternating Current (AC) related traffic.

Administration

Each Administered Connection is administered by the System Manager. The following items require administration.

- Endpoints

  If an Administered Connection involves local endpoints, the endpoints must be administered before the Administered Connection using those endpoints can be administered. An endpoint cannot be removed if it is involved in a locally administered Administered Connection, or if it is currently involved in an active Administered Connection. If the user desires to change any of the translation data associated with an endpoint (except for the “Name” field which may be changed at any time) that is involved in an active Administered Connection, the Administered Connection must first be disabled or removed.

- Administered Connection

  An Administered Connection must be administered on the same switch as the originating endpoint. The System Manager may change the attributes of an Administered Connection at any time, but not all changes take effect immediately. These attributes are as follows (included in each description is a statement as to whether or not changes take effect immediately):

— **Originating Address** — The address of the originating endpoint is its local extension on the originating switch. When this attribute is changed for an active Administered Connection, the connection will be dropped and reestablished using the new originating address.

— **Destination Address** — The destination address is used to route the Administered Connection to the desired destination. When this attribute is changed for an active Administered Connection, the connection will be dropped and reestablished using the new destination address.

— **Enable** — The “Enable” field allows the System Manager to specify whether the system should attempt to establish the connection when it is due to be active. Answering ***yes*** to the enable option indicates that the system should be established when the Administered Connection is due to be active. Answering ***no*** indicates that the System Manager does not want the Administered Connection to be considered for activation at this time (that is, held for future use). A disabled Administer Connection is displayed along with the other Administered Connections administered locally in response to the **list administered-connection** command. A disabled Administered Connection may be enabled at any time. Since Administered Connection administration is done on the originating switch, disabling an Administered Connection can only be done on the originating switch.

  If an Administered Connection is currently active, answering ***no*** causes the Administered Connection to be dropped immediately. If an Administered Connection is disabled, answering ***yes*** will cause the originating switch to attempt to establish the connection immediately if the Administered Connection is due to be active. *The disabling and enabling of an Administered Connection after an attribute of the Administered Connection has been changed guarantees that the change will take effect immediately.*

— **Name** — A one through 15 character long, optional “Name” field is provided to allow for additional identification information. Changing this field has no effect on the Administered Connection connection.

— **Authorized Time of Day** — An Administered Connection may be continuous (permanent) or scheduled. Scheduled Administered Connections are described by indicating the days of the week, start time, and the duration for which the Administered Connection is to be active. The modification of any of the attributes associated with the authorized time of day will not affect the current status of an Administered Connection unless the change results in activating or deactivating an Administered Connection.

— **Priority** — The System Manager can specify the priority of a given Administered Connection. This priority is used to determine the order in which Administered Connections are established if two or more Administered Connections are due to be active at the same time. The “Priority” field allows the user to specify a number between one and eight (with one being the highest and eight being the lowest). Changes to the priority attribute have no effect on an active Administered Connection.

— **Auto Restoration** — The System Manager may specify whether an attempt should be made to restore an Administered Connection, via the ‘Auto Restoration’ feature, if the connection is dropped due to failure and the connection was routed over SDDN trunk(s). Reestablishment (retry) of dropped connections is attempted regardless of the value specified in this field. This field has no effect on Administered Connections routed over non-SDDN trunks. The System Manager must disable and enable an active Administered Connection to have changes to this attribute take effect.

— **Retry Interval** — The System Manager must specify a retry interval of 1 to 60 minutes. The default is two minutes. This interval is the number of minutes waited before a retry is attempted. When this field is changed, the new interval will be used for the next retry. An Administered Connection, which is in retry mode when this field is changed, will retry after the old interval has elapsed and then use the new interval for the next retry time. Twenty-three of the Administered Connections will be restored within the required time. The remaining Administered Connections will be restored, but after the time limit.

— **Alarm Type** — An alarm type of none, warning, minor or major must be chosen. The default will be warning. ***None*** indicates that no alarms will be generated on establishment or restoration failure. A choice of ***warning*** will cause alarms to be generated and logged in the alarm and error log. A minor or major indication will also cause alarms to be generated and logged in the alarm and error log and forwarded to an Operations Support System such as INADS if OSS is administered.

Changing this field to ***none*** will cause an existing alarm to be cleared. Changing the field to one of the other values will cause the upgrading or the downgrading of an existing alarm.

— **Alarm Threshold** — The “Alarm Threshold” field indicates the number of consecutive failures (1 to 10) that must occur before an Administered Connection alarm is generated. Entering 1 in this field will cause alarms to be generated immediately upon failure to establish or reestablish an Administered Connection.

Changes to this field take effect immediately. A comparison of the new value and the current retry count will be made to determine if an alarm should be generated or possibly cleared due to the change.

■ Access Endpoint

The access endpoint has the following attributes which must be administered:

— **Extension** — This is a unique one- to five-digit identifier, consistent with the current dial plan, by which this access endpoint is addressed.

— **Port** — This is the port address of the DS1 or analog tie trunk port. A DS1 trunk can be used regardless of the signaling mode of the DS1 circuit pack.

— **Name** — This can be any alphanumeric string (up to 10 characters) representing a name that is useful to the customer.

— **Communication Type** — A communication type of 64K data, 56K data, or voice-band data must be assigned. An access endpoint on an analog tie trunk port is restricted to a communication type of voice-band data. In addition, a communication type of 64K is not allowed for access endpoints on DS1 circuit packs administered for robbed-bit signaling.

— **COR** — A Class of Restriction may be administered for each access endpoint.

— **COS** — A Class of Service may be administered for each access endpoint. Class of Service administration should be used to block call forwarding activation of an endpoint.

Hardware and Software Requirements

Hardware requirements vary depending on the type of Administered Connection desired. The following hardware may be required for Administered Connections:

- Access Endpoint Circuit Packs — TN767 DS1 Interface circuit pack (TN464B/C/D support A-law), TN760B Analog Tie Trunk (TN760D supports A-law).
- Data Endpoint Circuit Packs — TN726 Data Line or TN754 Digital Line (TN413, TN754B support A-law).
- Data Modules — 700A/700D PDM or MPDM, 700B/700C/700E TDM or MTDM, 7400D series voice terminal with DTDM, PC/PBX, 510D, 515BCT.
- Trunk Circuit Packs — TN767 DS1 Interface circuit pack (TN464B/C/D support A-law), TN760 Analog Tie Trunk (TN760D supports A-law).
- TN758 Pooled Modem circuit pack.

No additional software is required.

Administrable Language Displays

Feature Availability

This feature is available with Generic G3i-Global and all Generic V2 and later releases.

The user-defined display language option is only available with the Generic V2 and later releases.

Description

There is a set of messages that appear on a DCP/ISDN-BRI Station/Attendant users set that provide call related information.

The "Administrable Language Displays" feature allows 40-character display station users or an attendant user to select a display language for these messages. This display language selection is made via administration; with the options being English (default), French, Italian, or Spanish. Note that the messages themselves do not change, only the wording that is used to present the message is changed.

The Generic V2 and later releases allow the addition of one user customized language display message set to the system. This additional display message set is entered into the system via administration by either a customer or an AT&T in-country partner and is accessed by a user as their display language preference by selecting the user defined display language option.

Feature Displays

This section shows the English, French, Italian, and Spanish language display message sets subdivided by feature. Since none of the feature functions are modified, no explanation of the feature is made.

The user-defined language display message set is the user customized language display message set that was added to the system via administration. If user-defined is selected as the display language preference and the *user-defined* language display message set has not been entered into the system, all display messages appear as a string of asterisks.

When the time is displayed, only the English display set uses the AM/PM notation, all others use the 24 hour clock.

Automatic Wakeup

The following displays are associated with the "Automatic Wakeup" feature.

- *"AUTO WAKEUP - Ext: xxxxx Time: --:-- xM"* (English)
  - French - *"REVEIL AUTO. - POSTE: xxxxx HEURE: --:--"*
  - Italian - *"SERVIZIO SVEGLIA - Tel: xxxxx Ora: --:--"*
  - Spanish - *"DESPERT AUTOMA - EXT: xxxxx HORA: --:--"*
- *"INVALID EXTENSION - TRY AGAIN"* (English)
  - French - *"NUMERO DE POSTE EST ERRONE - REESSAYER"*
  - Italian - *"NUMERO ERRATO - RIPETERE"*
  - Spanish - *"EXTENSION NO VALIDO - INTENTE DE NUEVO"*
- *"WAKEUP ENTRY DENIED - INTERVAL FULL"* (English)
  - French - *"DEM. REVEIL REFUSEE - INTERVALLE PLEIN"*
  - Italian - *"SVEGLIA NON ATTIVATA - ORARIO OCCUP"*
  - Spanish - *"ENTRADA DENEGADA - INTERVALO COMPLETO"*
- *"WAKEUP ENTRY DENIED - NO PERMISSION"* (English)
  - French - *"DEM. REVEIL REFUSEE - SANS AUTORISATION"*
  - Italian - *"SVEGLIA NON ATTIVATA - NON PERMESSO"*
  - Spanish - *"ENTRADA DENEGADA - SIN PERMISO"*
- *"WAKEUP ENTRY DENIED - SYSTEM FULL"* (English)
  - French - *"DEM. REVEIL REFUSEE - ENCOMBREMENT"*
  - Italian - *"SVEGLIA NON ATTIVATA - CONGESTIONE"*
  - Spanish - *"ENTRADA DENEGADA - SISTEMA COMPLETO"*
- *"WAKEUP ENTRY DENIED - TOO SOON"* (English)
  - French - *"DEM. REVEIL REFUSEE - TROP TOT"*
  - Italian - *"SVEGLIA NON ATTIVATA - TROPPO PRESTO"*
  - Spanish - *"ENTRADA DENEGADA - MUY PRONTO"*
- *"WAKEUP REQUEST CANCELED"* (English)
  - French - *"DEMANDE DE REVEIL EST ANNULEE"*
  - Italian - *"RICHIESTA SVEGLIA CANENTRYATA"*
  - Spanish - *"SOLICITUD DE DESPERTADOR CANCELADA"*
- *"WAKEUP REQUEST CONFIRMED"* (English)
  - French - *"DEMANDE DE REVEIL EST CONFIRMEE"*
  - Italian - *"RICHIESTA SVEGLIA CONFERMATA"*
  - Spanish - *"SOLICITUD DE DESPERTADOR CONFIRMADA"*
- *"Wakeup Call"* (English)

— French - *‘‘APPEL DE REVEIL’’*

— Italian - *“Serv. Sveglia’’*

— Spanish - *“Despierte’’*

ASAI

The following display is associated with the "CallVisor Adjunct/Switch Applications Interface (ASAI)" feature.

- *“You have adjunct messages’’* (English)
  - — French - *‘‘MESSAGES SUPPLEMENTAIRES’’*
  - — Italian - *‘‘MESSAGGI AGGIUNTIVI’’*
  - — Spanish - *‘‘TIENE MENSAJES ADICIONALES’’*

Busy Verification of Stations and Trunks

Table 3-2 lists the displays associated with the "Busy Verification of Terminals and Trunks" feature.

Table 3-2. Busy Verification of Stations and Trunks Displays.

English Display	French Display	Italian Display	Spanish Display
“ALL MADE BUSY”	*“TOUS OCC.’’*	*“TUTTI OCCUPATI’’*	*“TODAS OCUPADAS’’*
“BRIDGED”	*“EN DERIVATION”*	*“OCCUPATO”*	*“PUENTEADA”*
“DENIED”	*“INTERDIT’’*	*“NON PERMESSO”*	*“DENEGADO”*
“INVALID’’	*“ERRONE”*	*“NON VALIDO”*	*“NO VALIDO”*
“NO MEMBER’’	*“AUCUN MEMBRE”*	*“NESSUN ELEMENTO’’*	*‘‘NINGUN MIEMBRO”*
“OUT OF SERVICE”	*“HORS SERVICE”*	*“FUORI SERVIZIO”*	*“FUERA SERVICIO’’*
“RESTRICTED’’	*“RESTREINT”*	*“RISTRETTO”*	*“RESTRINGIDO”*
“TERMINATED”	*“TERMINE’’*	*“TERMINATO”*	*“TERMINADO”*
“TRUNK SEIZED”	*“CIRCUIT SAISI”*	*“GIUNZIONE IMP.”*	*“ENLACE OCUPADO’’*
“VERIFIED”	*“VERIFIE’’*	*“VERIFICATO’’*	*“VERIFICADO’’*

Call Appearance Designation

For each of the display language options, the display to indicate call appearance designation appears as:

- *‘‘a =’’* (English)

  Call appearance buttons are designated on the display by a lower case letter (a through z for the first 26 call appearances then A through Z) in position 1, followed by an ‘‘=.’’

Call Detail Record

The following display is associated with the "Call Detail Recording (CDR)" feature.

- *“CDR OVERLOAD”* (English)
  - — French - *“SURCHARGE EDA”*
  - — Italian - *“SVRACCARICO DAC”*
  - — Spanish - *“SOBRECARGA DAT”*

Call Progress Feedback

Table 3-3 lists the call progress displays.

Table 3-3. Call Progress Feedback Displays

“English Display” (stands for)	**“French Display” (stands for)**	**“Italian Display” (stands for)**	**“Spanish Display” (stands for)**
“busy” (Extension Busy, Intrusion Not Allowed, Call Waiting Not Allowed)	*“OCCUPE”* (Occupe)	*“occ”* (Occupato)	*“OCUPADA”* (Ocupada)
“busy(I)” (Extension Busy, Intrusion Allowed, Call Waiting Not Allowed)	*“OCC.(E)”* (Entree ligne occupe)	*“occ(I)”* (Occupato-Intrusione)	*“OCUP(I)”* (Ocupada-intrusion)
“ringing” (Extension Ringing)	*“SONNE”* (Libre)	*“libero”* (Libero)	*“LIBRE”* (Libero)
“wait” (Extension Busy, Intrusion Not Allowed, Call Waiting Allowed)	*“ATTENTE”* (Attente)	*“auat”* (Autoattesa)	*“ESPERA”* (Espera)
“(I) wait” (Extension Busy, Intrusion Allowed, Call Waiting Allowed)	*“(E) ATTENTE”* (Entree ligne attente)	*“(I) auat”* (Intrusione-Autoattesa)	*“(I) ESPERA”* (Intrusion, en espera)

Class of Restriction

Table 3-4 lists displays associated with the "Class of Restriction (COR)" feature.

Table 3-4. Class of Restriction Displays

Restriction	‘‘English Display’’	‘‘French Display’’	‘‘Italian Display’’	‘‘Spanish Display’’
Toll	*‘‘TOLL’’*	*‘‘INT.’’*	*‘‘TASS’’*	*‘‘TARF’’*
Full	*‘‘FULL’’*	*‘‘COM.’’*	*‘‘DISB’’*	*‘‘LLEN’’*
No Restrictions	*‘‘NONE’’*	*‘‘AUC.’’*	*‘‘ABIL’’*	*‘‘NING’’*
Origination	*‘‘ORIG’’*	*‘‘DEP.’’*	*‘‘ORIG’’*	*‘‘ORIG’’*
Outward	*‘‘OTWD’’*	*‘‘SOR.’’*	*‘‘USCN’’*	*‘‘SALI’’*

Date/Time Mode and Formats

The following displays are associated with the ‘Date & Time’ feature.

If the time is *not* available:

- English - *‘‘SORRY, TIME UNAVAILABLE NOW’’*
- French - *‘‘HEURE ET DATE INDISPONIBLES’’*
- Italian - *‘‘ORA E DATA TEMP. NON DISPONIBILI’’*
- Spanish - *‘‘HORA Y FECHA NO DISPONIBLES AHORA’’*

If the time *is* available, Screen 3-1 lists the English date/time mode and formats attributes; Screen 3-2 lists the French, Italian, Spanish, or user-defined date/time mode and formats attributes;

— English -

```
<DATE/TIME>      <TIME><b><b><DATE>
<TIME>           <HR>:<MIN><b><M>
<HR>             1-12 (hour of day, no leading zeroes)
<MIN>            00-59 (minute of hour)
<M>              ‘‘am’’ or ‘‘pm’’
<DATE>           <DOW><b><MONTH><b><DOM>,<b><YEAR>
<DOW>            Day of week, upper case, unabbreviated
<MONTH>          Month of year, upper case, unabbreviated
<DOM>            1-31 (day of month, no leading zeroes)
<YEAR>           Year in 4 digits
<b>              Blank
```

Screen 3-1. Date/Time Mode and Formats — English

— French, Italian, & Spanish, user-defined -

```
<DATE/TIME>      <TIME><b><b><DATE>
<TIME>           <HR>:<MIN>
<HR>             0-23 (hour of day, no leading zeroes)
<MIN>            00-59 (minute of hour)
<DATE>           <DOW><b><DOM><b><MONTH>,<b><YEAR>
<DOW>            Day of week, upper case, unabbreviated
<DOM>            1-31 (day of month, no leading zeroes)
<MONTH>          Month of year, upper case, unabbreviated
<YEAR>           Year in 4 digits
<b>              Blank
```

Screen 3-2. Date/Time Mode and Formats — French, Italian, Spanish, or User-Defined

Table 3-5 lists the days of the week format.

Table 3-5. Days of the Week Format

"English Display"	**"French Display"**	**"Italian Display"**	**"Spanish Display"**
"SUNDAY"	*"DIMANCHE"*	*"DOMENICA"*	*"DOMINGO"*
"MONDAY"	*"LUNDI"*	*"LUNEDI"*	*"LUNES"*
"TUESDAY"	*"MARDI"*	*"MARTEDI"*	*"MARTES"*
"WEDNESDAY"	*"MERCREDI"*	*"MERCOLEDI"*	*"MIERCOLES"*
"THURSDAY"	*"JEUDI"*	*"GIOVEDI"*	*"JUEVES"*
"FRIDAY"	*"VENDREDI"*	*"VENERDI"*	*"VIERNES"*
"SATURDAY"	*"SAMEDI"*	*"SABATO"*	*"SABADO"*

Table 3-6 lists the months of the year format.

Table 3-6. Months of the Year Format

‘‘English Display’’	**‘‘French Display’’**	**‘‘Italian Display’’**	**‘‘Spanish Display’’**
‘‘JANUARY’’	*‘‘JANVIER’’*	*‘‘GENNAIO’’*	*‘‘ENERO’’*
‘‘FEBRUARY’’	*‘‘FEVRIER’’*	*‘‘FEBBRAIO’’*	*‘‘FEBRERO’’*
‘‘MARCH’’	*‘‘MARS’’*	*‘‘MARZO’’*	*‘‘MARZO’’*
‘‘APRIL’’	*‘‘AVRIL’’*	*‘‘APRILE’’*	*‘‘ABRIL’’*
‘‘MAY’’	*‘‘MAI’’*	*‘‘MAGGIO’’*	*‘‘MAYO’’*
‘‘JUNE’’	*‘‘JUIN’’*	*‘‘GIUGNO’’*	*‘‘JUNIO’’*
‘‘JULY’’	*‘‘JUILLET’’*	*‘‘LUGLIO’’*	*‘‘JULIO’’*
‘‘AUGUST’’	*‘‘AOUT’’*	*‘‘AGOSTO’’*	*‘‘AGOSTO’’*
‘‘SEPTEMBER’’	*‘‘SEPTEMBRE’’*	*‘‘SETTEMBRE’’*	*‘‘SEPTIEMBRE’’*
‘‘OCTOBER’’	*‘‘OCTOBRE’’*	*‘‘OTTOBRE’’*	*‘‘OCTUBRE’’*
‘‘NOVEMBER’’	*‘‘NOVEMBRE’’*	*‘‘NOVEMBRE’’*	*‘‘NOVIEMBRE’’*
‘‘DECEMBER’’	*‘‘DECEMBRE’’*	*‘‘DICEMBRE’’*	*‘‘DICIEMBRE’’*

Do Not Disturb (Hotel/Motel feature)

The following displays are associated with the Do Not Disturb (DND) feature.

- *‘‘DO NOT DIST - Group: xx   Time: --:-- xM’’* (English)
  - — French - *‘‘NE PAS DERANGER GROUPE: xx HEURE: --:--’’*
  - — Italian - *‘‘NON DISTURBARE - Grp: xx   Ora: --:--’’*
  - — Spanish - *‘‘NO MOLESTAR - GRUPO: xx   HORA: --:--’’*
- *‘‘DO NOT DIST - Ext: xxxxx Time: --:-- xM’’* (English)
  - — French - *‘‘NE PAS DERANGER POSTE:xxxxx HEURE: --:--’’*
  - — Italian - *‘‘NON DISTURBARE - Tel: xxxxx   Ora: --:--’’*
  - — Spanish - *‘‘NO MOLESTAR - EXT: xxxxx   HORA: --:--’’*
- *‘‘DO NOT DIST ENTRY DENIED - INTERVAL FULL’’* (English)
  - — French - *‘‘DEMANDE EST REFUSEE - INTERVALLE PLEIN’’*
  - — Italian - *‘‘SERVIZIO NON ATTIVATO - ORARIO OCCUP’’*
  - — Spanish - *‘‘ENTRADA DENEGADA - INTERVALO COMPLETO’’*
- *‘‘DO NOT DIST ENTRY DENIED - NO PERMISSION’’* (English)
  - — French - *‘‘DEMANDE EST REFUSEE - SANS AUTORISATION’’*

  - — Italian - *“SERVIZIO NON ATTIVATO - NON PERMESSO”*
  - — Spanish - *“ENTRADA DENEGADA - SIN PERMISO”*
- *“DO NOT DIST ENTRY DENIED - SYSTEM FULL”* (English)
  - — French - *“DEMANDE EST REFUSEE - ENCOMBREMENT”*
  - — Italian - *“SERVIZIO NON ATTIVATO - CONGESTIONE”*
  - — Spanish - *“ENTRADA DENEGADA - SISTEMA COMPLETO”*
- *“DO NOT DIST ENTRY DENIED - TOO SOON”* (English)
  - — French - *“DEMANDE EST REFUSEE - TROP TOT”*
  - — Italian - *“SERVIZIO NON ATTIVATO - TROPPO PRESTO”*
  - — Spanish - *“ENTRADA DENEGADA - MUY PRONTO”*
- *“INVALID GROUP - TRY AGAIN”* (English)
  - — French - *“GROUPE ERRONE - REESSAYER”*
  - — Italian - *“GRUPPO NON VALIDO - RIPETERE”*
  - — Spanish - *“GRUPO NO VALIDO - INTENTE DE NUEVO”*
- *“THANK YOU - DO NOT DIST ENTRY CONFIRMED”* (English)
  - — French - *“MERCI - DEMANDE EST CONFIRMEE”*
  - — Italian - *“NON DISTURBARE - RICHIESTA CONFERMATA”*
  - — Spanish - *“NO MOLESTAR - ENTRADA CONFIRMADA”*
- *“THANK YOU - DO NOT DIST REQUEST CANCELED”* (English)
  - — French - *“MERCI - DEMANDE EST ANNULEE”*
  - — Italian - *“NON DISTURBARE - RICHIESTA CANENTRYATA”*
  - — Spanish - *“MUCHAS GRACIAS - SOLICITUD CANCELADA”*

Expert Agent Selection

The following displays are associated with the "Expert Agent Selection (EAS)" feature.

- *“Skills:”* (English)
  - — French - *“Capacites:”*
  - — Italian - *“Capacita:”*
  - — Spanish - *“Capacidades:”*
- *“p”* (English for Primary)
  - — French (for Principale) - *“p”*
  - — Italian (for Primaria) - *“p”*
  - — Spanish (for Primaria) - *“p”*

- *‘‘s’’* (English for Secondary)
  - — French (for Secondaire) - *‘‘s’’*
  - — Italian (for Secondaria) - *‘‘s’’*
  - — Spanish (for Secundaria) - *‘‘s’’*

Field Separator

The following displays show field separation.

- <calling party> *‘‘to’’* <called party> (English)
  - — French - <calling party> *‘‘a’’* <called party>
  - — Italian - <calling party> *‘‘a’’* <called party>
  - — Spanish - <calling party> *‘‘a’’* <called party>

Integrated Directory Display Mode

The following displays are associated with the "Integrated Directory" feature.

- *‘‘DIRECTORY - PLEASE ENTER NAME’’* (English)
  - — French - *‘‘ANNUAIRE - ENTRER LE NOM’’*
  - — Italian - *‘‘ELENCO UTENTI - INTRODURRE NOME’’*
  - — Spanish - *‘‘GUIA TELEFONICA - INTRODUZCA NOMBRE’’*
- *‘‘DIRECTORY UNAVAILABLE - TRY LATER’’* (English)
  - — French - *‘‘ANNUAIRE INDISPONIBLE - REESSAYER’’*
  - — Italian - *‘‘ELENCO UTENTI TEMP. NON DISPONIBILE’’*
  - — Spanish - *‘‘GUIA TEL INDISPONIBLE - INTENTE DESPUES’’*
- *‘‘NO MATCH - TRY AGAIN’’* (English)
  - — French - *‘‘INTROUVABLE - REESSAYER’’*
  - — Italian - *‘‘NESSUNA CORRISPONDENZA - RIPETERE’’*
  - — Spanish - *‘‘NO CORRESPONDE - INTENTE DE NUEVO’’*

ISDN

The following displays are associated with the ‘ISDN’ feature.

- *‘‘ANSWERED BY’’* (English)
  - — French - *‘‘REPONDU PAR’’*
  - — Italian - *‘‘RISPOSTA DA’’*
  - — Spanish - *‘‘RESPONDIDO POR’’*
- *‘‘CALL FROM’’* (English)

  - — French - *“APPEL DE”*
  - — Italian - *“CHIAMATA DA”*
  - — Spanish - *“LLAMADA DE”*
- *“INTL”* (English) - “International”
  - — French - *“INTL”*
  - — Italian - *“INTL”*
  - — Spanish - *“INTL”*

Leave Word Calling

The following screens list displays associated with the "Leave Word Calling"feature.

The format of the *English* Leave Word Calling message is listed in Screen 3-3:

- English -

```
<CALLER_ID><b><DATE><b><TIME><M><b><C><b>CALL<EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <MONTH>/<DOM>
<MONTH>         1-12 (month of year, no leading zeroes)
<DOM>           1-31 (day of month, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            1-12 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<M>             ''a'' or ''p''
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
<b>             blank
```

Screen 3-3. Leave Word Calling — English

NOTE:
If nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

The format of the *French* Leave Word Calling message is listed in Screen 3-4:

- French -

```
<CALLER_ID><b><DATE><b><TIME><b><C><b>APPL<b><EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <DOM>/<MONTH>
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         1-12 (month of year, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
```

Screen 3-4. Leave Word Calling — French

NOTE:
If nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

The format of the **Italian** Leave Word Calling message is listed in Screen 3-5:

- Italian -

```
<CALLER_ID><b><DATE><b><TIME><b><C><b>TEL<b><EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <DOM>/<MONTH>
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         1-12 (month of year, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
```

Screen 3-5. Leave Word Calling — Italian

NOTE:
If nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

The format of the *Spanish* Leave Word Calling message is listed in Screen 3-6:

- Spanish -

```
<CALLER_ID><b><DATE><b><TIME><b><C><b>LLAM<b><EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <DOM>/<MONTH>
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         1-12 (month of year, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
```

Screen 3-6. Leave Word Calling — Spanish

NOTE:
If nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

The format of the *user-defined* Leave Word Calling message is listed in Screen 3-7:

- user-defined -

```
<CALLER_ID><b><DATE><b><TIME><b><C><b><b><EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <DOM>/<MONTH>
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         1-12 (month of year, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<C>             Number of calls received, 1 digit *
<          The user-defined language string for the
English string ''CALL''
<EXT_NO>        Calling extension number, up to 5 digits
```

Screen 3-7. Leave Word Calling — User-Defined

Following are the list of Leave Word Calling messages and their respective translations. Messages can be a maximum of 40 characters.

- *"CANNOT BE DELETED - CALL MESSAGE CENTER"* (English)
  - French - *"NE PEUT ETRE SUPP./APPELER RECEP. MESS."*
  - Italian - *"NON CANENTRYATO. CHIAMARE CENTRO MESSAGGI"*
  - Spanish - *"NO ELIMINADO-LLAMA CENTRO DE MENSAJES"*

- *''DELETED''* (English)
  - French - *''SUPPRIME''*
  - Italian - *''MESSAGGIO CANENTRYATO''*
  - Spanish - *''ELIMINADO''*
- *''END OF MESSAGES (NEXT TO REPEAT)''* (English)
  - French - *''FIN DES MESSAGES (SUIVANT POUR REPETER)''*
  - Italian - *''FINE MESSAGGI. <successivo> PER RIPETERE''*
  - Spanish - *''FIN DE MENSAJES (SIGUIENTE PARA REPITIR)''*
- *''GET DIAL TONE, PUSH Cover Msg Retrieval''* (English)
  - French - *''TONALITE D'ENVOI - <LECT. MESS. COUV.>''*
  - Italian - *''<rec mess copert> DOPO IL TONO DI CENTR''*
  - Spanish - *''OBTENGA TONO OPRIMA <RECUP MNSJE COBERT>''*
- *''IN PROGRESS''* (English)
  - French - *''EN COURS''*
  - Italian - *''ATTENDERE...''*
  - Spanish - *''EN CURSO''*
- *''MESSAGE RETRIEVAL DENIED''* (English)
  - French - *''LECTURE DE MESSAGES INTERDITE''*
  - Italian - *''LETTURA MESSAGGIO NON PERMESSA''*
  - Spanish - *''RECUPERACION DE MENSAJES DENEGADA''*
- *''MESSAGE RETRIEVAL LOCKED''* (English)
  - French - *''LECTURE DE MESSAGES BLOQUEE''*
  - Italian - *''LETTURA MESSAGGIO BLOCCATA''*
  - Spanish - *''RECUPERACION DE MENSAJES BLOQUEADA''*
- *''MESSAGES FOR''* (English)
  - French - *''MESSAGES POUR''*
  - Italian - *''MESSAGGI PER''*
  - Spanish - *''MENSAJES PARA''*
- *''MESSAGES UNAVAILABLE - TRY LATER''* (English)
  - French - *''MESSAGES INDISPONIBLES - REESSAYER''*
  - Italian - *''MESSAGGI TEMPORANEAMENTE NON DISPONIBILI''*
  - Spanish - *''MENSAJES NO DISPONIBLES, INTENTE DESPUES''*

- *''Message Center (AUDIX) CALL''* (English)
  - French - *''APPEL DE LA RECEPTION DE MESS. (AUDIX)''*
  - Italian - *''Chiamata dal Centro Messaggi (AUDIX)''*
  - Spanish - *''LLAMADA DEL CENTRO DE MENSAJES (AUDIX)''*
- *''NO MESSAGES''* (English)
  - French - *''PAS DE MESSAGES''*
  - Italian - *''NESSUN MESSAGGIO''*
  - Spanish - *''NINGUN MENSAJE''*
- *''WHOSE MESSAGES? (DIAL EXTENSION NUMBER)''* (English)
  - French - *''MESSAGES DE QUEL NO.? (ENTRER NO. POSTE)''*
  - Italian - *''LETTURA MESSAGGI. INTRODURRE NUMERO TEL.''*
  - Spanish - *''MENSAJES DE QUIEN? (MARCAR EXTENSION)''*

Malicious Call Trace

The following displays are associated with the "Malicious Call Trace (MCT)" feature:

- *''MALICIOUS CALL TRACE REQUEST''* (English)
  - French - *''DEPISTAGE D'APPELS MALVEILLANTS''*
  - Italian - *''RICHIESTA RINTRACCIO CHIAMATE MALEVOLE''*
  - Spanish - *''RASTREO DE LLAMADA MALINTENCIONADA''*
- *''MCT activated by: for:''* (English)
  - French - *''DAM ACTIVE par: pour:''*
  - Italian - *''RCM attivato da: per:''*
  - Spanish - *''RLM activada por: para:''*
- *''original call redirected from:''* (English)
  - French - *''redirection appel initial de: (EXTENSION)''*
  - Italian - *''chiamata iniziale rinviata da:''*
  - Spanish - *''llamada orig. transferida de:''*
- *''party: (EXTENSION)''* (English)
  - French - *''demandeur: (EXTENSION)''*
  - Italian - *''utente: (INTERNO)''*
  - Spanish - *''usuario: (EXTENSION)''*
- *''party: (ISDN SID/CNI)''* (English)
  - French - *''demandeur: (NIP/INA ISDN)''*

  - Italian - *‘‘utente: (NIC/INC ISDN)’’*
  - Spanish - *‘‘usuario: (ISDN NIE/INU)’’*
- *‘‘party: (PORT ID)’’* (English)
  - French - *‘‘demandeur: (REF. PORT ISDN)’’*
  - Italian - *‘‘utente: (ID DELLA PORTA ISDN)’’*
  - Spanish - *‘‘usuario: (ID DEL PUERTO ISDN)’’*
- *‘‘party: (ISDN PORT ID)’’* (English)
  - French - *‘‘demandeur: (REF. PORT)’’*
  - Italian - *‘‘utente: (ID DELLA PORTA)’’*
  - Spanish - *‘‘usuario: (ID DEL PUERTO)’’*
- *“END OF TRACE INFORMATION’’* (English)
  - French - *“FIN DES INFO DE DEPISTAGE”*
  - Italian - *‘‘INFORMAZIONI FINALI SUL RINTRACCIO’’*
  - Spanish - *‘‘FIN DE INFORMACION DE RASTREO’’*
- *‘‘voice recorder port:’’* (English)
  - French - *‘‘port enregistreur vocal:’’*
  - Italian - *‘‘porta del registratore:’’*
  - Spanish - *‘‘puerto de grabado de voz:’’*

Miscellaneous Attendant Features

The following displays are associated with miscellaneous attendant features.

Caller Information

- *‘‘Info:’’* (English)
  - French - *‘‘INFO.:’’*
  - Italian - *‘‘Info:’’*
  - Spanish - *‘‘INFORM:’’*

Emergency Access to Attendant

- *‘‘a=xxxxxxxxxxxxxxx Ext xxxxx xx in EMRG Q’’* (English)
  - French - *‘‘a=xxxxxxxxxxxxxxx POSTE xxxxx xx FIL URG’’*
  - Italian - *‘‘a=xxxxxxxxxxxxxxx Der xxxxx xx in C EMRG’’*
  - Spanish - *‘‘a=xxxxxxxxxxxxxxx EXT xxxxx xx EN C EMRG’’*

Queue Status

- *''HUNT GROUP <x> NOT ADMINISTERED''* (English)
  - French - *''GROUPE DE DIST. <x> NON ADMINISTRE''*
  - Italian - *''GRUPPO <x> NON AMMINISTRATO''*
  - Spanish - *''GRUPO BUSQUEDA <x> NO ADMINISTRADO''*

Queue Status Indication

- *''<15 chrs> Q-time xx:xx calls xx''* (English)
  - French - *''<15 chrs> TEMPS-F xx:xx APPELS xx''*
  - Italian - *''<15 chrs> T-coda xx:xx chiam xx''*
  - Spanish - *''<15 chrs> HORA-C xx:xx LLAMADAS xx''*

Miscellaneous Call Identifier

Table 3-7 lists displays associated with Miscellaneous Call Identifiers.

Table 3-7. Miscellaneous Call Identifiers

''English Display'' (stands for)	**''French Display'' (stands for)**	**''Italian Display'' (stands for)**	**''Spanish Display'' (stands for)**
''sa'' (ACD Supervisor Assistance)	*''AS''* (Assistance surveillant)	*''as''* (Assistenza Supervisoree)	*''AS''* (Ayuda del supervisor)
''ac'' (Attd Assistance Call)	*''AA''* (Appel assistance)	*''ao''* (Assistenza Operatore)	*''AO''* (Ayuda de operadora)
''tc'' (Attd Control Of A Trunk Group)	*''CF''* (Commande faisceau)	*''fc''* (Fascio Controllato)	*''CE''* (Control enlaces)
''an'' (Attd No Answer)	*''TR''* (Telephoniste sans reponse)	*''on''* (Operatore Non Risponde)	*''ON''* (Operadora no responde)
''pc'' (Attd Personal Call)	*''AP''* (Appel personnel)	*''cp''* (Chiamata Personale)	*''LP''* (Llamada personal)
''rc'' (Attd Recall Call)	*''RA''* (Rappel)	*''rc''* (Richiamata)	*''RL''* (Rellamada)
''rt'' (Attd Return Call)	*''RE''* (Retour)	*''rt''* (Ritornata)	*''RT''* (Retorno)

Continued on next page

Table 3-7. Miscellaneous Call Identifiers — *Continued*

''English Display'' (stands for)	**''French Display'' (stands for)**	**''Italian Display'' (stands for)**	**''Spanish Display'' (stands for)**
''sc'' (Attd Serial Call)	*''AS''* (Appel en serie)	*''ic''* (Inoltro a Catena)	*''LS''* (Llamada en serie)
''co'' (Controlled Outward Restriction)	*''RD''* (Restriction de depart)	*''cu''* (Controllata Uscente)	*''RS''* (Restriccion saliente)

Continued on next page

Table 3-7. Miscellaneous Call Identifiers — *Continued*

“English Display” (stands for)	**“French Display” (stands for)**	**“Italian Display” (stands for)**	**“Spanish Display” (stands for)**
“cs” (Controlled Station to Station Restriction)	*“RP”* (Restriction vers postes)	*“cd”* (Controllata Derivati)	*“CS”* (Control estacion)
“ct” (Controlled Termination Restriction)	*“AR”* (Restriction d’arrivee)	*“ct”* (Controllata Terminante)	*“RE”* (Restriccion entrante)
“db” (DID Find Busy Station With CO Tones)	*“OP”* (Occupation du poste)	*“po”* (Passante Occupata)	*“EO”* (Estacion occupada)
“da” (DID Recall Go To Attd)	*“RT”* (Rappel telephoniste)	*“pr”* (Richiamata su Passante)	*“RD”* (Rellamada directa)
“qf” (Emerg. Queue Full Redirection)	*“FP”* (File d’urgence pleine deviation)	*“de”* (Deviata Emergenza)	*“DE”* (Desvio de emergencia)
“hc” (Held Call Timed Reminder)	*“AG”* (Indicatif d’appel en garde)	*“at”* (Avviso Chiamata in tenuta)	*“LR”* (Recordatorio de llamada retenida)
“ic” (Intercept)	*“IN”* (Interception)	*“in”* (Intercettata)	*“IN”* (Intercepcion)
“ip” (Interposition Call)	*“AI”* (Appel interposition)	*“ip”* (Interposizione)	*“EP”* (Entre posiciones)
“ld” (LDN Calls on DID Trunks)	*“SD”* (Selection directe)	*“pd”* (Diretta Passante)	*“LD”* (Larga distancia)
“so” (Service Observing)	*“ES”* (ecoute du service)	*“is”* (Inclusione Supervisore)	*“SS”* (Supervision del servicio)

Continued on next page

Table 3-7. Miscellaneous Call Identifiers — *Continued*

‘‘English Display’’ (stands for)	‘‘French Display’’ (stands for)	‘‘Italian Display’’ (stands for)	‘‘Spanish Display’’ (stands for)
‘‘na’’ (Unanswered or Incomplete DID Call)	*‘‘SR’’* (Sans reponse)	*‘‘pn’’* (Passante Non Risposta)	*‘‘SR’’* (Sin respuesta)
‘‘ACB’’ (Automatic Callback)	*‘‘R. AUTO.’’* (Rappel automatique)	*‘‘PRN’’* (Prenotazione Automatica)	*‘‘RA’’* (Rellamada automatica)
‘‘callback’’ (Callback Call)	*‘‘RAPPEL’’* (Rappel)	*‘‘prenotaz’’* (Prenotazione)	*‘‘RELLAM’’* (Rellamada)
‘‘park’’ (Call Park)	*‘‘G. I.’’* (garde par) indicatif	*‘‘parch.’’* (Parcheggiata)	*‘‘ESTAC’’* (Estacionamiento de llamada)
‘‘control’’ (Control)	*‘‘CONTROLE’’* (Controle)	*‘‘cntr.op.’’* (Controllo Operatore)	*‘‘CONTROL’’* (Control)
‘‘ICOM’’ (Intercom Call)	*‘‘INTERCOM’’* (Intercommunicatio n)	*‘‘ICOM’’* (Intercom)	*‘‘INTERF’’* (Llamda interfono)
‘‘OTQ’’ (Outgoing Trunk Queuing)	*‘‘FFD’’* (File faisceaux de depart)	*‘‘RFO’’* (Richiamata su Fascio Occupato)	*‘‘EES’’* (Espera de enlace de salida)
‘‘priority’’ (Priority Call)	*‘‘PRIORITE’’* (Appel prioritaire)	*‘‘priorita’’* (Priorita’)	*‘‘PRIORIT’’* (Llamada prioritaria)
‘‘recall’’ (Recall Call)	*‘‘APP.RAP.’’* (Appel rappel)	*‘‘richiam’’* (Richiamata)	*‘‘REPET’’* (Rellamada)
‘‘return’’ (Return Call)	*‘‘RETOUR’’* (Retour)	*‘‘ritorno’’* (Chiamata Ritornata)	*‘‘RETORNO’’* (Llamada de retorno)
‘‘ARS’’ (Automatic Route Selection)	*‘‘SAA’’* (Selection de l’acheminement automatiqe)	*‘‘SAI’’* (Selez. Autom. Instradam.)	*‘‘SAR’’* (Seleccion automatica) de rutas)
‘‘forward’’ (Call Forwarding)	*‘‘RENVOI’’* (Renvoi)	*‘‘deviata’’* (Deviata)	*‘‘REENVIO’’* (Reenvio de llamada)

Continued on next page

Table 3-7. Miscellaneous Call Identifiers — *Continued*

''English Display'' (stands for)	**''French Display'' (stands for)**	**''Italian Display'' (stands for)**	**''Spanish Display'' (stands for)**
''cover'' (Cover)	*''SUPPL.''* (Suppleance)	*''copert.''* (Copertura)	*''COBER''* (Cobertura)
''DND'' (Do Not Disturb)	*''NPD''* (Ne pas deranger)	*''nd''* (Non Disturbare)	*''NM''* (No molestar)
''p'' (Call Pickup)	*''P''* (Prise)	*''a''* (Assente)	*''C''* (Captura de llamada)
''c'' (Cover All Calls)	*''s''* (Suppleance)	*''c''* (Copertura)	*''c''* (Cobertura de toda llamada
''n'' (Night Sta. Serv., Incoming No Answer)	*''N''* (Service nuit, entrant pas reponse)	*''n''* (Serv. Notte, Esterna Non Risposta)	*''N''* (Servicion noct. ext. no responde)
''B'' (All Calls Busy)	*''O''* (Tous occupes)	*''O''* (Tutte Occupate)	*''O''* (Todas ocupadas)
''f'' (Call Forwarding)	*''R''* (Renvoi)	*''d''* (Deviata)	*''R''* (Reenvio de llamada)
''b'' (Cover Busy)	*''o''* (Suppleance occupee)	*''o''* (Copertura per Occupato)	*''o''* (Cobertura ocupada)
''d'' (Cover Don't Answer)	*''n''* (Suppleance pas de reponse)	*''n''* (Copertura per Non Risposta)	*''n''* (Cobertura sin respuesta)
''s'' (Send All Calls)	*''E''* (Envoi tous appels)	*''r''* (Rinvio)	*''E''* (Envio de toda llamada)

Party Identifiers

Table 3-8 lists displays associated with Party Identifiers. Party Identifiers can show up in two different ways (through administration and through DCS). Identifiers administrable are not translated. Party identifiers appearing on a display, due to DCS calling, are translated.

Table 3-8. Party Identifiers Displays

Identifier	‘‘English Display’’	‘‘French Display’’	‘‘Italian Display’’	‘‘Spanish Display’’
Attendant[1]	*‘‘OPERATOR’’‘*	*‘‘TELEPHONISTE’’*	*‘‘OPERATORE’’*	*‘‘OPERADORA’’*
Conference Call	*‘‘CONFERENCE’’*	*‘‘CONFERENCE’’*	*‘‘CONFERENZA’’*	*‘‘CONFERENCIA’’*
Extension	*‘‘EXT’’*	*‘‘POSTE’’*	*‘‘DER’’*	*‘‘EXTENSION’’*
Paging[2]	*‘‘PAGING’’*	*‘‘PAGING’’*	*‘‘PAGING’’*	*‘‘PAGING’’*
Trunk Group[1]	*‘‘OUTSIDE CALL’’*	*‘‘APPEL EXT.’’*	*‘‘ESTERNA’’*	*‘‘LLAMADA EXT.’’*
Unknown	*‘‘UNKNOWN NAME’’*	*‘‘INTROUVABLE’’*	*‘‘NOME SCONOSC.’’*	*‘‘DESCONOCIDO’’*

1. These displays are administrable and appear translated if associated with a DCS call. If not associated with a DCS call, the name that appears is the name administered on the “Associated Administration” form.
2. This display is never translated.

Property Management System Interface

The following displays are associated with the "Property Management System (PMS) Interface" feature.

- *‘‘CHECK IN - Ext:’’* (English)
  - French - *‘‘ENREGISTREMENT - POSTE:’’*
  - Italian - *‘‘CHECK IN - Tel:’’*
  - Spanish - *‘‘REGISTRARSE - EXTENSION:’’*
- *‘‘CHECK IN: ROOM ALREADY OCCUPIED’’* (English)
  - French - *‘‘ENREGISTREMENT: CHAMBRE OCCUPEE’’*
  - Italian - *‘‘CHECK IN: CAMERA OCCUPATA’’*
  - Spanish - *‘‘REGISTRARSE: HABITACION OCUPADA’’*
- *‘‘CHECK IN COMPLETE’’* (English)
  - French - *‘‘ENREGISTREMENT EFFECTUE’’*
  - Italian - *‘‘CHECK IN COMPLETATO’’*
  - Spanish - *‘‘REGISTRO TERMINADO’’*

- *“CHECK IN FAILED”* (English)
  - French - *“ECHEC D’ENREGISTREMENT”*
  - Italian - *“CHECK IN ERRATO”*
  - Spanish - *“REGISTRARSE: FALLIDO”*
- *“CHECK OUT - Ext:”* (English)
  - French - *“DEPART - POSTE:”*
  - Italian - *“CHECK OUT - Tel:”*
  - Spanish - *“PAGAR LA CUENTA - EXTENSION:”*
- *“CHECK OUT COMPLETE: MESSAGE LAMP OFF”* (English)
  - French - *“DEPART: PAS DE MESSAGES”*
  - Italian - *“CHECK OUT COMPLETATO: NESSUN MESSAGGIO”*
  - Spanish - *“PAGO TERMINADO: NINGUN MENSAJE”*
- *“CHECK OUT COMPLETE: MESSAGE LAMP ON”* (English)
  - French - *“DEPART: MESSAGES”*
  - Italian - *“CHECK OUT COMPLETATO: MESSAGGI IN ATTESA”*
  - Spanish - *“PAGO DE CUENTA TERMINADO: MENSAJES”*
- *“CHECK OUT FAILED”* (English)
  - French - *“ECHEC PROCEDURE DE DEPART”*
  - Italian - *“CHECK OUT ERRATO”*
  - Spanish - *“PAGAR LA CUENTA: FALLIDO”*
- *“CHECK OUT: ROOM ALREADY VACANT”* (English)
  - French - *“DEPART - CHAMBRE INOCCUPEE”*
  - Italian - *“CHECK OUT: CAMERA NON OCCUPATA”*
  - Spanish - *“PAGAR LA CUENTA: HABITACION VACANTE”*
- *“MESSAGE LAMP OFF”* (English)
  - French - *“PAS DE MESSAGES”*
  - Italian - *“NESSUN MESSAGGIO IN ATTESA”*
  - Spanish - *“LUZ DE MENSAJE APAGADA”*
- *“MESSAGE LAMP ON”* (English)
  - French - *“MESSAGES”*
  - Italian - *“MESSAGGI IN ATTESA”*
  - Spanish - *“LUZ DE MENSAJE ENCENDIDA”*
- *“MESSAGE NOTIFICATION FAILED”* (English)

- — French - *‘‘ECHEC D’AVIS MESSAGES’’*
- — Italian - *‘‘NOTIFICA MESSAGGI ERRATA’’*
- — Spanish - *‘‘AVISO DE MENSAJE FALLIDO’’*

- *‘‘MESSAGE NOTIFICATION OFF - Ext: xxxxx’’* (English)
  - — French - *‘‘AVIS DE MESSAGES DESACTIVE - POSTE:xxxxx’’*
  - — Italian - *‘‘NOTIFICA MESSAGGI DISABIL. - Tel: xxxxx’’*
  - — Spanish - *‘‘AVISO DE MENSAJE APAGADO - EXT: xxxxx’’*
- *‘‘MESSAGE NOTIFICATION ON - Ext: xxxxx’’* (English)
  - — French - *‘‘AVIS DE MESSAGES ACTIVE - POSTE:xxxxx’’*
  - — Italian - *‘‘NOTIFICA MESSAGGI ABILITATA - Tel: xxxxx’’*
  - — Spanish - *‘‘AVISO DE MENSAJE ENCENDIDO - EXT: xxxxx’’*

Security Violation Notification

The following displays are associated with the "Security Violation Notification (SVN)" feature.

- *‘‘Barrier Code Violation’’* (English)
  - — French - *‘‘VIOLATION DU CODE D’ENTREE’’*
  - — Italian - *‘‘Violazione di codici di taglio’’*
  - — Spanish - *‘‘VIOLACIAON CONDIGO LIMITE’’*
- *‘‘Login Violation’’* (English)
  - — French - *‘‘VIOLATION DE L’ACCES A L’ADMINISTRATION’’*
  - — Italian - *‘‘Violazione di inizio di registrazione’’*
  - — Spanish - *‘‘VIOLACION CLAVE ACCESO’’*

Stored Number

The following displays are associated with the ‘Stored Number’ feature.

- *‘‘NO NUMBER STORED’’* (English)
  - — French - *‘‘AUCUN NUMERO EN MEMOIRE’’*
  - — Italian - *‘‘NESSUN NUMERO IN MEMORIA’’*
  - — Spanish - *‘‘NINGUN NUMERO ALMACENADO’’*

Stored numbers are displayed just as dialed. Numeric and touch-tone characters are not changed. Special codes appear as listed in Table 3-9.

Table 3-9. Stored Number Special Codes

‘‘English Display’’ (stands for)	‘‘French Display’’ (stands for)	‘‘Italian Display’’ (stands for)	‘‘Spanish Display’’ (stands for)
‘‘m’’ (Mark)	*‘‘M’’ (Marquer)*	*‘‘m’’ (Marcato)*	*‘‘M’’ (Marca)*
‘‘p’’ (Pause)	*‘‘P’’ (Pause)*	*‘‘p’’ (Pausa)*	*‘‘P’’ (Pausa)*
‘‘s’’ (Suppress)	*‘‘S’’ (Supprimer)*	*‘‘s’’ (Soppresso)*	*‘‘S’’ (Suprimir)*
‘‘w’’ (Wait)	*‘‘A’’ (Attendre)*	*‘‘a’’ (Attesa)*	*‘‘E’’ (Espera)*
‘‘W’’ (Indefinite Wait)	*‘‘a’’ (Attendre)*	*‘‘A’’ (Attesa)*	*‘‘e’’ (Espera)*

Time of Day Routing

The following displays are associated with the "Time of Day Routing" feature.

- *‘‘ENTER ACTIVATION ROUTE PLAN, DAY & TIME’’* (English)
  - French - *‘‘ENTRER PLAN D’ACTIVATION, JOUR ET HEURE’’*
  - Italian - *‘‘INTRODURRE PIANO DA ATTIV., GIORNO E ORA’’*
  - Spanish - *‘‘INTRODUZCA PLAN ACT DE RUTAS, DIA Y HORA’’*
- *‘‘ENTER DEACTIVATION DAY AND TIME’’* (English)
  - French - *‘‘ENTRER JOUR ET HEURE DE DESACTIVATION’’*
  - Italian - *‘‘INTRODURRE GIORNO E ORA DI DISATTIVAZ’’*
  - Spanish - *‘‘INTRODUZCA DIA Y HORA DE DESACTIVACION’’*
- *‘‘OLD ROUTE PLAN: x ENTER NEW PLAN:’’* (English)
  - French - *‘‘ACHEMINEMENT ANT.: x ENTRER NOUVEAU:’’*
  - Italian - *‘‘INSTRADAMENTO PREC: x INTROD IL NUOVO:’’*
  - Spanish - *‘‘PLAN RUTAS ANT: x INTRODUZCA EL NUEVO:’’*
- *‘‘OLD ROUTE PLAN: x NEW PLAN: y’’* (English)
  - French - *‘‘ACHEMINEMENT ANT.: x NOUVEAU PLAN: y’’*
  - Italian - *‘‘INSTRADAMENTO PREC: x NUOVO PIANO: y’’*
  - Spanish - *‘‘PLAN RUTAS ANT: x NUEVO PLAN: y’’*
- *‘‘ROUTE PLAN: x FOR yyy ACT-TIME: zz:zz’’* (English)

— French - *"ACHEM.: x POUR yyy ACT-HEURE: zz:zz"*

— Italian - *"INSTRADAMENTO: x PER yyy ATTIV ORE:zz:zz"*

— Spanish - *"PLAN RUTAS: x PARA yyy HORA-ACT: zz:zz"*

- *"ROUTE PLAN: x FOR yyy DEACT-TIME: zz:zz"*(English)

— French - *"ACHEM.: x POUR yyy DESACT-HEURE: zz:zz"*

— Italian - *"INSTRADAM.: x PER yyy DISATTIV ORE:zz:zz"*

— Spanish - *"PLAN RUTAS: x PARA yyy HORA-DESACT:zz:zz"*

For the above displays, x and y denotes the Route Plan Number (RPN 1-8), ***yyy*** is a three letter abbreviation for the day of the week and ***zz:zz*** is the activation time (military time). The three-letter abbreviations for the day of the week are listed in Table 3-10.

Table 3-10. Three Letter Abbreviation for Days of Week

"English Display"	**"French Display"**	**"Italian Display"**	**"Spanish Display"**
"Mon"	*"LUN"*	*"Lun"*	*"LUN"*
"Tue"	*"MAR"*	*"Mar"*	*"MAR"*
"Wed"	*"MER"*	*"Mer"*	*"MIE"*
"Thu"	*"JEU"*	*"Gio"*	*"JUE"*
"Fri"	*"VEN"*	*"Ven"*	*"VIE"*
"Sat"	*"SAM"*	*"Sab"*	*"SAB"*
"Sun"	*"DIM"*	*"Dom"*	*"DOM"*

— To enter the day of the week, the user dials 1 for Sunday, 2 for Monday, and so on.

Considerations

Each DCP/ISDN-BRI user and each attendant can select the language of their choice. In other words, one user can view his or her call related messages in Italian and another user can view his or her call related messages in English. Language selection is made via administration (English is the default language). Once the language is selected and administered on the "Station/Attendant" form, all display messages (except those that are administered, for example, station and group names) are in the language selected.

Users of a 32-character display set do not have the option of choosing a display language. These sets (in particular, the hybird MERLIN 7315H and the 7317H sets) default to English.

One additional language display message set can be added to the system. This allows the flexibility of one additional language (in addition to those already provided by the system) in which a user can view his or her call related messages. This user customized language display message set is entered into the system via administration by either a customer or an AT&T in-country partner and is accessed by a 40-character display user as their display language preference by selecting the display language option, user-defined.

If user-defined is selected as the display language preference and the *user-defined* language display message set has not been entered into the system, all display messages appear as a string of asterisks.

Interactions

See the ‘‘Feature Displays’’ section above for the details about the displays associated with specific features.

Administration

The ‘‘Attendant’’ form and the ‘‘Station’’ form (if the station is equipped with a forty-character display) ask for the display language preference. The choices are English (the default), French, Italian, Spanish, and user-defined.

The addition of the user customized language display message set to the system (the user-defined language display message set) is accomplished by translating the English language display message set into another language via administration. This process is done either by a customer or an AT&T in-country partner. To select this language display message set as a user’s display language, set the display language option to user-defined. If user-defined is chosen as the display language preference, and a *user-defined* language display message set has not been entered into the system, all display messages appear as a string of asterisks.

Hardware and Software Requirements

None required.

Administration Without Hardware (AWOH)

Feature Availability

Administration Without Hardware is available with all Generic 3 releases.

Description

Provides the ability to administer “Station” forms without specifying a port location. Such stations are referred to as *phantom* and do not generate alarms and errors when the station is translated but not yet installed. The "Administration Without Hardware (AWOH)" feature works for administration the same way as administration with hardware translation does. For example, when terminals are moved, user-activated features such as call forwarding and send all calls are preserved and functional.

An AWOH station is considered *disassociated* when no hardware ports are assigned to the station. Once a port is assigned, the AWOH station is considered *associated.*

The "Administration Without Hardware (AWOH)" feature supports the following applications:

- The ability to ‘Administer Station’ forms without specifying a port location.
- The ability by use of a phantom extension to provide call coverage (including AUDIX) for users who do not have stations physically located on the switch.
- The ability to use phantom extensions for ACD Dialed Number Identification Service (DNIS). This application allows a phantom extension to be administered on the switch for each call type that needs to be identified to ACD agents. The phantom extension is either *Call Forwarded* (via an attendant console) to an ACD split, or its coverage path is defined to include the ACD split. The “Name” field that is administered for the phantom extension will identify to the ACD agent which service the caller is attempting to reach, allowing the agent to properly address the caller.
- The ability to store station templates that can later be used with the **duplicate station** command when implementing many ‘Station’ forms of the same type in the system.

NOTE:
G3rV1, G3V2, and later releases offer two methods for joining an AWOH translation to a port location. The terminal can be administered to include a valid port location (this method is also available for G3iV1) or, for G3rV1, G3V2, and later releases, the

Terminal Translation feature can be used. For more information see the "Terminal Translation (with Security Measures)" feature.

Considerations

The primary use of the "Administration Without Hardware (AWOH)" feature is to streamline system initializations, major additions, and rearrangement/changes by allowing voice terminal translations to be entered before the actual ports are assigned. Port assignments can be done at a later time, as required.

For G3i prior to V2, use of this feature is limited to voice-only terminal types including analog, Digital Communications Protocol (DCP) 7400D series of terminals, and hybrid terminal types.

For G3r prior to V2, use of this feature is limited to analog, Basic Rate Interface (BRI) sets, certain data modules, attendants, queue warning ports, DCP (7400D series of terminals), and hybrid terminal types.

For G3V2 and later releases all terminal types mentioned above are supported.

Interactions

Unless otherwise stated, AWOH terminals act like busy terminals. This section describes how feature interact if association or disassociation is performed while a feature is active.

- G3iV1 and G3vsV1/G3sV1 supports a limited form of AWOH as described in ‘‘Considerations’’ above but does not support TTI. Association/disassociation gives a port to a station formerly translated with an ***x*** in the “Port ID” field.
- For G3r and G3V2 and later releases, see the ‘Terminal Translation Initialization (TTI)’ feature for more information on interactions.

Voice Terminal Interactions

The following are descriptions of how voice terminal features interact with the "Administration Without Hardware (AWOH)" feature.

- Abbreviated Dialing

  Abbreviated dialing used to call a station administered without hardware behaves as a normally dialed station-to-station call to an extension administered without hardware. A station with abbreviated dialing that becomes disassociated will not lose any entries in its lists.

- ACD

  A voice terminal user may transfer an ACD call to a bridged appearance of an AWOH station, put the call on hold, and/or receive another call. A bridged appearance of an AWOH extension cannot be logged into an ACD split and cannot receive calls directed to a hunt group to which it belongs.

- Automatic Callback

  Attempting automatic callback to any station administered without hardware translation sends an intercept tone to the caller, indicating that the feature cannot be activated.

- Bridged Call Appearance

  A normal station with a bridged call appearance can originate a call on the bridged call appearance even if the primary extension associated with the bridged appearance has been administered without hardware translation. Bridged call appearance buttons are lit on a normal set when the bridged appearance has been administered without hardware translation. Calls can be made to the bridged call appearance of terminals administered without hardware translation. A station without hardware translation may contain bridged call appearances.

  A bridged appearance of an AWOH extension cannot be logged into an ACD split and cannot receive calls directed to a hunt group to which it belongs.

- Busy Verification of Terminals and Trunks

  A terminal administered without hardware translation will appear to be out of service when busy verification of terminals and trunks is attempted on that terminal.

- Call Coverage

  Stations administered without hardware translation interact with "Call Coverage" feature as if all their call appearances were busy. A disassociated station can have call coverage active. The call coverage groups referred to are:

  — coverage answer groups

  — hunt groups

  — intercom groups

  — pickup groups

  — terminating extension groups

- Call Forward

  A station administered without hardware translation can have call forwarding activated while in a disassociated state. When the station is associated, call forward is active and must be turned off by the user. Association of the station turns on the call forward light.

- Call Park

  A call to a station administered without hardware translation can only be parked if the primary extension of that station is a bridged call appearance on a normal station. When a call is parked from a bridged call appearance, it is parked on the primary extension assigned to the bridged call appearance.

- Call Waiting Termination

  Call waiting termination can be administered on a single-line terminal administered without hardware translation, but, the caller receives a busy signal, as long as the station is disassociated.

- Conference

  No direct conferencing of stations without hardware translation is allowed. A bridged appearance button can be used to make conference calls.

- Customer-Provided Equipment (CPE) Alarm

  If a CPE alarm is activated for a terminal without hardware translation, no equipment is available to ring or light.

- Data Buttons

  Data buttons are not lit for data modules administered without hardware translation.

- Display

  The display for calls originating/terminating from a bridged call appearance for stations administered without hardware translation, is the same as for normal bridging.

- Facility Busy Indication

  Normal stations can have busy indicator lights for stations administered without hardware but the lights are not lit.

  The reason for this particular interaction is that Facility Busy Indication is designed to indicate if a station is off or on-hook. Since a station without hardware translation is always on-hook (even if it is treated as busy), any busy indicator lights pointing to it are not turned on.

  When a port is assigned to the AWOH station, busy indicator lights function the same as for a normal station. Busy indicator lights can be administered on stations administered without hardware translation.

  The operations invoked by pressing a facility busy indicator button do not deviate from the current operation of the feature. That is, depressing the button rings the station.

- Incoming Destination

  If the incoming destination is a station without hardware, the caller hears a ringback tone from the central office. Incoming destination calls are routed based on features active for the station, such as, call forwarding and call coverage.

- Leave Word Calling

  A station administered without hardware can be left a leave word calling message.

  A leave word calling message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension number assigned to the bridged call appearance. When a user calls a primary extension and activates Leave Word Calling, the message is left for the primary extension even if the call was answered at a bridged call appearance.

- Manual Message Waiting

  If Manual Message Waiting is activated, and the receiving terminal is administered without hardware translation, then only the originating terminal's light goes on. The status of this light is held by the receiving terminal while in a disassociated state, and is turned on upon association.

- Manual Signaling

  If the receiving terminal is administered without hardware translation, Manual Signaling is denied. The status light for the Manual Signaling button at the originating voice terminal flutters briefly to indicate the denial.

- Personal Central Office Line

  Stations administered without hardware translation can be administered with the "Personal Central Office Line (PCOL)" feature. If a call is terminated at a station administered without hardware translation and it has no coverage, the caller receives ringback (no answer). Otherwise, the call is routed to coverage.

- Priority Calling

  A priority call terminating to a station administered without hardware translation hears a busy signal.

- Send All Calls

  Send all calls can be active on a station administered without hardware translation.

- Station-to-Station Call

  A call attempting to terminate on an extension administered without hardware translation is treated like a call to a station with all call appearances busy.

- Transfer

  See ‘‘Station-to-Station Call’’ and ‘‘Call Coverage.’’

Attendant Interactions

Below are descriptions of how active attendant features interact with the AWOH characters.

- Attendant Group

  If all attendants of the group are administered without hardware, the caller receives ringback tone indefinitely for all calls made internally. Attendants administered without hardware translation will behave the same as stations without hardware in group interactions.

  Responses are similar for these features:

  — Centralized Attendant Service (CAS)

  — Inter-PBX Attendant Service (IAS)

  — Attendant Call Waiting (cannot be activated on a call to a terminal administered without hardware translation)

  — Attendant Direct Extension Selection with “Busy Lamp” field

  — Attendant-to-Station Call (the attendant is not allowed to extend a call to a terminal administered without hardware translation)

- Emergency Access to Attendant

  If emergency access to the attendant is activated while all attendants are administered without hardware translation, and there is no backup extension or the backup extension is also administered without hardware translation, the originating party will receive a busy signal. This is the same situation as an attendant queue being full and no backup extension.

- Interposition Calling (Attendant to Attendant)

  Using the individual attendant access extension, as opposed to the group, the caller hears a busy signal for all attendants administered without hardware translation.

- Night Station Service

  If a night console is not assigned or not operational and an attendant activates the night station service feature, response to the caller depends on the administration of the destination endpoint. For instance, if the endpoint is a station administered without hardware translation, the caller hears a busy signal. This response is governed by the endpoint rather than the feature.

- World Class Attendant

  This feature and its two subfeatures apply.

  — Attendant Override

  If an attendant activates Attendant Override (and therefore bypasses coverage), the attendant hears a busy signal for calls to terminals administered without hardware.

  — Attendant Serial Calling

  Attempting to extend a call to a terminal administered without hardware translation causes a busy signal.

Data Terminal Interactions

Below are descriptions of how active data terminal features interact with the "Administration Without Hardware (AWOH)" feature

- Data Call Setup

  Three methods are provided to set up a data call:

  — Data Terminal Dialing — An attempt to use keyboard dialing to terminate a call to a data endpoint administered without hardware translation causes a BUSY message on the screen, indicating that the terminal is in use, out of service, or administered without hardware translation.

  — Voice Terminal Dialing — see ‘‘Station-to-Station Call’’ section.

  — Other Devices — [like a modular processor data module (MPDM) equipped with an automatic calling unit (ACU) interface module] Depending on the attached hardware, the call is handled like a data call.

  — Voice Terminal Dialing.

- Hunt-Group (UCD/DDC) See ‘‘Coverage’’ section.
- Incoming Destination

  See ‘‘Incoming Destination’’ section
- Administered Connections

  Endpoints for ACs can be administered without hardware translation.
- Terminal-to-Data Module Call

  If the data endpoint has been administered without hardware translation, the result is either be a printed BUSY on the screen or a busy signal, depending on the hardware associated with the terminal originating the call. See ‘‘Station-to-Station Call’’ section.

Association/Disassociation of Terminal Feature Interactions

This refers to whether a voice terminal has been assigned a port location. Association/Disassociation interactions are described next. Some of these functions are limited to multibutton sets.

- Automatic Callback

  If a station becomes disassociated while a normal station has automatic callback activated for that station, the normal station's automatic callback light is turned off and the sequence is broken.

- Bridged Call Appearance

  If a station has a bridged call appearance of another station, which is off-hook, the station administered with the bridged call appearance can disassociate at any time and not disrupt the call in progress on the bridge.

  If a station with a bridged appearance of a normal station associates itself while the extension for the bridged appearance is on a call, the station associating itself can join the call after it has completed the association sequence.

  Disassociation cannot be performed from a bridged call appearance. Disassociation (TTI) of a station must be performed from the port on which the set resides. For example, the port from which disassociation is performed, must match the port for the extension that is to be disassociated.

- Call Coverage

  If a station disassociates while Send-all-calls or Go-to-coverage is active, then these features remain active while the station has no hardware translation.

- Call Coverage Answer Group

  Members in a group cannot be disassociated while there is an incoming call to the group. In this case all sets are seen as active and therefore cannot be disassociated.

  If any endpoint previously administered without hardware translation is inserted into translation via TTI or station administration, that endpoint is excluded from all transactions taking place in the call coverage answer group. This means that a member cannot join the other members of an incoming call for one that is taking place during the binding of the station to the port location. The member can join in all subsequent calls to the group.

- Call Forward

  A station can disassociate while call forwarding is active. If a destination extension for call forwarding disassociates, call forwarding to that extension remains active.

- Call Park

  A terminal is not allowed to be disassociated while placed in a parked state by another set. It is virtually put on hold. If a normal station parks an incoming call, the station that parked the call can disassociate itself after the call is parked and still perform the retrieval sequence from another voice terminal.

- Call Pickup

  The primary extension of a call pickup group cannot disassociate while a call is being attempted. All secondary members are allowed to disassociate while a call is ongoing to a primary extension.

If a call is ongoing to any extension in the group, any member of the group can disassociate. Any member of the group can associate. That member will not join the group for the call that is currently in progress, but is available for all subsequent calls to the group.

- CPE Alarm

  If a station that has been administered with a CPE alarm enters the system translation while an alarm is active, then the newly associated station receives the alarm indication upon entering the system.

- Hunt Group UCD/DDC

  Any member of a hunt group that is not the target of the current call (not ringing) can disassociate itself. The primary, or ringing, extension cannot.

  If any endpoint previously administered without hardware translation is inserted into translation via TTI or station administration, that endpoint is excluded from all transactions taking place in the hunt group. This means that a member does not become available to take an incoming call that was already taking place during the binding of the station to the port location. The member is available for subsequent calls to the group.

- Hold

  A station that is on hold or has put another call on hold cannot undergo disassociation.

- Incoming Destination

  See ‘‘Station-to-Station Call’’ section

- Intercom Group - Auto/Dial

  See ‘‘Station-to-Station Call’’ section

- Message Light

  All messages needn’t be deleted before disassociation. If a station receives messages while it is in the disassociated state, when the terminal receives hardware translation, the message light is updated.

- Send All Calls

  Send all calls remains activated when a station becomes disassociated.

- Station-to-Station Call

  No disassociation can be performed while a call is in progress. No disassociation can be performed while a set is ringing.

- Terminating Extension Group

  Members in a group cannot be disassociated while there is an incoming call to the group. This is because all members of the group are seen as active, since they are ringing and busy indicator lights are flashing.

  If any endpoint previously administered without hardware translation is inserted into translation via TTI or station administration, that endpoint is excluded from all transactions taking place in the terminating extension group. This means that a member cannot join the other members of an incoming call for one that is taking place during the binding of the station to the port location. The member can join in all subsequent calls to the group.

- Transfer

  After a connection has been made from first to third party, the second party, the one who performed the transfer, can be disassociated. Parties one and three are treated as a station-to-station call.

- Attendant

  Because the attendant is a central focus for incoming calls, it is advisable to have the attendant in position busy mode so as to route incoming calls. Attendant Priority Queuing, which a World Class Core feature, provides that individual attendant access and interposition calling are not reclassified and directed to the attendant group. These calls queue for the individual attendant and prevent the attendant from disassociating. To prevent the attendant from attempting to disassociate while in position-available mode, calls queued, held, or seen as active for the attendant, will prevent disassociation.

- Incoming Destination

  An incoming destination is not allowed to be disassociated while a call is in progress to that extension, regardless of whether the extension is a station or an attendant.

- Attendant Night Service

  The night service station cannot be removed while in night service. For the removal of any endpoint administered without hardware translation while night service is activated, see ‘‘Night Service’’ section.

- Attendant Release Loop Operation

  All calls held with the release loop operation by the attendant are reclassified as attendant group calls if the attendant disassociates before the attendant timed reminder interval expires.

- Night Service-Trunk Group

  If the night service destination is a station, disassociation operates the same way as disassociation for stations. See “Station-to-Station” section. If the night service destination is an attendant, disassociation operates the same way as disassociation for attendants. See “Attendants”.

World Class Attendant

Below are ‘World Class Attendant’ features that interact with AWOH.

- Serial Calling

  In a serial call, the attendant is not in a busy state after releasing a call. In this situation, the attendant is allowed to perform disassociation. If the attendant is disassociated upon return from a call that has been extended to a station, the call is reclassified as an attendant group call and is routed to the group.

- Attendant Return Call

  If the attendant is disassociated upon return from a call that has been extended to a station, the call is reclassified as an attendant group call and is routed to the group.

Data Modules

This section describes association/disassociation of data terminals. Data modules can be associated/disassociated by:

— Data Terminal Dialing

— Voice Terminal Dialing

— Other Devices - this includes using a default set type to make the association, and then removing the default set type and replacing it with the proper data endpoint.

Since DTDM's reside on certain station types, the port is automatically inherited from the host station. The DTDM receives its port identification when the station is associated/disassociated.

- Hunt Group (UCD/DDC)

  See ''Hunt Groups'' section.

- Incoming Destination

  See ''Incoming Destination'' section.

- Administered Connections

  If an administered connection is administered without hardware translation the system attempts to establish a connection only when both endpoints are associated with hardware translation.

  AC can be disassociated by changing the port for the data module to an X, either by administration or via TTI.

- Terminal to Data Module Call

  See ''Station-to-Station Call'' section.

- Transfer

  See ''Transfer'' section.

Administration

AWOH is administered on a per-voice terminal basis by the System Manager. Normal station administration is required with the exception of entering an X in the “Port” field to indicate that there is no hardware associated with the station.

Four maintenance commands can be executed on terminals administered without hardware translation:

- Busy-out extension object
- Release extension object
- Status extension object
- Test extension object

In all cases, the message: `hardware not administered` is displayed on the G3-MT.

Hardware and Software Requirements

No additional hardware or software is required.

Advice of Charge

Feature Availability

Advice of Charge (AOC) is available with G3V4 and later releases. It is only available in France, Germany, Australia, and countries with public networks running European Telecommunication Standards Institute (ETSI) compatible ISDN-PRI. It is not available in the United States.

Description

AOC collects charge advice information from the public network for each outgoing call and enters the information in that call's Call Detail Record (CDR). Charge advice is a number representing the cost of a call; it is recorded as either a charging or currency unit. AOC is the ISDN equivalent of Periodic Pulse Metering (PPM).

AOC is only available for outgoing calls placed on an ISDN-PRI trunk. It is not available for incoming calls.

The system administrator can use AOC information to account accurately for the cost of outgoing calls without waiting for the next bill from the local telecommunications provider. This is especially important in countries where telephone bills are not itemized, but contain only the total number of units used. AOC information can also be used to let employees know the cost of their phone calls, thereby encouraging employees to manage the company's telecommunications expenses. Note, however, that AOC information cannot necessarily be relied upon to dispute telephone bills with the network service provider.

The type of AOC information that is received, the way in which the information is received, and the switch administration required to receive AOC information, all vary from country to country. For example:

- In some countries AOC is only received at the end of a call. In others it can accumulate during a call as well.
- In some countries, users must subscribe to AOC from the public network. In others it is available simply by administering the switch to receive AOC information.
- In some countries once subscribed to, AOC information will be received automatically for each call. In others, the system must request AOC information for each call.

See the country application notes or your AT&T representative for country-specific parameter settings for the "Advice of Charge" feature.

When Information Is Received

Depending upon what is available in any given country, and how the DEFINITY switch is administered, AOC information can be received either during and at the end of a call, or only at the end of a call. The following sections describe these options.

During and End

With this option, the public network provides AOC information periodically throughout a call and at the end of the call. Cumulative charge values are stored and the final value received is reported on the call CDR. For the ‘CDR Call Splitting’ feature to work properly, updates during a call are required. However, periodic updates increase message activity on the signaling channel and may reduce the maximum call capacity of the DEFINITY Communications System. This is especially true in countries such as Germany and France where the network sends charging information updates as often as every 3 to 10 seconds for each active international call. See "CDR Call Splitting" later in this description for more information about AOC with the ‘CDR Call Splitting’ feature.

End Only

With this option, the public network provides AOC information for a call when the call is dropped. The value is reported on the call CDR. AOC only at the end of the call eliminates the performance cost due to extra ISDN-PRI messaging on the signaling channel associated with periodic updates. However, if the ‘CDR Call Splitting’ feature is enabled, the CDR record associated with the final party on the call will contain the AOC for the entire call. This is true because no charge advice data is available until the final party drops off the call. The charge advice in all earlier CDR records will be zero. See the "CDR Call Splitting" feature for more information about the "Advice of Charge" with the ‘CDR Call Splitting’ feature.

AOC in CDR Reports

Because AOC information is cumulative, it is assumed that each succeeding value will be larger than the previous value. If a charge update is received that is smaller than the previous value, the new smaller value is ignored.

If an ISDN-PRI trunk group has CDR enabled, and has been administered to expect AOC information, and if the selected CDR output format contains an “ISDN Call Charge (ISDN CC)” field, then the “ISDN Call Charge” field will contain the following information:

- If the Call Splitting or Attendant Call Recording feature is enabled with AOC being received during the call, and if a CDR record is generated because a call has been transferred for the first time, the “ISDN Call Charge” field contains the cumulative charge most recently received from the network. For all subsequent transfers, the “ISDN Call Charge” field contains the difference between the cumulative charge most recently received and the value generated in the previous CDR record for the same call.

- When a CDR record is generated because a call is dropping, the "ISDN Call Charge" field contains the last cumulative charge received from the network.

Four CDR report formats support AOC:

- Enhanced 24-word Standard ASCII unformatted record (unformatted)
- Enhanced 24-word Standard ASCII expanded record (expanded)
- International ISDN Expanded record (int_isdn)
- Customized record (customized)

A zero appears in the "Call Charge" field when: no AOC information is received; a value of zero is the last charge information received; or the outgoing trunk group is not administered for AOC.

Reported Units

The public network may send AOC information either as charging units or currency value. This charging unit or currency encoding is country dependent. The DEFINITY Communications System does not differentiate the stored charge as currency or charging unit.

Considerations

The DEFINITY Communications System does not tandem AOC information via ISDN-PRI messages through a private network to other DEFINITY switches. Therefore, the CDR adjunct that records AOC information must receive its input from the DEFINITY system directly connected to the public network.

Interactions

- Call Forwarding — Off Net

  AOC for a call to a station whose calls are forwarded over a public-network PRI trunk is charged to the forwarding station, not the calling station.

- Call Transfer

  If a transferred call is routed over a public-network ISDN-PRI trunk group, AOC administration for the outgoing trunk group controls whether AOC information is requested or recorded for the call. If two or more outgoing trunks are connected together via trunk-to-trunk transfer, the DEFINITY Communications System may receive AOC information from the network for each outgoing trunk involved in the call.

- CDR Call Splitting

  If Call Splitting is enabled, when an outgoing call is transferred, a CDR record is issued for the initial portion of the call and the “ISDN Call Charge” field reports the AOC information received thus far. Subsequent AOC information received from the network for the outgoing call is charged to the party remaining on the call until the call is dropped or transferred again. Attendant Call Recording, a form of Call Splitting, generates a CDR record when an attendant drops from a call.

  Users who rely on Call Splitting or Attendant Call Recording should subscribe to or request AOC information during the call in order to have correct AOC information recorded for each party that participated in the call. However, this increases message activity on the signaling channel and reduces Busy Hour Call Capacity of the DEFINITY system.

  In some countries, or with specific protocols, AOC information during a call is not available. However, the Elapsed Time in the CDR records can be used to allocate the AOC among the parties on the call.

  For more information about Call Splitting and its interactions see the "Call Detail Recording (CDR)" feature.

Administration

Administration of AOC is dependent upon the country where the feature is being used. See *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for general instructions for administering AOC. See the country application notes or AT&T representative for country-specific instructions for administering AOC.

Hardware and Software Requirements

A CDR adjunct or printer is required to record AOC information. No additional software or hardware is necessary.

Agent Call Handling

Feature Availability

Agent Call Handling available with all Generic 3 releases. ACD software is required.

Description

Provides ACD agents with the capabilities required to answer and process ACD calls.

The agent capabilities provided by this feature are:

- Agent Log-In and Log-Out
- Agent Answering Options
  - Automatic Answer (zip tone)
  - Manual Answer
- ACD Work Modes
  - Auxiliary Work Mode
  - After Call Work
  - Auto-In
  - Manual-In
- Agent Request for Supervisor Assistance
- ACD Call Disconnecting (Release button)
- Stroke Counts
- Call Work Codes
- Forced Entry of Stroke Counts and Call Work Codes

This feature description also explains Agent Sizing.

NOTE:
All of the agent capabilities listed above are also supported through the CallVisor ASAI. For information on CallVisor ASAI, consult the "CallVisor Adjunct/Switch Applications Interface (ASAI)" feature.

Agent Log-in and Log-out

The information below applies generally to traditional ACD. See the "Expert Agent Selection (EAS)" feature for additional EAS procedures.

To receive ACD calls, the agent must log into the system. An agent logging into a split automatically enters the Auxiliary Work mode (described later) for that split. An agent can be logged into multiple splits. An agent will be required to enter a log-in identification number when logging in if the hunt group is measured via CMS or BCMS. If the hunt group is not measured, entry of a login ID is optional. Entry of a login is also optional with BCMS.

To log in, an agent must go off-hook and dial the log-in feature access code, followed by the two-digit split group number (three-digit split group number for G3r) and the log-in identification number (if required). If the log-in procedure is successful, the agent enters the Auxiliary Work mode and the lamp associated with that split's Auxiliary Work button, if provided, lights steadily on the agent's terminal and the agent hears confirmation tone. At the same time, the system sends two messages to the CMS or BCMS (if it is a measured split): a message that the agent has logged in (including the identification number) and a message that the agent has entered the Auxiliary Work mode.

If, during the log-in process, any of the following situations occur, the log-in attempt is canceled and the agent receives Intercept Treatment.

- The agent dials an invalid log-in feature access code.
- The agent dials an invalid split group number (that is, the agent dials the number of a split that does not exist or of one to which the agent is not assigned).
- The agent is already logged into the maximum number of splits. In this case, Intercept Treatment is received after dialing the split group number.
- The agent dials a split group number for a split that he or she is already logged into.
- The agent dials the wrong number of digits.

The agent should log out when he or she leaves his or her position for an extended period of time (such as the end of a shift or when changing voice terminals) and is therefore unavailable for ACD calls. If an agent logs out and is administered for measurement by CMS or BCMS, a message is sent to the BCMS/CMS so that it no longer measures the agent's status. If an agent is logged into more than one split, he or she should log out of each split. AUX work mode is typically used for temporary situations when BCMS/CMS tracking of the Aux-Work time is desired. Otherwise, log out.

To log out of a split, the user goes off-hook and dials the log-out feature access code followed by the split group number. If the log-out attempt is successful, the agent hears confirmation tone and all lamps associated with work mode buttons (described later) go dark. If the agent is logged into more than one split, logging out of one split does not affect the state of the other split.

If, during the log-out process, any of the following situations occur, the log-out attempt is canceled, and the agent receives Intercept Treatment.

- The agent dials an invalid log-out feature access code.

- The agent dials an invalid split group number.
- The agent dials a split group number for a split that he or she is not logged into.

If an agent is in the Automatic Answer mode (described later) and is using a handset, the agent can log out simply by hanging up. (This does not mean pressing the release button on a Call Master voice terminal.) If an agent in the Automatic Answer mode is using a headset instead of a handset, the agent can log out by turning off the headset. This does not apply to quick-disconnect. If this method is used to log out, the agent is automatically logged out of all splits that he or she has logged into.

If calls are in the split queue, the last available agent in a non-vector controlled split can still log out of the split by dialing the log-out feature access code.

Agent Answering Options

An agent can answer ACD calls by using either a headset, handset, or speakerphone. An agent can be assigned one of two answering options: Automatic Answer or Manual Answer.

Automatic Answer

The following procedure applies to traditional ACD and EAS environments.

An agent assigned to Automatic Answer is connected directly to incoming calls without ringing. Instead of the usual process where an agent receives ringing and then goes off-hook and answers the call, the agent hears zip tone through the headset, handset, or speakerphone and is automatically connected to the incoming ACD call.

It is recommended that Automatic Answer be used with a headset. In this case, the agent hears zip tone through the headset and is then automatically connected to the call. (If the incoming trunk group is data restricted, the zip tone is not heard. If the agent’s extension is data restricted, the zip tone is not heard. A headset user should not be assigned data restriction.)

Although possible, it is not recommended that a handset or speakerphone be used with Automatic Answer. In order for an agent with Automatic Answer and a handset or speakerphone to answer an ACD call, the handset or speakerphone must be off-hook (handset lifted or speakerphone turned on) at all times. While off-hook, the agent hears zip tone through the handset or speakerphone.

NOTE:
Automatic Answer can be administered to apply only to ACD calls or to apply to all calls terminating to the agent’s set. If all calls terminating to the set are automatic answer and if the agent receives direct extension calls, he or she should always activate Call Coverage, Call Forwarding or Send All Calls when leaving his or her position and make himself or herself

unavailable for ACD calls (by logging out or entering AUX work mode) so calls will not terminate to an unmanned station.

Manual Answer

An agent is assigned to Manual Answer hears ringing, and then goes off-hook to answer the incoming call. If the agent does not go off-hook, the call will continue ringing. The agent can use either a headset, handset, or speakerphone to answer the call.

ACD Work Modes

This information generally applies to a traditional ACD environment. See the "Expert Agent Selection (EAS)" feature for more specific EAS procedures. If the "Multiple Call Handling" feature is enabled, it will affect when agents can enter different work modes and when calls will be delivered to agents in Manual-In or Auto-In work modes. See the "Multiple Call Handling" feature for detailed information.

At any given time, an agent can be in one of four work modes. An agent can change work modes at any time. If an agent is not active on a call or does not have a call on hold, the mode change is immediate. However, if an agent tries to change modes while he or she is active on a call or has a call on hold, the mode is not changed until the agent is disconnected from the calls. An agent can change modes by using either button or dial access. The four work modes are described in the following paragraphs.

- Auxiliary Work
- Auto-In
- Manual-In
- After Call Work

Auxiliary Work Mode: An agent should enter the Auxiliary Work mode for a particular split whenever he or she is doing non-ACD activities such as taking a break or going to lunch. This makes the agent unavailable for ACD calls to that split (and the agent is not in the most idle agent [MIA] queue), but BCMS/CMS tracking of the agent continues.

When an agent logs into a split, he or she automatically enters this mode for that split. To change to the Auxiliary Work mode while in another mode, the agent can dial the feature access code for the Auxiliary Work mode followed by the split group number or can press the Auxiliary Work button for that split. If the attempt to change modes is successful and the agent has no active or held calls, the lamp associated with the Auxiliary Work button lights steadily and the BCMS/CMS is informed of the mode change for that agent. If the attempt to change modes is successful, but the agent has any active or held calls, the lamp flashes until all calls are dropped, at which point the AUX lamp lights steadily and the BCMS/CMS is informed of the agent’s state change. The attempt is canceled and the agent receives intercept treatment if the agent:

- Tries to enter the Auxiliary Work mode for an invalid split
- Tries to enter the Auxiliary Work mode for a split of which he or she is not a member
- Dials an invalid feature access code

If an agent is the last agent logged into a non-vector controlled split and calls are in queue for that split, the agent cannot enter the Auxiliary Work mode until the queued calls are handled. Refer to the considerations section near the end of this feature description for more details. An attempt to enter the Auxiliary Work mode under these conditions prevents new calls from entering the queue for that split.

Once an agent has entered the Auxiliary Work mode for a particular split, the agent is no longer available to answer other ACD calls to that split. However, the agent may be available for ACD calls to other splits that the agent is logged into depending on the agent's state in those splits and the agent is still available for non-ACD calls. The BCMS/CMS is notified whenever an agent in the Auxiliary Work mode receives an incoming non-ACD call or makes an outgoing call.

NOTE:
Agents in vector-controlled splits can go into AUX work even if they are the last agent and calls are queued to that split.

Auto-In Mode: When an agent enters the Auto-In mode, he or she, upon disconnecting from an ACD call, automatically becomes available for answering new ACD calls.

To change to the Auto-In mode while in another mode, the agent can dial the feature access code for the Auto-In mode followed by the split group number or can press the AUTO-IN button for that split. If the attempt to change modes is successful, the lamp associated with the AUTO-IN button lights steadily and the BCMS/CMS is informed of the mode change for that agent. If the attempt to change modes is successful, but the agent has any active or held calls, the lamp flashes until all calls are dropped, at which point the lamp lights steadily and the BCMS/CMS is informed. If the agent tries to enter the Auto-In mode for an invalid split or for a split of which he or she is not a member, or if the agent dials an invalid feature access code, the attempt is canceled and the agent receives intercept treatment.

Manual-In Mode: When an agent enters the Manual-In mode, he or she, upon disconnecting from an ACD call, automatically enters the After Call Work mode (described later) for that split, and is not available for any ACD calls. The agent must then manually reenter either the Auto-In mode or Manual-In mode to become available for ACD calls.

To change to the MANUAL-IN mode while in another mode, the agent can dial the feature access code for the Manual-In mode followed by the split group number or can press the MANUAL-IN button for that split. If the attempt to change modes is successful, the lamp associated with the MANUAL-IN button lights steadily and the

CMS is informed of the mode change for that agent. If the attempt to change modes is successful, but the agent has any active or held calls, the lamp flashes until all calls are dropped, at which point the lamp lights steadily and the CMS is informed. If the agent tries to enter the Manual-In mode for an invalid split or for a split of which he or she is not a member, or if the agent dials an invalid feature access code, the attempt is canceled and the agent receives intercept treatment.

After Call Work Mode: An agent should enter the After Call Work (ACW) mode when he or she needs to perform ACD-related activities. For example, an agent may need to fill out a form as a result of an ACD call. The agent can enter the ACW mode to fill out the form. The agent is unavailable for ACD calls to all splits while in the ACW mode, although the agent is placed in the MIA queue upon entering this state and advances in queue until entering the AUX state or until he or she becomes available and reaches the front of the MIA queue and receives a call.

When an agent is in the Manual-In mode and disconnects from an ACD call, he or she automatically enters this mode. To change to the ACW mode while in another mode, the agent can dial the feature access code for the ACW mode followed by the split group number, or press the ACW button for that split. (Since BCMS/CMS statistics for ACW provide the average ACW time, entering the ACW node manually may skew reports.) If the attempt to change modes is successful, the lamp associated with the ACW button lights steadily and the BCMS/CMS is informed of the mode change for that agent. If the attempt to change modes is successful, but the agent has any active or held calls, the lamp flashes until all calls are dropped, at which point the BCMS/CMS is informed. If the agent tries to enter the ACW mode for an invalid split or for a split of which he or she is not a member, if the agent is already in the ACW mode for another split, or if the agent dials an invalid feature access code, the attempt is canceled and the agent receives intercept treatment.

Once an agent has entered the ACW mode for a particular split, the agent is no longer available to answer ACD calls to that or any other split. [The agent is automatically placed in the AUX work mode for any other split(s).] However, the agent is still available for non-ACD calls. The BCMS/CMS is notified whenever an agent in the ACW mode receives an incoming non-ACD call or makes an outgoing call.

Agent Request for Supervisor Assistance

Agents can request assistance (whether on an active ACD call or not) from the split supervisor by pressing the ASSIST button or by putting the call on hold and dialing the Assist feature access code, followed by the split group number. The agent must be logged into the split. Assist generates three burst ring at the supervisor’s station. If a split supervisor is not assigned, the agent receives intercept tone.

To request supervisor assistance using the Assist button, the agent does as follows:

- If the agent is active on an ACD call, the agent presses the ASSIST button for that split. This automatically places the ACD call on hold and places a call to the split supervisor. The BCMS/CMS is notified of the request and the supervisor's display (if provided) shows that the call is a request for assistance. After the agent has talked to the supervisor, the agent can drop the assist call and return to the ACD call, or the agent can set up a conference call with the agent, the supervisor, and the calling party. The agent can also transfer the call to the split supervisor, if desired. The calling party hears silence or music on hold (if administered) when the agent depresses the assist button.

  If the agent is an attendant, he or she should first press the START button before pressing the ASSIST button. This will allow the attendant to later transfer the call. This rings like a priority call at the supervisor's set.

- If the agent is not active on a call, the agent goes off-hook and presses the ASSIST button. This automatically places a call to the split supervisor. The BCMS/CMS is notified of the request and the supervisor's display (if provided) shows that the call is a request for assistance. This rings like a priority call at the supervisor's set.

To request supervisor assistance using the Assist feature access code, the agent does as follows:

- If the agent is active on an ACD call, the agent places the call on hold, receives dial tone, and dials the Assist feature access code followed by the split group number. The Assist call is then placed to the split supervisor. The Call Management System is notified of the request for assistance, and the supervisor's display (if provided) shows the call is a request for assistance. After the agent has talked to the supervisor, the agent can drop the Assist call and return to the ACD call, or the agent can set up a conference call with the agent, the supervisor, and the calling party. The agent can also transfer the call, if desired. This rings like a priority call at the supervisor's set. Assist calls will not follow the supervisor's call coverage path.
- If the agent is not active on an ACD call, the agent goes off-hook and then dials the Assist feature access code followed by the split group number. The Assist call is then placed to the split supervisor. The CMS is notified of the request for assistance, and the supervisor's display (if provided) shows that the call is a request for assistance. This is an exception report and must be set up. This rings like a priority call at the supervisor's set. Assist calls will not follow the supervisor's call coverage path.

ACD Call Disconnecting

An agent can be disconnected from an ACD call in either of four ways.

- The agent can press the RELEASE button (if provided). Dial tone is not heard after the RELEASE button is pressed.
- The agent can press the DROP button (if provided). The agent hears dial tone after pressing the DROP button and is not available for calls.

- The call can be dropped by the calling party.
- The agent can go on-hook (hang up).

  Agents using Automatic Answer are logged out of all splits when they disconnect from an ACD call by going on hook (hanging up). The preferred method of operation is to use the RELEASE button (if provided).

Stroke Counts

Stroke Counts provide ACD agents with the ability to record up to nine customer-defined events on a per-call basis when the adjunct CMS is active. The feature also provides a tenth event count to track audio difficulty. Stroke Count "0" is reserved for Audio Difficulty and the other nine Stroke Counts are customer definable.

Stroke counts are reported to the CMS in real time. The switch does not store any Stroke Count information. Therefore, Stroke Counts are only useful when CMS is connected and ACD splits are administered to be measured by CMS

ACD Stroke Counts allow agents to record relevant events on a per-call basis. A Stroke Count represents an event that the customer wants to measure. For example, a Stroke Count may be used to keep track of the number of inquiries about a specific item. Each time an agent receives an inquiry on a specific item, he or she can enter the Stroke Count (one through nine) assigned to that item.

Stroke count "0" is used to indicate audio difficulty. For troubleshooting purposes, CMS will record the equipment location for the trunk the agent was using when the Audio-difficulty button was depressed. This count refers to calls with poor transmission quality experienced by ACD agents. Audio transmission quality is not improved by pressing the Audio Difficulty Stroke Count button, the specific trunk used for the call is recorded

Ten button types maybe assigned for use with Stroke Counts: 0-Stroke, 1-Stroke, 2-Stroke, 3-Stroke, and so on.

An ACD agent enters a Stroke Count by pressing a Stroke Count button while off-hook. The system then validates that the activating agent is either active on an ACD call or in the ACW work mode for an ACD split. If these conditions are met, the feature lamp lights steadily for two seconds to indicate successful activation and a message is sent to the CMS containing the Stroke Count. If the preceding conditions are not met, the feature lamp will flutter to indicate unsuccessful activation and no message will be sent to the CMS. The lamp goes dark after two seconds.

Call Work Codes

Call Work Codes allow ACD agents to enter up to 16 digits for an ACD call to record the occurrence of customer-defined events (such as account codes, social security numbers, or phone numbers). The switch does not store any Call Work Code information. Call Work Codes are sent to an adjunct CMS for storage. Release 3 of the CMS is required to record Call Work Code information.

Data will be sent to the CMS only for the splits that are measured by the CMS and only when the link to the CMS is up. The activating agent must be on an ACD call or in the ACW mode after disconnecting from a call while in the Manual-In mode or in the Auto-In mode and pending for the ACW mode. Activation in any other work mode is denied.

To enter a Call Work Code, while off-hook and active on an ACD split, the agent presses the CALL WORK CODE button.

To enter a Call Work Code after disconnecting from a Manual-In ACD call (Auto-In and pending ACW applies as well, the agent must remain off-hook after disconnecting from the ACD call. To disconnect from the ACD call while remaining off-hook, the agent can either press the RELEASE button or wait for the far-end caller to hang up.

The system then validates that the activating agent is either active on an ACD call or in the ACW mode after disconnecting from an ACD call. If these conditions are met, the associated lamp lights steadily and a C: prompt appears on the display. This indicates that the feature is ready for digit collection. After receiving this visual indication, the agent can enter the desired digits. (The agent must wait for the visual indication before entering the digits, or the calling party will hear the touch-tone digits being dialed). The agent may enter up to 16 digits. Any digits beyond 16 will be ignored by the system and will not appear on the display. The agent then presses **#** to send the Call Work Code entry to the CMS. The Call Work Code lamp will go dark and the display will return to normal mode. If an error is made while entering digits, the user may press ***** to erase all previous digits and begin entering digits again.

If any button is pressed at the agent's voice terminal, or if the agent hangs up during digit collection, the Call Work Code entry is aborted and no message is sent to the CMS. Also, the Call Work Code lamp is extinguished and the display is cleared (only one CWC may be entered per call).

The Call Work Codes function may be used by as many as 40 agents simultaneously. If 40 agents are simultaneously using this function, and another agent attempts to enter a call work code, this agent will receive a display message telling the agent to try again later.

This feature requires display-equipped voice terminals (for example, CallMaster).

Forced Entry of Stroke Counts and Call Work Codes

An agent is always allowed to enter a Stroke Count and/or Call Work Code for an ACD call (as long as the agent is on an ACD call or in the ACW mode after disconnecting from a call while in the manual-in mode). Activation in any other work mode is denied. However, each split can be administered so that agents in that split are forced to complete a Stroke Count and/or a Call Work Code entry for every call answered in the Manual-In mode. For details on Forced Entry of Stroke Counts and Call Work Codes, see "Automatic Call Distribution (ACD)" on page 3-170.

Agent Sizing (G3V3 and later releases)

This feature adds a maximum capacity to the number of logged-in ACD agents. It can be used to limit the number of logged-in ACD agents to a number less than (or equal to) the maximum supported by the system Hunt Group member capacity of the hardware configuration.

The new logged-in ACD agents limit applies to ACD agents in traditional ACD and Expert Agent Selection (EAS) calls. Auto-Available Split (AAS) agent ports are logged in and counted when they are first assigned, while the non-AAS agents are counted when they actually log in. Each logged-in agent is counted as a single agent independent of the number of splits or skills logged in to for the new logged-in ACD agents limit.

In addition to the logged-in ACD agents limit, the number of agents supported is dependent on the upper limits that the hardware platform supports. The following existing limits must also be considered in sizing a call:

- System Hunt Group members — Agent/split pairs assigned or agent/skill pairs logged in
- Hunt Group member per group (referred to as per group member limit herein)
- EAS agent login IDs per system
- Call Management System (CMS) measured agent/split pairs assigned or agent/skill pairs logged in
- Basic Call Management System (BCMS) measured agents assigned in non-EAS and logged-in EAS

When the maximum number of non-EAS ACD agents are logged in, an ACD agent who attempts to log in hears reorder (fast busy) tone. In the EAS environment, if the maximum number of agent members are currently logged in (that is, the system has reached the logged in ACD agent limit) and an agent attempts to log in the logical agent ID, the system denies the login and the agent hears reorder (fast busy) tone.

This maximum capacity is administrable by authorized AT&T personnel via the Agent Sizing option on the “System-Parameters Customer-Options” form. The maximum number of allowed logged in ACD agents are set to correspond to the configuration a customer purchases. These size limits correspond to price element codes (PECs) and permit the customer to purchase agent capacity based on their individual needs.

For Agent Sizing, customers with agents working in shifts should purchase enough agent capacity to allow for a smooth shift change. If agents on a subsequent shift are logging in before agents in the previous shift have logged out, agents could be denied login because too many agents are currently logged in. Additionally, Call Center managers need to be aware of their logged-in ACD agent limit when adding agents to handle a traffic peak or when planning a special campaign.

In G3V3 and later releases, the new logged-in ACD agent limit is added to the existing limits. Any of the following limits may cause a log-in attempt to fail:

- The number of agents currently logged in matches the logged-in ACD agent limit that was purchased.
- The number of agents currently logged in to this split/skill matches the per group member limit for this G3 hardware configuration.
- The number of non-ACD hunt group members plus agent/skill pairs currently logged in matches the system hunt group members limit for this G3 hardware configuration.
- The number of EAS agent/skill pairs currently logged in matches the CMS measured agent/skill pairs limit for this CMS memory configuration.
- The number of EAS agents currently logged in matches the BCMS measured agents limit for this G3 hardware configuration.

The administrator of a non-EAS system can also be blocked from adding agents to splits if the agent/split pairs plus non-ACD hunt group members currently administered match the system hunt group members limit or the split/agent pairs currently administered match the CMS measured agent/split limit or the number of agents administered matches the BCMS measured agents limit.

In addition to purchasing the correct number of logged-in ACD agents, the customer must also purchase the correct hardware configuration and CMS memory configuration to actually use all of the logged-in ACD agents purchased. There is no additional configuration required for BCMS. The G3 hardware platform has always limited the capacity of ACD agents and will continue to do so. The following are G3V2 limits that will carry into G3V3 and later releases.

- The system hunt group member maximum provides an upper limit on the total number of hunt group members. Hunt group members consist of non-ACD members assigned, AAS agent ports assigned (traditional ACD and EAS), traditional ACD agent members assigned and logged-in EAS agent members. Agents in multiple splits/skills count as multiple system hunt group members.

- The BCMS measured agents maximum provides an upper limit on the number of BCMS measured non-EAS agents assigned or the number of BCMS measured EAS agents logged in.
- The CMS measured agents maximum provides an upper limit on the number of CMS measured non-EAS agent/split pairs assigned or the number of CMS measured EAS agent/skill pairs logged in.

The users of ‘Agent Sizing via the Customer Option’ form are provisioning personnel. When a customer orders a G3V3 or later release with traditional ACD or EAS capability, he/she also orders the maximum number of logged in agents. Using the ‘Customer Option’ form, provisioning sets the allowable maximum number of logged-in agents.

NOTE:
The Customer Option form field “Logged-in ACD Agents” sets the maximum number of ACD agents that can log in simultaneously. The switch keeps a separate count of logged in agents. On a log-in attempt, the value for “Logged-in ACD Agents” administered on the form is compared to the count of logged in agents kept by the login/logout software. If the number of logged-in agents is already equal to the number of “Logged-in ACD Agents,” then new login attempts are denied.

The value for “Logged-in ACD Agents” cannot be compared with the allowable number CMS measured agents. The switch has multiple limits with respect to ACD agents. The limit on the CMS should be four times the number of agents purchased and administered on the switch.

Considerations

The "Agent Call Handling" feature is really a combination of features and functions that allow ACD split agents to handle ACD calls quickly and efficiently.

An agent, although he or she can be assigned to one or more splits, can be logged only into four splits at a time.

The number of digits in the log-in identification number must equal the number assigned through system administration (0 to 9). The agent’s individual identification number is used for record keeping purposes on CMS or BCMS. To track individual agent data, the login ID must be at least one digit. The system checks the number of digits in the identification number and verifies that it is not already active. It does not check to see if the identification number is a valid number although CMS and BCMS use the login identification for reports.

For each split to which an agent is assigned, he or she can be assigned a maximum of one of each of the following feature function buttons:

- Manual-In
- Auto-In

- Auxiliary Work
- After Call Work

A terminal or console can be assigned a maximum of one ACD RELEASE button. This button is in addition to the fixed RELEASE button on the attendant console.

For non-vector-controlled splits, the last available agent in a split cannot enter the Auxiliary Work mode if any calls are remaining in the split queue. An attempt by the last available group member to enter the Auxiliary Work mode results in the following:

- The Auxiliary Work button flashes.
- New calls on the ACD split either receive the busy tone or redirect to coverage. Calls in the split queue continue to route to the last available agent until the split queue is empty.
- At the last available voice terminal or console, the status lamp associated with the Auxiliary Work button, if provided, flashes until the split queue is empty. When no more calls remain in the split queue, the Auxiliary Work mode is entered and the associated status lamp, if provided, lights steadily. (The same sequence applies when the Auxiliary Work mode is dial activated instead of button activated, except there is no status lamp.)

NOTE:
The agent can, however, log out.

If an agent is logged into more than one split, the agent may become unavailable for calls to one split, because of activity at another split. For example, an agent may enter the After Call Work mode for one split. This makes the agent unavailable for calls to other splits the agent is logged into.

An ACD agent on conference with more than three parties may cause inaccurate CMS measurements.

An agent should not log into a split while a call is on hold at his or her extension.

On direct calls to ACD agents with Auto-Answer, incoming notification (zip tone) will be sent instead of ringing the station once. This includes attendants. If the agent is active on a call, the direct call will provide a single ring.

Any calls to non-ACD Auto-Answer agents will be announced by Incoming Call ID tone. Ringing not longer be heard.

Calls for CALLMASTER digital voice terminals and attendant stations will be announced by double tones. The user hears part of the first tone and all of the second tone. The tones that are doubled are Zip (Auto-Answer ACD agent calls) and Incoming Call ID (all other Auto-Answer calls).

Agents should not be used for hunt group calls and ACD split calls simultaneously. Otherwise, all of the calls from one split (either ACD or hunt group) will be answered first. For example, if the ACD calls are answered first, none of the hunt group calls will be answered until all of the ACD calls are answered.

The oldest call waiting termination is only supported for agents who are servicing ACD calls only.

Interactions

The following features interact with the "Agent Call Handling" feature.

- Abbreviated Dialing

  An agent may have Abbreviated Dialing buttons assigned to make the log-in process easier. An Abbreviated Dialing button can be programmed to dial the access code, split number, and/or identification number.

- Auto-Avail-Split/Skill (AAS)

  AAS split or skill members are logged in and counted as members towards the system hunt group member, the per group member, and the logged-in ACD agents limits during administration. Reaching any one of the limits can prevent some of the skills or the Login ID from being logged in (see Req 2). If the port for an AAS split or skill member goes out of service (that is, RONA), the count of agents logged in is not updated. These counts for these members are updated only during administration.

- Bridging

  ACD split/skill calls are not bridged. When an agent handles an ACD split/skill call, bridged appearances of that agent will not be bridged to the call.

  Station calls are bridged and agents are able to bridge onto them. If an agent bridges onto a call, the call is considered a non-ACD extension-in call. The agent will not be available for an ACD call at this time unless the agent is a member of a "many-forced," "one-forced," or "one-per-skill" MCH split/skill. The agent can hold this call and become available to receive ACD calls even in non-MCH splits/skills if only bridged appearances are active.

- Call Pickup

  A MCH agent can pick up a call with calls on hold. When a call is answered by an ACD agent via the "Call Pickup" feature, the call will be treated as an incoming non-ACD call. The agent can also put a call pickup call on hold and become available for additional calls.

- Expert Agent Selection (EAS)

  When EAS is active, all ACD hunt groups are assigned as skills and all skills are vector controlled. Agents must log in using Logical Agent IDs which are preassigned on the ‘Login ID’ form. Skills can be preassigned to login IDs, however, preassignment on the ‘Login ID’ form does not actually assign a non-AAS login ID to the skills (that is, not counted as a hunt group member or logged-in ACD member) until the ID is logged in. When the login ID is logged in, each skill is counted as a hunt group member towards the system hunt group member limit, the per group member limit, and each agent is counted as a logged-in ACD agent.

- Hunt Groups (Non-ACD Hunt Groups)

  While non-ACD hunt group members are not counted as part of the logged-in ACD agents limit, they can affect the actual number of traditional ACD or EAS agents that may be logged in if the system hunt group members maximum is reached.

  The number of non-EAS system hunt group members is non-ACD hunt group members assigned plus agent/split pairs assigned.

  The number of EAS system hunt group members is non-ACD hunt group members assigned plus AAS agent/skill pairs assigned plus non-AAS agent/skill pairs logged in.

- Hunt Groups (Unmeasured ACD Hunt Groups)

  Unmeasured ACD hunt group members are counted as part of the logged-in ACD agents limit and the system hunt group members limit.

- Port Sizing

  Port Sizing is a customer option that allows AT&T to make port capacity price sensitive on a per customer basis. All ports used on a system are included in this customer optioned size. While each logged-in agent uses one port, additional ports are needed for trunks, announcements, etc. For this reason, there is no error checking done between the “Port Sizing Option” field and the “Logged-in ACD Agents” field.

Administration

Agent Call Handling is administered by the System Manager. The following items require administration on a per-terminal or per-console basis:

- Whether it has Automatic Answer or Manual Answer
- Whether or not it has Idle Appearance Preference (for placing calls)

The following items are optional:

- Manual-In button
- Auto-In button
- Auxiliary Work button
- After Call Work button
- Assist button
- Release button (required with CallMaster voice terminal)

- Stroke Count buttons
- Call Work Code buttons
- VuStats display and buttons

In addition to the above, the following items require administration on a per-system basis:

- Feature access codes:
  - Agent Log-In
  - Agent Log-Out
  - Manual-In
  - Auto-In
  - After Call Work
  - Auxiliary Work
  - Assist
- Number of digits in log-in identification

Each split must be assigned Forced Entry of Stroke Counts and Call Work Codes if agents in that split are to be required to enter these items.

'System-Parameters Customer-Options' form:

- The "Logged-In ACD Agents" field must be administered for the appropriate maximum number of agents allowed to log in.

Hardware and Software Requirements

No additional hardware is required, although CallMaster voice terminals are recommended for ACD agents. ACD software is required.

Alphanumeric Dialing

Feature Availability

Alphanumeric Dialing is available with all Generic 3 releases.

Description

Alphanumeric Dialing enhances Data Terminal Dialing by allowing data terminal users to place a data call by entering an alphanumeric name. This capability makes Data Terminal Dialing both convenient and user-friendly. Instead of dialing a long string of numbers, the users can enter a simple alphanumeric name.

When an alphanumeric name is entered from a user's terminal, the system's call processing software converts the name to a sequence of digits by searching through an administered Alphanumeric Dialing Table. The system then dials those digits just as if the user had entered the digits. If the entered name is not found in the Alphanumeric Dialing Table, the call attempt is denied and the user receives either an `Invalid Address` message (for DCP) or a `Wrong Address` message (for ISDN-BRI).

Since data terminals access the switch via DCP or ISDN-BRI data modules, the procedures for using Alphanumeric Dialing vary. For data terminals using DCP, users type the alphanumeric name and enter a carriage return at the `DIAL:` prompt. For data terminals using ISDN-BRI, users type ***d***, enter a space, type the alphanumeric string, and enter a carriage return at the `CMD:` prompt.

Alphanumeric dialing does not apply to endpoints with Hayes interface.

More than one alphanumeric name can refer to the same digit string. Also, multiple names (mixed with number strings) can be used to dial a number. For example, a company may administer the Alphanumeric Dialing Table to convert the alphanumeric name ***home*** to the digit string for the area code and office code of the home office. In this example, a data terminal user with a DCP data module could access extension 3797 at the home office by typing ***home 3797*** and entering a carriage return at the `DIAL:` prompt. A data terminal user with an ISDN-BRI module could access extension 3797 at the home office by typing ***d home 3797*** and entering a carriage return at the `CMD:` prompt.

Considerations

Alphanumeric Dialing allows a data terminal user to place a data call by entering an alphanumeric name. This makes Data Terminal Dialing both convenient and user-friendly. Instead of dialing a long string of numbers, the user can enter a simple alphanumeric name.

Interactions

The following feature interacts with the "Alphanumeric Dialing" feature.

- Data Call Setup

  Alphanumeric Dialing enhances Data Terminal Dialing by allowing a data terminal user to place a data call by entering an alphanumeric name.

Administration

Alphanumeric Dialing is administered by the System Manager. In addition to those items listed in the "Data Call Setup" feature description, elsewhere in this chapter, alphanumeric names and associated digit strings must be assigned in the Alphanumeric Dialing Table.

Hardware and Software Requirements

No additional hardware or software is required.

Alternate Facility Restriction Levels (AFRL)

Feature Availability

This feature is available with Generic 3rV1 and all Generic V2 and later releases.

Description

This feature allows the DEFINITY Communications System to adjust facility restriction levels for devices (lines or trunks) as well as authorization codes within a given system.

Alternate Facility Restriction Levels provides an alternate mode of operation than that prescribed by the original set of facility restriction levels. Facility Restriction Levels (FRLs) are used to determine the privileges an originating party can have when making an outgoing trunk call. This originator can be either a line or a trunk. An FRL is assigned to all devices and/or device groups through its associate Class Of Restriction (COR). When a device originates an outgoing trunk call its FRL is compared to the FRL of the preference route in the case of AAR/ARS routing. If the FRL of the originator is greater than or equal to the terminated to FRL, the call can proceed. If the FRL of the originator is less than that of the terminated to trunk, the call is blocked (unless an authorization code is used to override the blockage, this is explained later). For example, if the FRL assigned to a particular trunk group is three (where seven is least accessible and zero most accessible from the perspective of someone wishing to access the trunk group) only originators with an FRL of three or greater are allowed to use this trunk. FRL’s are administered within the system to allow or restrict outgoing calls according to their particular destination, tariffs applied on certain calls at certain times of day, or to facility problems (trunk outages for example).

The "Alternate Facility Restriction Levels (AFRL)" feature allows a change of the administered FRLs for all originating devices and authorization codes to a new value. With the "Alternate Facility Restriction Levels (AFRL)" feature, a change of an FRL level can be established by the technician or the customer. For example, through AFRL all FRLs less than three are assigned a value of three and FRLs greater than three can be assigned a value of seven. Whether the original FRL value or the AFRL value is used, it is controlled through any attendant or station with an AFRL feature button. By depressing the button, the user can control whether calls in the switch which require FRL’s for determining access are using FRL’s or AFRL’s.

AFRL affects two types of users. The attendant and a voice terminal user. Therefore, two perspectives are addressed. These are:

- The perspective of the physical terminal user, for example, the person who wishes to make an outgoing trunk call.

- The perspective of the attendant who must enable or disable the AFRL feature for the system.

Terminal User Perspective

In general, persons who desire to make outgoing trunk calls should see no changes to their mode of operation. A minor change is the ability or inability to make certain calls based upon the administration of the switch at a given time. This means that a user may suddenly not be able to make a call that at another time of day was able to be completed by the system. Although the user does not know why the call was disallowed, policies regarding the application of this feature and its effect on the users should be made known to all users by the technician prior to the AFRL feature being activated and used.

Attendant Perspective

An ALT-FRL feature button can be administered to any attendant consoles and one station per entire system. An ALT-FRL button on the attendant console or the station activates or deactivates the "Alternate Facility Restriction Levels (AFRL)" feature. This button simply toggles whether the feature is activated or not. The depression of the feature button on any console always affects the status of the button on all other consoles and station administered with this feature button.

When the feature is active, all AFRL feature buttons on equipped consoles/station are lit. When the feature is deactivated, all feature buttons on equipped consoles and stations are extinguished. Also, as the feature is activated and deactivated as the ALT-FRL button is depressed, the status of the AFRL for the 'System Management' form is changed so that the technician can monitor the state of the machine at any given time.

When AFRL is activated the user perceives a sudden increase or decrease in calling privileges. For example the user may have been able to make long distance calls but due to the activation of AFRL's (which have been administered to add restrictions to the calling parties) the call is no longer allowed to be completed. In this mode the user can be anyone of three types of alterable FRL entities. These include:

- An originating line
- An originating trunk (for example, an incoming trunk call)
- A dialed authorization code

Line Originator Case

A line in this case is any device (other than a trunk), which wishes to complete an outgoing trunk call. This includes Analog Stations, Multi-Function Stations, Digital Stations, Data Terminals, Attendants, etc. Each of these facilities is administered to have a class of restriction, which in turn is assigned a Facility Restriction Level. This FRL is used to determine if the originating party has access to this outgoing trunk. When the AFRL feature is activated, the FRL for the device shall be the AFRL values administered by the technician. The user has no control over the activation or deactivation of the feature, nor the engineering of the AFRL values.

Trunk Originator Case

In this case, the originator is in reality an incoming trunk. Incoming trunk calls can be DID calls to a station, incoming calls on access tie trunks, or the possibility of intertandem tie trunk calls. In either case the FRL of the device is first used to determine if it is at a high enough level to select the outgoing device. If it is not, then the FRL associated with this signaling entity, called a Traveling Class Mark (TCM), is used. By default, every device has associated with it an FRL (indirectly through the application of its assigned COR). However, tandem trunk groups are administered to pass along the FRL level associated with the other switch as part of the interoffice signaling protocol. This can be done inband as part of the addressing information which is sent between two switches using tone sending on the trunk itself, or can be done using special ISDN Q.931 messaging in the case of an ISDN facility. In either case, the seizure of the outgoing trunk is not done until either the TCM is received or (in the case of inband signaling) an interdigital timeout occurs while waiting for the TCM digit.

If AFRLs are active, the FRL associated with the incoming trunk group is set to a new FRL. If after doing so, the originator is blocked from the trunk facility due to insufficient FRL, the TCM, if any, is used. The TCM, which is in reality just another FRL, is also set to a new FRL value. Therefore, the TCM information recorded in the billing data (CDR) is the AFRL value, not the original TCM.

Due to the application of AFRL on tandem and tie trunk applications, entire networks can be affected by the application of the "Alternate Facility Restriction Levels (AFRL)" feature. This means that a judicious use of this feature must be applied when engineering AFRLs as *now calls* which may be part of a cross-country PBX network and may be blocked from completion due to the engineering of a restrictive AFRL arrangement.

Authorization Code Case

The "Authorization Codes" feature is used to prevent unauthorized access to various facilities on the system. It can be used to restrict access to certain trunk groups or to Remote Access trunk groups as examples. When a user dials an Authorization Code, it is validated by the system against all the technician administered Authorization Codes. If invalid, the call is routed to an administrable intercept. If the code is valid, an associated Class Of Restriction (COR) is determined. This COR has an FRL associated with it that can be changed with the "Alternate Facility Restriction Levels (AFRL)" feature.

Considerations

You need to consider the impact on your operations when there are sudden changes in the calling privileges of your users. You should consider announcing the change and preparing your telecommunications department to handle user inquiries.

Interactions

AFRL interacts with the "Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs)" feature by setting up alternate levels to those normally provided by FRL. See the "Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs)" feature for more information on interactions. AAR/ARS call routing is affected because ARFL could change route preferences. AFRL could have an impact on the cost of usage-sensitive calls.

Administration

The administration of the "Alternate Facility Restriction Levels (AFRL)" feature is done through the use of the **CHANGE Alternate FRL** command. The commands which affect the AFRL are the **CHANGE** and the **DISPLAY** commands. There is no **ADD** or **DELETE** command associated with an AFRL since the default values are set to the same value as the FRL.

Field Description

There are only two field types which can be changed using the “CHANGE Alternate FRL” screen. These are:

- For AFRL

  The values entered must be within the current range allowed for FRL entries (0 through 7) found in the “ADD/CHANGE COR” forms for example.

- For Status

  The value entered is either active or inactive.

Error Messages

There are only two types of error messages returned for the “CHANGE Alternate FRL” screen.

- For AFRL

  Error messages returned are identical to those returned for FRL entries found in the “ADD/CHANGE COR” forms.

  `Entry out of range` - used when value not 0 through 7.

  `Field cannot be blank` - used when value entered is blank.

- For Status

  Error message states that value entered is not active or inactive.

  Status must be active or inactive - when invalid status value entered.

Help Messages

There are only two types of help messages returned for the CHANGE AFRL screen.

- For AFRL

  `Enter a number between 0 through 7.`

- For Status

  `Either active or inactive.`

Initial Values

There are only two field types needing initialization. These are:

- For AFRL

  The initial (default) values for the AFRL associated with an FRL are the same value as the FRL. This means the FRL 0 has an AFRL of 0, FRL 1 a AFRL of 1 etc.

- For Status

  Will be defaulted to inactive which means either the technician or an attendant with an assigned "Alternate Facility Restriction Levels (AFRL)" feature button must activate the feature.

Administration of AFRL Feature Button

A new option is allowed when engineering the feature buttons on a station. The new feature button type of ALT-FRL can be used to allow one station the ability to activate or deactivate the changing of originating FRL to AFRL values. The extension number of this station is administered on the ‘Console Parameter’ form.

Attendant Class of Operations

An attendant or a station with console permissions, can be set up to have an "Alternate Facility Restriction Levels (AFRL)" feature button on it. By depressing this button and activating the feature (also can be activated by technician from the SAT or SAT PC) the following takes place:

- Calls requiring FRL authorization checks have the FRL set to a new AFRL value.
- Calls using authorization codes have the FRL associated with the authorization code mapped to a new AFRL value.

- The status light associated with all the "Alternate Facility Restriction Levels (AFRL)" feature buttons assigned in the system is lit.
- The status of the "Alternate Facility Restriction Levels (AFRL)" feature is set to active and can be displayed using either the **CHANGE AFRL** or **DISPLA AFRL** commands from the administration terminal.

By depressing the button again, the feature is deactivated (also can be activated by technician from the SAT or SAT PC as) and the following occurs:

- Calls requiring FRL authorization checks now use the FRL value.
- Calls using authorization codes now use the FRL associated with the authorization code.
- The status light associated with all the "Alternate Facility Restriction Levels (AFRL)" feature buttons assigned in the system is turned off.

The status of the "Alternate Facility Restriction Levels (AFRL)" feature is set to inactive and can be displayed using either the **CHANGE AFRL** or **DISPLA AFRL** commands from the administration terminal.

The administration of the "Alternate Facility Restriction Levels (AFRL)" feature is done through the use of the **CHANGE AFRL** command. The commands which affect the AFRL are the **CHANGE** and the **DISPLAY** commands. There is no **ADD** or **DELETE** command associated with an AFRL since the default values are set to the same value as the FRL.

Hardware/Software Requirements

No additional hardware is required for this feature.

Answer Detection

Feature Availability

Answer Detection is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Improves the accuracy of the call duration in CDR call detail records by detecting the state of outgoing trunk calls that do not receive Network Answer Supervision.

A timer is used to determine when the called party has answered. Since Network Answer Supervision may or may not be sent back, a normal outgoing trunk call without the "Answer Detection" feature relies upon the Far End Answer Supervision timer once an outgoing call is placed.

The Far End Answer Supervision timer is an internal, administrable timer that estimates when the far end should have answered the call. The interval for the Far End Answer Supervision timer is specified on the 'Trunk Group' form. As soon as the specified Far End Answer Supervision time interval is reached, CDR generates a call detail record. However, if the call terminates before the timeout interval is reached, CDR does not generate a call detail record.

Answer Detection allows the system administrator to set the answer supervision timeout to larger values while generating CDR call detail reports on short duration calls.

The "Answer Detection" feature uses a port on the Call Classifier circuit pack to detect various tones and voice-frequency signals received from the outgoing trunk, and classifies the call as answered, unanswered, busy, etc. based on this tone detection. A call that has been classified as answered by this tone detection capability is assumed to have been answered for the purpose of generating a CDR call record, even if Network Answer Supervision has not been received and/or the Far End Answer Supervision timer has not been timed out.

This provides a more accurate time of call duration and therefore improves Call Detail Recording and any subsequent billing based on CDR records.

NOTE:
A CDR call record is generated if a call classifier is involved and the call is classified prior to the Far End Answer Supervision timer expiring.

Considerations

The "Answer Detection" feature does not accurately detect all types of tones especially in countries whose tone schemes are not similar to the US. For a normal answered call, the call will usually be correctly classified as answer. However, some calls may be misclassified as Fast Busy when they are actually answer. Miscellaneous tones, such as the PBX tones (that is, confirmation) will be classified as answer. In addition, loud background noise may activate Answer Detection, causing the call to be classified as answer, even if the call is not connected. If Answer Detection is accidentally activated, but the call is not connected, the call is recorded in CDR records on the PBX. If Answer Detection is accidentally activated and the call is connected, the caller is billed erroneously for the time that elapses between Answer Detection activating and the called party answering.

Administration

While Answer Supervision is a system-wide option, the timer interval that applies to a particular trunk must be administered on a trunk group basis. Only trunks administered as CO, FX, or WATS trunks receive this treatment.

Interactions

The following features interact with the "Answer Detection" feature.

- CallVisor ASAI

  Answer Detection competes with CallVisor ASAI switch-classified calls for ports on the Call Classifier circuit pack. Answer Detection triggers reporting of a Connect Event to ASAI.

- Call Prompting

  Answer Detection competes with Call Prompting for ports on the Call Classifier circuit pack.

- CDR

  Answer Detection provides more accurate CDR records where tone detection is possible and Network Answer Supervision is not received. Answer Detection triggers reporting of a Connect Event to ASAI.

Hardware and Software Requirements

Requires a TN744 Call Classifier circuit pack.

Attendant Auto-Manual Splitting

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the attendant to announce a call or consult privately with the called party without being heard by the other party on the call.

This feature is activated automatically when the attendant, active on a call, presses the START button, a Hundreds Select button and a Direct Extension Selection button (if provided), or a Trunk Group Select button. Any of these actions temporarily separates the party on the call from the connection and allows the attendant to call and talk privately with another party.

The connection is reestablished when the attendant presses one of the following buttons:

- **Cancel** — Cancels the call attempt and reconnects the attendant and the separated party.
- **Split** — Establishes a three-way conversation with the attendant, the separated party, and the called party.
- **Release** — Connects the separated party and the called party and disconnects the attendant.

Considerations

Attendant Auto-Manual Splitting provides for splitting the calling party away so attendant can confidentially determine if the called party can accept the call.

Administration

To use Auto-Manual Splitting, one SPLIT button must be assigned per console and "Auto Start'' must be disabled on "System Parameter'' form for G3i-Global, G3rV1, G3V2, G3V3, G3V4, and later releases.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Call Waiting

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows an attendant originated or extended calls to a busy single-line voice terminal to wait at the called terminal. The attendant is free to handle other calls.

Attendant Call Waiting is activated for a single-line station whenever an attendant originates or extends a call to a busy single-line voice terminal. The attendant hears a Call Waiting ringback tone and the busy voice terminal user hears a call waiting tone. This burst tone is heard only by the called voice terminal user. The number of bursts (one, two, or three bursts) is administrable in G3i-Global, G3V2; and later releases.

When the Attendant Call Waiting is activated the attendant may choose to cancel the call, release the call, or hold the call on the console. However, releasing an attendant-originated call results in the call being dropped completely. If the attendant activates the "Attendant Call Waiting" feature, and the administered Return Call Time-Out or Timed Reminder on Hold interval expires without the call being answered, the call returns to the console.

The call in progress at the voice terminal can be placed on hold. In order to answer the waiting call, after receiving recall dial tone, the user then dials the answer call waiting access code. After answering the waiting call, the voice terminal user can use the "Hold" feature to return to the held call or toggle back and forth between the two calls.

As an example of how Attendant Call Waiting is used, assume extension 123, a single-line voice terminal, is busy. An attendant extends a call to extension 123 and hears the Call Waiting Ringback Tone which indicates that Attendant Call Waiting is activated. The attendant may choose to announce the call waiting condition to the calling party. However, after doing this, the attendant cannot cancel the call. The attendant could cancel the call and ask the calling party to call again later, or the attendant could release the call or place the call on hold at the console. This allows the attendant to handle other calls. The voice terminal user at extension 123 hears a two-burst tone and knows a call is waiting. The voice terminal user at extension 123 can then terminate the call in progress, or place the call in progress on hold, and answer the waiting call. If the waiting call is not answered before a preassigned time interval (Return Call Timeout or Timed Reminder on Hold) expires, the call returns to the attendant.

Considerations

Attendant Call Waiting allows an attendant to originate or extend calls to a busy single-line voice terminal while allowing the attendant to handle other calls. Since the attendant is able to handle other calls while a call is waiting, more calls can be answered.

Attendant Call Waiting applies only for calls to single-line voice terminals within the system. Only one call per voice terminal can wait at a time.

Interactions

The following features interact with the "Attendant Call Waiting" feature.

- Automatic Callback

  If Automatic Callback is activated at the called voice terminal, Attendant Call Waiting is denied.

- Call Coverage

  Attendant Call Waiting calls can be redirected to coverage if the called voice terminal has Data Privacy or Data Restriction activated. If one of these conditions exist, and

  - Call Coverage is assigned to a voice terminal, and
  - Send All Calls is activated or coverage criteria are met,

  the call does not wait and can be redirected to the coverage path. In some cases, the call can wait and then be redirected to coverage. In other cases the call returns to the console, rather than be redirected to coverage. The operation is as follows:

  — The Coverage Don't Answer interval (two to nine ringing cycles or the equivalent time) specifies how long a call remains directed to the called voice terminal before redirecting to coverage. This interval applies to both the Busy and Don't Answer criteria. If Attendant Call Waiting is applicable on the call, this feature is active for the duration of the Don't Answer interval only. At the expiration of this interval, the call redirects to coverage.

  — If the Return Call Timeout (Timed Reminder) interval expires before the Don't Answer interval expires, the call does not go to coverage but returns to an attendant console. If the Don't Answer interval expires first, the call redirects to coverage, but can still return to the attendant console if a coverage point does not answer the call before the Return Call Time-out.

  — If Send All Calls is active or if the redirection criterion is Cover All Calls, the call immediately redirects to coverage instead of waiting.

  — An attendant can release from an extended call at any point during the call, without affecting the preceding operations.

- Data Privacy

  If Data Privacy is activated at the called voice terminal, Attendant Call Waiting is denied.

- Data Restriction

  If Data Restriction is activated at the called voice terminal, Attendant Call Waiting is denied.

- DDC and UCD

  Calls to a DDC or UCD group do not wait; however, such calls can enter the group queue, if provided.

- Loudspeaker Paging Access

  If Loudspeaker Paging Access is activated at the called voice terminal, Attendant Call Waiting is denied.

- Music-on-Hold Access

  Music-on-Hold can be heard by the calling party, if the call is a trunk transferred call and this type of call is administered to receive Music-on-Hold for call waiting calls. Otherwise, the calling party does not hear Music-on-Hold, but hears ringing.

- Recorded Telephone Dictation Access

  If Recorded Telephone Dictation Access is activated at the called voice terminal, Attendant Call Waiting is denied.

- Timed Reminder

  The Timed Reminder interval determines how long a call waits before returning to an attendant console. If the call is not answered or does not redirect to coverage before this interval expires, the call returns to the attendant console.

Administration

"Attendant Call Waiting" is a standard system feature. Attendant Call Waiting is assigned to single-line voice terminals on a per-terminal basis. The call waiting interval is administered through the "Timed Reminder and Attendant Timers" feature by the System Manager. Also, transferred trunk calls that are waiting to be answered can be administered to receive Music-on-Hold.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Control of Trunk Group Access

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the attendant to control trunk groups, and prevents voice terminal users from directly accessing a controlled trunk group.

Each attendant console has 12 designated Trunk Hundreds Select buttons to be used with the "Attendant Direct Trunk Group Selection" feature. Each console may have up to 12 of its feature buttons administered as additional Trunk Hundreds Select buttons, for a total of 24 Trunk Hundreds Select buttons per console. The attendant gains direct access to an outgoing trunk group by merely pressing the button assigned to that trunk group.

All Trunk Hundreds Select buttons (including any administered on the console's feature buttons) have a Busy lamp which lights when all trunks in the associated trunk group are busy. If one of the two-lamp feature buttons on a basic console is administered as a Trunk Hundreds Select button, the bottom lamp is used as the Busy lamp (the top lamp is not used). Six of the designated buttons (basic console) or all 12 designated buttons (enhanced console) have two additional lamps that are used for Attendant Control of Trunk Group Access. The two additional lamps are as follows:

- Warn (warning) lamp

  Lights when a preset number of trunks are busy in the associated trunk group (the busy threshold of the trunk group is reached).

- Cont (control) lamp

  Lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group.

The attendant activates Attendant Control of Trunk Group Access by pressing a CONT ACT (Control Activate) button followed by the desired Trunk Hundreds Select button. The Trunk Group Select button used must have a Cont. (control) lamp. If a user attempts to access a controlled trunk group directly, the call automatically redirects to the attendant. If the attendant decides to allow the call to go through, the attendant can connect the user to the desired trunk group by pressing the associated Trunk Group Select button. The attendant can then release the call or hold the call on the console.

Calls already in queue for a trunk are not affected by the activation of Attendant Control of Trunk Group Access for that trunk group. For example, if an attendant activates Attendant Control of Trunk Group Access for a specific trunk group while a user is waiting in queue for an outside trunk in that trunk group, the call is not affected. The call remains in queue until an idle trunk becomes available, at which time the call is connected to that idle trunk.

The attendant deactivates Attendant Control of Trunk Group Access by pressing the CONT DEACT (Control Deactivate) button followed by the desired Trunk Hundreds Select button. [The Trunk Group Select button used must have a Cont. (control) lamp.] Attendant Control of Trunk Group Access is activated and deactivated separately for each trunk group.

After an attendant presses a CONT ACT or CONT DEACT button, the attendant can perform other operations before pressing the desired Trunk Hundreds Select button. This has no effect on the activation or deactivation of the feature. For example, if the attendant presses the CONT ACT button and then has to answer another call, the desired Trunk Hundreds Select button can be pressed after answering the call. Attendant Control of Trunk Group Access is then activated for the associated trunk group.

Considerations

By activating Attendant Control of Trunk Group Access, the attendant obtains control of access to specific trunk groups. This allows the attendant to monitor the use of these trunk groups. By watching the lamps associated with the trunk groups, the attendant can determine if the number of busy trunks in a specific trunk group has reached a preset warning level and if all trunks in a specific trunk group are busy. The attendant can then handle other calls to these trunk groups accordingly.

This feature can be activated for any trunk group assigned to a Trunk Group Select button with an associated control lamp. Each attendant in the system can control access to six (basic console) or 12 (enhanced console) different trunk groups.

If Attendant Control of Trunk Group Access is activated, and no attendant is assigned, or the attendant is later removed, calls to a controlled trunk group route to the attendant queue.

Interactions

The following features interact with the "Attendant Control of Trunk Group Access" feature.

- Attendant Direct Trunk Group Selection

  This feature must be assigned with "Attendant Control of Trunk Group Access" feature.

- Attendant Display

  When a call redirects to the console because Attendant Control of Trunk Group Access is activated, the alphanumeric display identifies the calling party and shows that the call has attempted to access a controlled trunk group.

- Automatic Route Selection and Automatic Alternate Routing (ARS/AAR)

  Activating Attendant Control of Trunk Group Access removes the controlled trunk group(s) from the Automatic Route Selection and Automatic Alternate Routing patterns. Deactivating the feature reinserts the group(s) into the patterns. Automatic Route Selection calls are not routed to the attendant.

- Trunk Group Busy/Warning Indicators to Attendant

  This feature keeps the attendant informed of trunk group status. This status can be used to determine when to activate control.

- Uniform Dial Plan (UDP)

  Activating Attendant Control of Trunk Group Access removes the controlled trunk group(s) from preferences. Deactivating the feature enables the (UDP) to access the trunk groups.

- Authorization Codes

  When a trunk group has an incoming destination set to the Attendant, authorization codes are *not* collected.

Administration

Attendant Control of Trunk Group Access is assigned on a per-attendant console basis by the System Manager. The following items require administration:

- Attendant Console
  - Trunk groups which are to be controlled
  - The Control Activate button is ACT-TR-GRP and the Control Deactivate button is DEACT-TR-GRP.
- Controlled Trunk Groups
  - Busy Threshold

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Direct Extension Selection With Busy Lamp Field

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the attendant to track extension status (idle or busy) and to place or extend calls to extension numbers assigned to the system without having to dial the extension number. The attendant can use this feature in two ways:

- Using standard Direct Extension Selection (DXS) access. If you use an attendant console that has one or more Hundreds Select buttons, you can press a Hundreds Select button and a DXS button to access a station.
- Using enhanced DXS access. The functionality is useful when:
  - You use an attendant console that does not use Hundreds Select buttons.
  - You use an attendant console with Hundreds Select buttons, but you have one or more hundreds groups not administered by a Hundreds Select button. In this case, you can use a Group Select button and dial the first two or three digits of the station and then use the DXS button to access the station. Please see the "Enhanced DXS Tracking" subsection below.

In both cases (using a Hundreds Select button or the Group Select button), when the system is tracking a group of extensions, the attendant can place or extend subsequent calls to extensions in that group simply by pressing the DXS button, without having to reselect the group. Both of these capabilities eliminate the need to dial extensions.

The extension numbers may be voice terminal extensions, hunt group extensions, off-switch extensions (such as UDP extensions), or nonvoice terminal extensions.

Whichever method you use to access and track DXS extensions, you can view the group of extensions currently being tracked by using a Group Display feature button. This button, on the console display, indicates the range of extensions being tracked by the selector console.

Standard DXS Tracking with Hundreds Select Buttons

Eight Hundreds Select buttons and 100 DXS buttons are located on the basic selector console. The enhanced selector console has 20 Hundreds Select buttons and 100 DXS buttons. Twelve additional Hundreds Select buttons can be assigned to feature buttons on the attendant console. However, if these feature buttons are used, the total number of Hundreds Select buttons per attendant (including both the attendant console feature buttons and the selector console buttons) cannot exceed 20. Each Hundreds Select button is labeled with a different hundreds group number used in the system. For example, if a system uses four-digit extension numbers, the Hundreds Select buttons could be labeled 2400, 2500, 2800, and so on. Likewise, a three-digit system could have these buttons labeled as 100, 200, 300, and so on. A two-digit system would have a 0 Group Select number (G3iV1 and G3vsV1/G3sV1 only). (For G3i-Global and G3rV1, 2 digit extensions can be dialed from a DXS, but the BLF lamps does not track them. This is done by selecting an unadministered hundreds group button on the DXS and then pressing the DXS extension button with the 2 digits of the extension you wish to reach. G3V2 and later releases support tracking of 2-digit extensions via the BLF as well as dialing of 2-digit extensions via the BLF.) A five-digit system, for example, could have group select buttons labeled `28400`, `28500`, `28600`, and so on.

The 100 DXS buttons are labeled `00` to `99`, and each button represents the last two digits of an extension number. Each DXS button, when combined with a Hundreds Select button, represents a unique extension number. To place a call to an extension number, the attendant merely presses the appropriate Hundreds Select button followed by the appropriate DXS button. For example, to call extension 4321, the attendant would press Hundreds Select button `4300` followed by DXS button `21`.

A lamp associated with each Hundreds Select button indicates the selected hundreds group. A selected hundreds group remains selected until another Hundreds Select button is pressed. The associated lamp lights and remains lighted until another Hundreds Select button is pressed. Each DXS button also has an adjacent lamp, which is used to determine the idle/busy active status of the facility associated with the button. When a facility is busy/active, the lamp at the associated DXS button is lighted. When the associated facility is idle the lamp is dark. For multifunction voice terminals with a speakerphone or the CallMaster voice terminal, the facility is busy when a user is active on a call appearance (talking or originating a call) and idle when all call appearances are idle, ringing, or held. For other voice terminals, the facility is busy when the station is off-hook and idle when the station is on-hook.

The 100 lamps adjacent to the DXS buttons are referred to as a “Busy Lamp” field. Although the Group Select and DXS buttons may be used to dial any extension, the “Busy Lamp” field only reflects the status of on-switch resources.

After the Hundreds Select button is pressed, if the lamp adjacent to the desired DXS button is lighted to indicate busy status, the call can sometimes still be placed or extended. Attendant Call Waiting is activated for a single-line voice terminal. A multiappearance voice terminal user receives the call on an idle appearance. If no idle appearances are available, the call can route to coverage, if available, or receive busy tone.

Enhanced DXS Tracking

Enhanced DXS Tracking can help you if you have more than 100 stations, but you use a console that does not have Hundreds Select buttons administered, or if you have more stations than you do Hundreds Select buttons (and thus have hundreds groups that are administered with Hundreds Select buttons). This means that if you have the basic selector console but you have more than 800 stations, or if you have the enhanced selector console but you have more than 2,000 stations, you can still use the selector console to track the *extra* unadministered stations and extend calls to them.

To do this, you must have a Group Select feature button. This feature button allows you to track and extend calls to stations that do not have Hundreds Select buttons administered for them. This feature button is also useful if the attendant prefers to use the dial pad to access a group of stations rather than pressing a Hundreds Select button.

The Group Select feature button works for extensions of 3-, 4-, or 5-digits (extension numbers 100 through 99999).

NOTE:
The Enhanced DXS Tracking feature supports only stations that reside on the same PBX as the attendant console. The attendant can extend calls to another DCS switch via the Enhanced DXS feature, but the DXS console does not show busy/idle status of stations on the other DCS.

To view extension status or extend calls using the Group Select button, you press the Group Select button and dial the 2- or 3-digit prefix of the extension you want and you press the pound (*#*) button. The lamps on the selector console light for all extensions in that group that are busy. For example, to view the status of extension 84463, you press the Group Select button, dial ***844,*** and press the pound ***(#)*** button. The lamps on the selector console indicate busy extensions for the group `84400-84499`. At this time, you can handle the call as normal. If you want to extend the call to a station in the group, you press the DXS button for that station.

If you press any Hundreds Select button on the DXS module and the console currently is in the Group Select mode, the console exits this mode and enters the normal mode.

The Group Display Button

You can administer a Group Display button to help you with tracking extension status. The purpose of the button is to enable you to see which group of extensions the system is currently tracking. You press this button and the system indicates on the display panel the range of extensions currently being tracked by the selector console.

NOTE:
You can administer the Group Display button for either the feature area or the display area of the console.

If you select this button, the system identifies the digits associated with a Hundreds Select button, unless it finds no Hundreds Select button is lit, in which case it identifies the digits last entered with the Group Select button. The system continues to track the selected group of extensions until you either press a Hundreds Select button, or press the Group Select feature button and dial the prefix for the new extension group and the pound (#) button. In either case, the system tracks the new group of stations.

Considerations

With the "Attendant Direct Extension Selection With Busy Lamp Field" feature, the attendant can place calls to 800 extensions using the basic selector console, and 2,000 extensions using the enhanced selector console; or up to 99899 extensions using the Group Select feature button (extension numbers from 100 to 99999). If the desired hundreds group is being tracked, using either a Hundreds Select button or the Group Select feature button, the attendant needs only to press the desired DXS button to access a station.

This feature also provides the attendant with a visual indication of the idle/active status of the extension numbers assigned to the selected hundreds group. A maximum of 100 extension numbers can be monitored for idle/active status at any one time.

The enhanced DXS functionality does not support extensions that have less than 3-digits.

Station tracking is possible only for the PBX on which the attendant resides.

Interactions

- Attendant Display

  When the attendant uses the "Attendant Direct Extension Selection With Busy Lamp Field" feature, the call is identified on the alphanumeric display through the "Attendant Display" feature.

- Call Coverage

  If Send All Calls is activated, or if the Call Coverage redirection criteria are met, then an extended call redirects to the coverage path.

- CAS

When a DXS button is used to make a CAS call, it takes a few seconds before the attendant hears ringback tone.

Administration

If you are using Hundreds Select buttons, you must administer the hundreds group assignment for each of the Hundreds Select buttons. Have your System Manager make these assignments on the 'Attendant Console' form.

If you are using the Group Select and Group Display feature buttons, you must have feature button assignments for GROUP-SEL and GROUP-DISP. Have your System Manager make these assignments on the 'Attendant Console' form.

Hardware and Software Requirements

Requires a selector console. No additional software is required.

Attendant Direct Trunk Group Selection

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the attendant direct access to an idle outgoing trunk by pressing the button assigned to the desired trunk group.

Each attendant console has 12 designated Trunk Hundreds Select buttons to be used with the "Attendant Direct Trunk Group Selection" feature. In addition, each console may have up to 12 of its feature buttons administered as additional Trunk Hundreds Select buttons, for a total of 24 Trunk Hundreds Select buttons per console. Each button allows the attendant direct access to an outgoing trunk group by simply pressing the button assigned to that trunk group.

If the attendant is talking on a call, then that call is split away and a new call is placed to the outgoing trunk specified by the trunk group select button. The attendant can then press release to connect the split away parties to the trunk's dial tone, or the attendant can dial the destination and press release to connect the split away party to the called party.

All Trunk Hundreds Select buttons (including any administered on the feature buttons) have a Busy lamp that lights when all trunks in the associated trunk group are busy. If one of the two-lamp feature buttons on a basic console is administered as a Trunk Hundreds Select button, the bottom lamp is used as the Busy lamp (the top lamp is not used). Six of the designated buttons (basic console) or all 12 designated buttons (enhanced console) also have a Cont (control) lamp and a Warn (warning) lamp. The Warn lamp lights when a preset number of trunks in the associated trunk group are busy. The Cont lamp lights when the attendant has activated the "Attendant Control of Trunk Group Access" feature for the associated trunk group.

Instead of trunk groups, Loudspeaker Paging zones can be assigned to Trunk Hundreds Select buttons. In this case, the Busy lamp indicates the idle/busy status of the associated Loudspeaker Paging zone.

Considerations

Attendant Direct Trunk Group Selection eliminates the need for the attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. A label associated with each Trunk Hundreds Select button identifies its destination or use, for example, Chicago, FX, or WATS. Pressing the button selects an idle trunk in the desired group.

Each attendant console has 12 designated Trunk Hundreds Select buttons. Each console may have up to 12 of its feature buttons administered as additional Trunk Hundreds Select buttons, for a total of 24 Trunk Hundreds Select buttons per console.

Interactions

If the "Attendant Control of Trunk Group Access" feature is provided, this feature must also be provided.

Administration

Attendant Direct Trunk Group Selection is assigned on a per-attendant basis by the System Manager. Administration consists of assigning trunk groups or Loudspeaker Paging zones to the Trunk Group Select button.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Display

Feature Availability

This feature is available with all Generic 3 releases.

Description

The "Attendant Display" feature shows call-related information that helps the attendant to operate the console more efficiently. It also shows personal service and message information. Information is shown on the alphanumeric display on the attendant console.

For G3i-Global and G3V2 and later releases, attendants can select one of several available display message languages. The language choices are English (default), French, Italian, or Spanish. The system administrator can also define the displays for one additional language. See the "Administrable Language Displays" feature for more information.

The following display modes can be assigned to the eight buttons in the display area of the console, or to any of the programmable feature buttons on the console. The Normal and Test modes are always provided; the others are optional.

- Normal Mode

  Displays call-related information for the active call appearance. The alphanumeric display is in the Normal mode unless the attendant selects one of the other modes. The display must be in the normal mode to answer incoming calls.

- Inspect Mode

  Displays call-related information for a call on hold or an unanswered call.

- Stored Number Mode

  Displays the number assigned to a button administered through the "Facility Busy Indication" feature or the number assigned to an Abbreviated Dialing button.

- Date/Time Mode

  Displays the current date and time of day.

- Test Mode

  Displays a test pattern representing each of the 40 characters that can be displayed. The Lamp Test switch is provided on the console; an additional button assignment is not needed.

- Elapsed Time

Displays elapsed time in hours, minutes, and seconds. The timing starts or stops when the button is pressed.

- Integrated Directory

  Turns off the touch-tone signals and allows the touch-tone buttons to be used to key in the name of a system user. After a name is keyed in, the display shows that name and associated extension number. (Refer to the "Integrated Directory" feature.)

- Coverage Message Retrieval Mode

  Retrieves and displays Leave Word Calling and Call Coverage messages for system users. Messages can be retrieved at any time. The attendant can be active on a call and still retrieve messages.

Three additional buttons should be assigned to the console when the Coverage Message Retrieval mode or the Integrated Directory mode is assigned:

- Next Message

  Retrieves and displays the next message, displays `NO MESSAGES`, or displays `END OF MESSAGES, (PUSH Next TO REPEAT)` when in the Coverage Message Retrieval Mode. Displays the next name in the alphabetical listing when in the Integrated Directory mode. This button should be assigned when the Retrieval mode button is assigned.

- Call Display

  Automatically returns the call requested by the currently displayed message or the currently displayed name and extension number. This button is optional.

- Delete

  Deletes the currently displayed message. This button must be assigned when the Retrieval mode button is assigned. This button is not used with the "Integrated Directory" feature.

With G3V4 and later releases, a Call Type button can also be administered for the Attendant Console. When pressed, this button displays the call type of the active call. The call type appears for one second and then the display returns to normal mode. The following list describes the three possible call types.

- Call Type 1: An outgoing public network call is classified as Type 1 when it is ringing or answered. An incoming call is classified as Type 1 when it is answered.

- Call Type 2: An incoming external public network call that has not yet been answered is classified as Type 2.

- Call Type 3: All other calls, that is, all internal calls, conference calls, and tie trunk calls of any type are classified as Type 3.

Call-Related Information

The system provides the following call-related information.

Call Appearance Identification

The attendant call appearance buttons are labeled alphabetically beginning with the letter ‘a’. The display shows, for example, a= for a call incoming on the first call appearance button, b= for a call incoming on the second call appearance button, and so on.

Calling Party Identification

When the call is from a system user, the display shows the caller’s name or a unique identification administered for the voice terminal being used, along with the calling party’s extension number. When the call is from outside the system, the display shows the trunk identification, such as `CHICAGO`, and the trunk access code assigned to the trunk group used for the call.

With the "Integrated Services Digital Network (ISDN) — Primary Rate Interface" feature, additional calling party information is provided. See the "Integrated Services Digital Network (ISDN) — Primary Rate Interface" feature for details.

Called Party Identification

On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party’s name and extension number. If no name is assigned, only the called party’s extension number is displayed.

On outgoing calls, the display shows the digits as they are dialed or the name and trunk access code assigned to the trunk group being used. The System Manager can suppress the name of any trunk group.

With the "Integrated Services Digital Network (ISDN) — Primary Rate Interface" feature, additional called party information is provided. See the "Integrated Services Digital Network (ISDN) — Primary Rate Interface" feature for details.

Internal COR

All system users have a COR to define their calling privileges. The COR is a two-digit number followed immediately by a hyphen and a four-character identifier. A COR button must be pressed to display a user’s COR. The COR information can be obtained from the System Manager. The restriction identifiers follow:

ORIG — Origination restriction

OTWD — Outward restriction

TOLL — Toll restriction

CODE — Code restriction

NONE — No restriction

Call Progress Feedback

Call Progress Feedback including `ringing`, `busy`, and `call waiting` is presented if a G3i-Global or G3V2 or later release switch is used.

Call Purpose

This refers to calls directed, redirected, or returning to the console. The English language call purpose identifiers follow:

an — `Attendant No Answer` — indicates the call was redirected to another attendant because the ‘‘attendant no answer’’ and ‘‘attendant alerting interval’’ timers expired.

co — `Controlled Outward Restriction Call` — indicates a call from an internal user has been redirected to the attendant because the user has Controlled Outward Restriction and has attempted to make an outgoing call.

ct — `Controlled Termination Restriction Call` — indicates a call has been redirected to the attendant because a user has Controlled Termination Restriction and the calling party has tried to call that user.

cs — `Controlled Station-to-Station Restriction Call` — indicates a call from an internal user has been redirected to the attendant because the user has Controlled Station-to-Station Restriction and has tried to make a station-to-station call.

da — `DID Recall` — indicates the Central Office operation has activated the DID recall feature when the switch disconnected first from an incoming DID call placed by the operator. The resulting call is routed to the attendant group. DID Recall is used only in some countries.

hc — `Held Call` — indicates, for G3iV1, G3vs/G3sV1, G3i-Global, and G3V2 or later release, that a held call for the attendant has been on hold for longer than the ‘‘held call’’ timeout value.

ic — `Intercept Call` — indicates the incoming call has been redirected to the attendant as a result of Intercept Treatment.

ip — `Inter-Position Call` — indicates the incoming call was placed to the attendant’s individual extension by another attendant.

ld — `DID LDN Call` — indicates the incoming call is a Listed Directory Number (LDN) call on a DID trunk.

n — `night service` — indicates the call was routed to the attendant due to night service on a trunk group or hunt group.

na — `no answer` — indicates an incoming DID call was unanswered for longer than the ‘‘no answer’’ timeout value.

p — `Call Pickup` — indicates the attendant answered the call via the "Call Pickup" feature.

pc — `Personal Call` — indicates the incoming call was placed to the attendant’s individual extension, and not to the attendant group.

rt — `Returned Call` — indicates an attendant-extended call was not answered within the administered interval and the call has returned to the console.

qf — `Queue Full` — indicates the call has been redirected to this attendant (or switch) because the emergency queue is full.

rc — `Recall Call` — indicates an internal user, active on a call held on the console, is requesting attendant assistance.

sc — `Serial Call` — used in G3i-Global and G3V2 or later release to indicate a serial call.

tc — `Trunk Control` — indicates an internal user attempted to access an attendant-controlled trunk and the call was redirected to the console.

f — `Call Forwarding` — indicates an internal user has calls forwarded automatically to the attendant.

When the "Call Coverage" feature is active and the attendant is a covering user, the following call purpose identifiers are displayed:

s — `Send All Calls` — indicates the called voice terminal user is temporarily sending all calls to coverage.

d — `Don’t Answer or Cover` — indicates the called voice terminal was not answered or the calling system user has sent the call to coverage, or the called voice terminal user is not available. This identifier also indicates the called voice terminal user has a temporary bridged appearance of the call.

b — `Busy` — indicates the called voice terminal user is active on a call, and the called voice terminal user has a temporary bridged appearance of the call.

B — `Busy` — indicates the called voice terminal user is active on a call, and the called voice terminal user does not have a temporary bridged appearance of the call. All calls to single-line voice terminals that go to coverage displays `B`.

c — `Cover All` — indicates the called voice terminal user has had all calls redirected to coverage.

sc — `Serial Call` — indicates the incoming call is a serial call.

Sample Displays

The attendant console has a one-line 40-character alphanumeric display. Some typical displays follow:

Internal call originated by the attendant:

a=3602

then

a=   TOM BROWN   3062

or

a=   EXT 3602   3602

Outgoing trunk call originated by the attendant:

b=87843541

Where 8 is the trunk access code and 784-3541 is the number dialed.

then

b=   OUTSIDE CALL   8

or

b=   WATS   101

Where 101 is the trunk access code of the outgoing trunk group.

Incoming trunk call to the attendant:

a=   OUTSIDE CALL   102

Where 102 is the trunk access code of the incoming trunk group.

Conference call originated by the attendant:

b=   CONFERENCE 4

Where 4 is the number of conferees. The number does not include the attendant.

Internal call redirected to coverage:

b=   EXT 3174 to EXT 3077   d

or

b= BOB SMITH to JOYCE THOMAS d

Where d indicates that Go to Cover was activated by the calling voice terminal user.

Incoming trunk call redirected to coverage:

b= OUTSIDE CALL to DON SMITH s

Where s indicates that Send All Calls was activated by the called voice terminal user.

Coverage Message Retrieval

IN PROGRESS

then

MESSAGES FOR BETTY R. SIMS

then

JOE JONES 10/16 11:40a 2 CALL 3124

This message means that Joe Jones called Betty R. Sims the morning of October 16. The second message was stored at 11:40 a.m. Joe wants Betty to call his extension number, 3124.

Integrated Directory mode:

CARTER, ANN 3408 3

This display shows the name and extension number as administered in the system. The 3 indicates that three buttons were pressed to reach this particular display.

Considerations

If the attendant group is administered for system-wide message retrieval, attendants can retrieve messages for voice terminal users. Permission to have coverage message retrieval must also be administered for the voice terminal user. It is not possible for selected attendants to retrieve messages for selected voice terminal users.

The display must be in the Normal mode for the attendant to answer incoming calls. See *DEFINITY Communications System Generic 1 and Generic 3 Console Operation*, 555-230-700, for details on Incoming Call Identification.

Interactions

With the "Bridged Call Appearance— Multi-Appearance Voice Terminal" feature, a call from the primary extension number or a bridged call appearance of the primary extension number is displayed as a call from the primary extension number.

Hospitality

When Hospitality Services are optioned, incoming trunk calls that return to the attendant after being extended by the attendant to a client room do not have the standard return call display but show the client name and room number along with the ***rt*** Indicator. Client room calls to the attendant that return to the attendant after being extended to a second client room display the client name and room number with a ***ex*** Reason Indicator.

NOTE:
The ***ex*** Reason Indicator does not appear for any other type of call.

Administration

The "Attendant Display" feature is administered on a per-attendant basis by the System Manager. Administration consists of assigning feature related buttons to each attendant console. The following buttons can be assigned:

- Call Display (optional, used with the Retrieval mode or the Integrated Directory mode)
- Call Type
- Coverage Message Retrieval
- Date and Time (one button)
- Delete Message (must be assigned if the Coverage Message Retrieval button is assigned)
- Elapsed Time
- Inspect Mode
- Integrated Directory
- Next Message (must be assigned if the Coverage Message Retrieval button is assigned)
- Normal Mode
- Stored Number
- COR

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Intrusion (Call Offer)

Feature Availability

This feature is available for all G3 switches.

Description

The "Attendant Intrusion (Call Offer)" feature enables an attendant to enter an existing call on either a multifunction station or analog station to offer a new call or message to the called party. Upon intrusion, tone may be applied if administered. Upon the attendant's release from the intruded call, the calling party's call is held by the "Call Waiting Termination" feature for the analog intruded-on party or rings on an available line appearance for multifunction set. The intruded-on party must be on an analog station in order for the "Call Waiting Termination" feature to be activated.

Considerations

Only attendant users can activate this feature by pressing the intrusion button.

Interactions

The following are interactions with other features:

- If a station is on a conference call with administered maximum number of conferees, the attempt to intrude on the station is denied.
- If there is one call already Call Waiting for the intruded-on party, the source (split from attendant) party is unable to wait for the intruded party using Call Waiting.
- If a call is established with Data Privacy activated, the attempt to intrude on the call is denied.
- If a station in a call is administered with Data Restriction, the attempt to intrude on the call is denied.
- If an attendant attempts to intrude on a call on a station which is a forward-to point of another station, the intrusion is denied.
- If an attendant attempts to intrude on a busy station where the station is talking to another attendant, the intrusion is denied.
- The attendant display shows the character `l` adjacent to `wait` or `busy` if an intrusion is possible. Otherwise, the display shows `wait` or `busy`.
- G3i-Global, G3V2, and later releases provide Attendant Intrusion on Satellite PBX stations via TGU/TGE trunks.

Administration

An Attendant Intrusion button is required at the attendant console. There can be only one button per console. If intrusion tone is desired, this must be enabled on the ‘‘System Parameter’’ form.

Hardware/Software Requirements

No special hardware is required for this feature in a stand alone system.

Attendant Override of Diversion Features

Feature Availability

This feature is available with all Generic 3 releases.

Description

Diversion Override enables an attendant to bypass any diversion features invoked by and/or associated with a dialed extension. A diversion feature is any feature that, when activated, causes the call to alert at a point different from the dialed station. Specifically, the diversion features are "Send All Calls", "Call Coverage", and "Call Forwarding All Calls" including cases in which the call alerts at the dialed station and is later transferred (as in the case of Busy Don't Answer).

Depressing the OVERRIDE button before originating a call invokes the feature and causes the OVERRIDE lamp to light.

If an invalid extension is dialed, the override button lamp gives denial. If the CANCEL, RELEASE, or FORCED RELEASE button is depressed during dialing, the diversion override feature is deactivated. A second depression of the button deactivates the feature and extinguishes the lamp. The feature is also deactivated when the call to the dialed extension is terminated, or if a trunk access code is dialed.

Activation or deactivation of the override feature while dialing is permitted.

Considerations

This feature together with "Attendant Intrusion (Call Offer)" feature can be used to get an emergency or urgent call through to a station user.

Interactions

Calls directed to a station on which diversion features are active, which would normally cause the call to terminate on another extension, are treated as though the diversion features are not active.

Administration

Only one OVERRIDE button is allowed per console.

Hardware/Software Requirements

No special hardware requirements exist for this feature.

Attendant Priority Queue

Feature Availability

This feature is available with all Generic 3 releases. Priority by call type is available with G3V4 and later releases.

Description

The "Attendant Priority Queue" feature handles incoming calls to the attendant group or to an individual attendant when the call cannot be immediately terminated to an attendant. These calls are placed in the Attendant Priority Queue in an order based on a priority queue level and timestamp associated with the call. Calls within the same priority level are served on a first come, first serve basis. The calling party hears ringback until the call is answered by an attendant. Priority queuing is not transparent over DCS.

Twelve different categories of incoming attendant calls are defined. Each has a designated Attendant Priority Queue level. Although each of these categories is given a default level, the user can specify a priority queue level for any or all categories via a system administration function. There are twelve priority queue levels available.

The category for ‘‘Emergency Call to the Attendant Group’’ always defaults to the highest priority queue level: all other categories default to a lower priority.

A call placed in the Attendant Priority Queue is associated with one of the following priority queue categories:

- Emergency Call to the Attendant Group - Originated by a station user who has dialed the customer- administrable emergency access code.
- Assistance Call to the Attendant Group - Originated by a station user who has dialed the attendant group access code, or originated by a station that has the "Manual Originating Line Service" feature activated.
- Attendant Group Call over a CO/FX/WATS Trunk - An incoming trunk call directed to an attendant group (this does not include trunk calls returned to the attendant group after a timeout or some type of deferred attendant recall).
- Attendant Group Call over a DID Trunk - Same as previous category except for a DID trunk.
- Attendant Group Call over a Tie Trunk - Same as previous category except for a tie trunk (dial-repeating or direct types).
- Redirected DID or Redirected ACD Call - A DID or ACD call which times out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable and is rerouted back to the attendant group.

- Redirected Call - A call initially assigned to one attendant, but subsequently reassigned to the group because the position has become busy.
- Attendant Return Call - A call returned to the attendant after a timeout of an extended station or trunk call. Such a call is intended to return to the attendant who extended it. However, if that console is busy on another call, the extended call is returned to the attendant group. This category is a type of Attendant Redirected Call with its own identity to allow assignment of a Attendant Priority Queue level.
- Serial Call - Originated by the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a station, and then the station user goes on-hook. If the attendant who extended the serial call is busy on another call, the serial call is redirected to the attendant group.
- Individual attendant access call - Originated by a station user, incoming trunk call, or a system feature which uses the Individual Attendant Access (IAA) extension to direct a call to a specific attendant. If the individual attendant is busy on another call, the Individual Attendant Access call is queued until the individual console is idle. Then the queued IAA call is terminated to the individual attendant console.
- Interposition Call - Originated by one attendant who directs a call to another attendant by dialing the Individual Attendant Access extension. This category is a type of Individual Attendant Access call with its own identity to allow assignment of a Attendant Priority Queue level.
- Miscellaneous Call - Other calls (such as Automatic Circuit Assurance calls) not covered in the above call categories.

A priority level is assigned to each category, so that calls are answered on a priority basis. The assignment of a priority level to each category is administrable. The same priority level can be assigned to more than one category. By assigning all categories the same priority level, a first-in/first-out queue is achieved.

Calls are prioritized within a Attendant Priority Queue level on a first-come first-serve basis by a timestamp associated with each call. This timestamp indicates the relative time (with respect to all calls in the queue) when a call was placed in the Attendant Priority Queue after attempting to terminate to the attendant group or an individual attendant.

When at least one call is queued in the Attendant Priority Queue, the Calls Waiting lamp is lighted steady on all active attendant consoles. If the number of calls in the queue reaches the customer-administrable attendant group calls-waiting threshold, the Queue Warning lamp is lighted steady on all active attendant consoles.

Priority by Call Type

With DEFINITY Communications System G3V4 and later releases, it is possible to further define the priority assigned to calls in the Attendant Priority Queue. Calls are first assigned a priority level based on one of the twelve priority categories. Then, within each priority category, calls are assigned a further priority based on call type. Then, within each call type, calls are prioritized by time (first-in first-out). The following list describes the call types.

- Call Type 1: An outgoing public network call is classified as Type 1 when it is ringing or answered. An incoming call is classified as Type 1 when it is answered.
- Call Type 2: An incoming external public network call that has not yet been answered is classified as Type 2.
- Call Type 3: All other calls, that is all internal calls, conference calls, and tie trunk calls of any type are classified as Type 3. This includes calls over RLTs, DCS, inter-PBX attendant calls, and other private networking features.

Within each priority category, Type 1 calls have priority over Type 2 calls, which have priority over Type 3 calls. In this way external public network calls have priority over all other calls including conference calls. And, answered external public network calls have priority over public network calls that have not yet been answered.

With G3V4 and later releases, it is also possible to assign a Call Type button on the attendant console. When pressed, this button displays the Call Type of the active call.

Considerations

Note that even though the "Attendant Priority Queue" feature may reroute an incoming call from an individual attendant to the attendant group under some conditions, the routing reason (and hence the associated Attendant Priority Queue level) for the call is not typically changed. Hence, the reason code display on the answering attendant’s alphanumeric display remains the same for the rerouted call.

Interactions

In general, interaction with System Features that cause a call to be routed to the attendant group or an individual attendant assign a ‘‘routing reason’’ to the call. Each routing reason is used by the "Attendant Priority Queue" feature to determine the Attendant Priority Queue category (and hence the particular Priority Queue level) for the call.

Therefore, the routing reason designated by a feature determines the priority of a call in the "Attendant Priority Queue". Note that even though the "Attendant

Priority Queue" feature may reroute an incoming call from an individual attendant to the attendant group under some conditions, the routing reason (and hence the associated Attendant Priority Queue level) for the call is not typically changed.

- Individual Attendant Access Call

  An Individual Attendant Access (IAA) call requires special handling by the "Attendant Priority Queue" feature. If the console to which an IAA call is directed is in the active mode because the handset/headset is plugged in and the module associated with the individual console is operational, a queued IAA call is not terminated to the individual attendant until that attendant is able to receive the call.

  If the handset/headset for the individual attendant console is unplugged, or the module associated with that attendant is not operational, a queued IAA call is rerouted to the global attendant queue.

Interactions for Priority by Call Type (G3V4 and later releases)

- Multi-Party Calls

  Multi-party calls are always treated as Type 3 calls when placed into the Attendant Priority Queue. If a multi-party call becomes a single-party call while in the queue due to one or more parties disconnecting, it remains a Type 3 call.

- Night Service - Hunt Group

  When Night Service Hunt Groups are used, calls may be retrieved from hunt groups rather than the Attendant Priority Queue. Since the Call Type prioritizing is not applied to hunt groups, calls may not be retrieved in order of Call Type when Night Service calls terminate in hunt groups. However, if and only if the Attendant Priority Queue is used as the termination for Night Service calls, ordering by Call Type can be maintained.

- Off Premises Station

  Calls from Off-Premises Stations (OPS) always are identified as Type 3 calls.

Administration

System administration functions are provided to allow the customer to:

- Designate Attendant Priority Queue levels for any or all Attendant Priority Queue call categories (including the highest priority level).
- Designate the attendant group calls-waiting threshold, which is used to light the Queue Warning Indication lamp on all active consoles if that threshold is exceeded.
- Indicate whether calls should be subordered by Call Type.

Note that the “Emergency Call to the Attendant Group’’ category is set as a default to the highest level of 1, and all other call categories are set to a lower priority of 2. If an attempt is made to change the Emergency Call category from the highest priority level, a warning message is displayed on the administration terminal.

For instructions for administering a Call Type button for an attendant console, see "Attendant Console" of the *DEFINITY Communications System Generic 3 Version 4 Implementation* manual, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

Hardware/Software Requirements

There is no special hardware required for this feature.

Attendant Recall

Feature Availability

This feature is available with all G3 releases.

Description

Allows voice terminal users on a two-party call, or on an Attendant Conference call held on the console, to recall the attendant for assistance.

Single-line users press the RECALL button or flash the switchhook to recall the attendant.

Multi-appearance users press the Conference or Transfer button to recall the attendant, and remains on the connection when either button is used.

Considerations

Attendant Recall provides a convenient means for a voice terminal user, on a call held on the console, to recall the attendant if further assistance is required.

The call must be held on the console.

Interactions

The following features interact with the "Attendant Recall" feature.

- Individual Attendant Access

  If a hunt group call to an individual attendant is being held on the console, a system user, active on the call, cannot recall the attendant. However, he or she can transfer calls or make conference calls.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Release Loop Operation

Feature Availability

This feature is available with all G3 releases.

Description

Allows the attendant to hold the connection of any call off the console if completion of the call is delayed (such as a call extended to a busy single-line voice terminal or to a voice terminal that does not answer). This feature frees the attendant to handle other calls.

When an incoming call arrives on a call appearance at an attendant console and is answered, extended, and released by the attendant, the call is released from that call appearance. The console is then available to receive the next call.

Timed Reminder (Return Call Time-out) starts once the call is off the console. If the called terminal or coverage point user does not answer before the administered interval expires, the call returns to the attendant queue. Once the call comes out of queue and terminates at a console, the special recall tone is applied and the alphanumeric display shows the call identification. G3i-Global and G3r Versions 1 apply the normal ringing tone to recall calls. For Version 2 and later releases, the ringing may be administered to apply either secondary ringing, (recall ringing) or primary ringing (incoming call ringing).

For G3i-Global, G3rV1, G3V2, and later releases, Timed Reminders attempt to return the call to the attendant who previously handled it; only when the original attendant is unavailable are calls returned to the queue. For G3vsV1/G3sV1, and G3iV1, Timed Reminders are returned to the attendant queue for distribution to the first available attendant (which is not necessarily the attendant who previously handled the call).

Once a Timed Reminder occurs, the call is returned to the queue and the called station is no longer notified of the call.

Considerations

Attendant Release Loop Operation improves efficiency in handling calls by allowing the attendant to release from a call without having to wait for an answer. The attendant is immediately available to handle other calls.

Interactions

The following features interact with the "Attendant Release Loop Operation" feature.

- Timed Reminder

  Timed reminder tone is provided by this feature.

  The Return Call Time-out interval is provided by this feature.

- Attendant Display

  Call identification is provided by this feature.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Room Status

Feature Availability

This feature is available with G3V3 and later releases.

Description

Beginning with G3V3, the DEFINITY Communications System offers room status access from the attendant console. The "Attendant Room Status" feature enables an attendant to see whether a room is vacant or occupied and what the housekeeping status of each room is without having to use a property management system (PMS).

Feature History and Description

This feature is designed for those customers who require the ability to quickly get room status using the DXS module without having to have a PMS. It is modeled after two features in the PMS Interface: ‘Check-In/Check-Out’ and ‘Housekeeping Status’. These two features combined make up the "Attendant Room Status"feature.

The Dimension 400 PBX and Dimension 2000 PBX systems contained a feature similar to "Attendant Room Status", which was called ‘Room Status and Selection’. This feature was part of ‘Dimension’s Hotel/Motel’ feature package. It provided the capability for the switch to store and display the occupancy and cleaning status of each guest room. Dimension also provided the first interface to a general trade PMS. However, in the systems that followed Dimension, beginning with System 75, the ability to display this information on an attendant console was based solely on a PMS. In other words, without a PMS, the room status information could not be seen easily. The switch systems still stored the information and the information was available by printing it on hard copy (as is still the case with all the DEFINITY Communications System releases), but the switches no longer displayed the information on an attendant console.

Therefore, the new "Attendant Room Status" feature, which allows you to see room status on the attendant console without a PMS, fills the gap in releases that came between Dimension and the DEFINITY Communications System G3V3.

Check-In/Check-Out Status

Check-in/Check-out status is available by pressing the OCC-ROOMS (occupied rooms) button on the attendant console. Rooms are either occupied or vacant. This function enables you to identify which rooms are occupied and which are vacant.

NOTE:
The OCC-ROOMS button must be administered on the 'Attendant Console' form.

When you press the OCC-ROOMS button on the attendant console, the console enters the Check-in/Check-out mode and the DXS lamps light for all rooms that are occupied (that is, checked in) and stay unlit for all rooms that are vacant (that is, checked out). Also, the console displays a message indicating that it is in the Check-in/Check-out mode. As with other display messages, this message may be administered for a specific language.

If your console includes hundreds keys, you can view the status of stations in hundred groups using the same procedure you would follow to view idle/busy status in hundreds groups: by pressing the appropriate hundreds buttons.

The occupancy status normally updates as guests are checked in and out. While the console remains in this mode, it updates lamps for any rooms that change status.

Maid Status

Maid status is available by pressing the MAID-STAT (maid status) button on the attendant console. A room may be in one of six maid states (also known as housekeeping states). This function enables you to see which rooms are in a specified maid state.

NOTE:
The MAID-STAT button must be administered on the "Attendant Console" form.

When you press the MAID-STAT button on the attendant console, the console enters the Maid Status mode and prompts you to enter a status number (1 through 6). If the attendant enters an incorrect number, the system displays an error message. When the attendant enters a valid Maid Status number, the display indicates the status that corresponds to that number and lights the DXS lamps for all rooms currently in that status. The DXS lamps are unlit for all rooms that are not in that state.

While the console is in Maid Status mode, the attendant can view another status by pressing the status number — that is, the attendant does not need to repress the MAID-STAT button. When the attendant presses a new status number, the console display identifies the new status and the DXS lamps indicate the rooms that are in the new maid status.

The cleaning status of a room is updated when a maid or inspector dial from the room and change the status. While the console remains in the Maid Status mode, it automatically changes the lamps on the DXS module when a room's Maid Status changes. That is, while the console and DXS lamps display rooms in one status, and a room changes to or from that status, the lamp for that room automatically changes.

If your console includes hundreds keys, you can view the status of stations in hundred groups using the same procedure you would follow to view idle/busy status in hundreds groups by pressing the appropriate hundreds buttons.

The Maid Status messages that appear on the console identify each of the six states and are user-defined. For example, you might define room state 1 as ***clean, ready to use*** and room state 2 as ***occupied, needs cleaning***. The messages are administered on the 'System-Parameters Hospitality' form. Your system administrator has the ability to change these messages to display appropriate information. As with other display messages, the Maid Status prompts and status messages may be administered for a specific language.

Interactions

When the console is in the Check-in/Check-out mode, the attendant may make outgoing calls via the key pad. The only difference between this mode and Normal mode is in the state of the DXS lamps. The attendant can return to Normal mode or any other mode by pressing the appropriate button on the console. The Check-in/Check-out mode does not affect other attendant operations.

When the console is in the Maid Status mode, the attendant may not make outgoing calls via the key pad. However, the attendant may make calls via the DXS module or the feature buttons. The console remains in the Maid Status mode until the attendant changes it to another mode. To change to Normal or another mode, press the appropriate button on the console. When the console is no longer in Maid Status mode, the DXS lamps return to the normal state (showing whether a station is busy or idle).

Considerations

Only attendants may use this feature. The feature is not available for attendants connected to the switch by a distributed communications system (DCS). Also, attendant consoles are the only terminals that can access this feature.

Administration

Attendant Console Form

The OCC-ROOM and MAID-STAT buttons must be administered on the ‘Attendant Console’ form.

System-Parameters Customer-Options Form

The “Enhanced Hospitality” field must be enabled on the ‘System-Parameters Customer-Options’ form. Note that this field cannot be enabled unless the G3 version is V3 or later. The Hospitality option must be enabled on the ‘System-Parameters Customer-Options’ form before you can administer associated forms and fields. See your AT&T support representative for assistance with the ‘System-Parameters Customer-Options’ form.

System-Parameters Hospitality Form

The six maid state display messages should be administered on the ‘System-Parameters Hospitality’ form. Defaults are:

for maid status 1: “Rooms in State 1’’
for maid status 2: “Rooms in State 2’’
for maid status 3: “Rooms in State 3’’
for maid status 4: “Rooms in State 4’’
for maid status 5: “Rooms in State 5’’
for maid status 6: “Rooms in State 6’’

‘Language Translations Property-Management’ form

Translations for display messages may be administered on the ‘Language Translations Property-Management’ form.

Hardware and Software Requirements

You must have a DXS module to use this feature.

Attendant Serial Calling

Feature Availability

This feature is available with all G3 releases.

Description

The "Attendant Serial Calling" feature enables the attendant to transfer trunk calls that return to the same attendant position after the called party hangs up. The returned call may then be transferred to another station within the switch and this can continue to recur. This feature is particularly useful if trunks are scarce and Direct Inward Dialing services unavailable. This can cause an outside caller to have to redial often to get through to a location because trunks are so busy. Once callers have been able to get through to a switch attendant and have several calls to make to others on the switch, this feature permits them to keep the use of the line into the switch until all their calls are completed.

The Attendant's display shows that the incoming call is a Serial Call. This information is displayed in the Call Purpose area (far right hand side) of the display. The reason code displayed is sc.

Once the "Attendant Serial Calling" feature has been activated it remains activated until either the trunk drops from the switch or the attendant deactivates the feature manually (by depressing the Serial Call button). Once the attendant answers the serialized call the lamp associated with the serial call button is turned on. If that button is not administered then the feature is still activated, however no external indication shows that the feature is active (except the attendant's display). If an attendant received a serialized call but has no serial call button then the feature cannot be deactivated until the trunk hangs up or until an attendant with a serial call button becomes the controlling party.

If no attendants are available then the call is placed in the attendant's priority queue.

Considerations

Only attendant users can activate this feature. The "Attendant Serial Calling" feature is only valid on calls that have only one trunk on the connection. Only one serial call button may be administered per attendant console. The Serial Call button cannot be assigned to an analog or digital station. The feature can be activated on a conference call as long as only one trunk is on the conference.

Interactions

The following features interact with the "Attendant Serial Calling" feature:

- Priority Queue: A special priority exists for serialized calls.
- CDR/Call Charging: CDR/Call Charging must consider that a single outgoing trunk call may have to be broken up and charged to the different parties that dealt with the call.
- The "Attendant Serial Calling" feature only works in a DCS environment if the attendant who is activating the "Attendant Serial Calling" feature resides on the same node as the trunk that the attendant is currently connected to. In addition, the attendant must not be conferenced in with a DCS party when activating Serial Call since this would have two trunks on the connection that is not allowed for Serial Call activation.

Administration

The Serial Call button can only be administered on an attendant console on the “Console Attendant” form.

Hardware/Software Requirements

No special hardware is required for this feature.

Audible Message Waiting

Feature Availability

Audible Message Waiting is available with Generic 3rV1 and all Generic V2 and later releases. However, this feature may not be applicable in countries that restrict the characteristics of dial tones provided to users.

Description

Audible Message Waiting is typically, but not necessarily, used on voice terminals without message waiting lights. (Often these are analog terminals.) The feature places a stutter at the beginning of station dial tone on a station that has a message waiting for it. The message can be waiting in system memory (to be accessed via display or via the voice synthesizer), PMS, MSA, or AUDIX.

Considerations

The system administrator must inform the user where to call to retrieve messages that can only come from one messaging system source.

Interactions

None

Administration

Audible Message Waiting is first enabled on a system-wide basis on the 'System Parameter Customer Option' form. When turned on for the system, an additional field appears on the 'Station' form for enabling the feature on a station-by-station basis.

Hardware and Software Requirements

No additional hardware or software is required. Requires a separate software right to use fee.

Audio Information Exchange (AUDIX) Interface

Description

AUDIX is a message-handling system for recording and distributing spoken messages or voice mail. The system contains stored voice prompts that guide users when creating, sending, retrieving, answering, saving, and forwarding spoken messages.

NOTE:
Your administrator is urged to consult the *GBCS Products Security Handbook,* 555-025-600, for information on measures to help secure this feature from possible toll fraud. Also see the "Security Violation Notification (SVN)" feature.

The following is a non-inclusive list of AUDIX applications. This list depends on the type of AUDIX system you are using:

- Voice Mail — allows users to send and receive messages to and from their voice mail
- Call Answer — provides switch extensions answer coverage via AUDIX
- Automated Attendant — presents callers with a voice menu of options, then routes calls according to the keys the caller presses
- Bulletin Board — (sometimes called Information Service) plays a recorded message to callers
- Broadcast Message — enables an AUDIX administrator to send broadcast messages to all AUDIX users on the system
- Directory Service — provides a directory of subscribers to the caller

The following activities are available for use by AUDIX subscribers:

- Create Message — Record or modify a new message, address it, schedule it for delivery, and save a copy (optional).
- Scan Incoming Mail— Review new messages and reply or redirect them with an added comment, and review or delete old saved messages.
- Personal Greeting Administration — Record or modify one or more personal greetings to be played for callers who reach AUDIX through the Call Answer feature, or select the standard AUDIX greeting.
- Scan Outgoing Mail— Review, modify, or redirect messages scheduled for delivery; check the status of delivered messages; and review, modify, redirect, or delete messages saved in the file cabinet.

- Password and List Administration — Change user's personal AUDIX password and create, modify, review, or delete mailing lists.
- AUDIX Networking — Send and receive network messages, status information, and administrative update information to and from other members of the AUDIX product family.

NOTE:
Two types of networking are available: Audio Message Interchange Specification (AMIS) analog networking and digital networking. AMIS networking is available with all AUDIX configurations; digital networking is available when AUDIX is configured as a system external to the DEFINITY switch, as discussed below.

Several documents, quick references, and other material that fully describe how to use AUDIX are available through the Customer Information Center.

Several versions of AUDIX are available: the first, referred to as ''DEFINITY AUDIX,'' is a system comprised of circuit packages resident in the switch; the second, referred to as ''AUDIX,'' is a system that is external to the DEFINITY switch and connected to the switch by station lines and data links. Additionally, AUDIX systems can be networked through switches or other AUDIX machines. Examples of such systems are AUDIX Voice Power™ (VP), AUDIX VP Lodging and AUDIX VP Auto Attendant. They rely on a data link between the AUDIX adjunct on the switch.

The following two sections describe only the ''DEFINITY AUDIX'' and ''AUDIX'' offerings.

DEFINITY AUDIX

The DEFINITY AUDIX offering is a circuit package *sandwich* assembly that fits inside the DEFINITY system carrier and requires five contiguous slots in the switch carrier. For a full description of DEFINITY AUDIX, see the following DEFINITY AUDIX documents:

- *DEFINITY AUDIX System — System Description*, 585-300-205
- *DEFINITY AUDIX System — Feature Descriptions*, 585-300-206
- *DEFINITY AUDIX — Documentation Guide*, 585-300-011
- *Planning for DEFINITY AUDIX System*, 585-300-904
- *DEFINITY AUDIX System Installation*, 585-300-111
- *DEFINITY AUDIX System Installation Checklist*, 585-300-109
- *Switch Administration for DEFINITY AUDIX System*, 585-300-509
- *DEFINITY AUDIX System Administration*, 585-300-507
- *DEFINITY AUDIX System Forms Reference*, 585-300-207
- *AUDIX Administration and Data Acquisition Package*, 585-302-502

- *AMIS Analog Networking*, 585-300-512
- *A Portable Guide for Voice Messaging*, 585-300-701
- *Voice Messaging Quick Reference*, 585-300-702
- *Voice Messaging Subscriber Document Artwork*, 585-300-703
- *Voice Messaging Wallet Card*, 585-300-704
- *AUDIX Business Card Sticker*, 585-304-705
- *DEFINITY AUDIX System Maintenance*, 585-300-110
- *DEFINITY AUDIX System Announcement Customization*, 585-300-516

For details on DEFINITY AUDIX, refer to the above documents.

AUDIX (external to the DEFINITY Switch)

NOTE:
This section describes AUDIX systems external to the DEFINITY switch and connected to the switch by station lines and a data link. Do not refer to this section if AUDIX is configured using circuit packages resident in the DEFINITY switch. For those systems, called DEFINITY AUDIX systems, see the previous section, entitled "DEFINITY AUDIX".

The AUDIX offering is an interface between the DEFINITY system and an AUDIX consisting of up to 32 analog (voice) connections for exchange of voice messages, and a data link for status and control information exchange. An AUDIX adjunct is available in both one-cabinet and two-cabinet configurations. The one-cabinet configuration provides up to 16 ports. The two-cabinet configuration provides up to 32 ports. The analog port interface on the switch can be provided by TN742, TN746B, or TN769 circuit packs.

Up to eight AUDIX systems may be connected to a Generic 3r, or G3vs/G3s switch, but only one AUDIX system may be directly connected to a Generic 3i switch. However, all switches allow the use of AUDIX in a DCS arrangement. Each switch can have its own AUDIX which serves only the users connected to that switch; or a single AUDIX connected to the switch may serve other switches in a DCS network.

For a full description of AUDIX, see the following AUDIX documents:

- *AUDIX Administration*, 585-305-501
- *AUDIX Administration and Data Acquisition Package*, 585-302-502
- *AUDIX Call Detail Recording Package*, 585-305-506
- *AUDIX Documentation Guide*, 585-300-010
- *AUDIX Feature Descriptions*, 585-305-203
- *AUDIX Installation*, 585-305-105

- *AUDIX Networking*, 585-300-903
- *Planning for DEFINITY AUDIX System*, 585-300-904
- *AUDIX System Description*, 585-305-201
- *AUDIX Training Tape*, 585-300-050
- *AUDIX Upgrade Instructions*, 585-302-108
- *Switch Administration for AUDIX Voice Messaging*, 585-305-505

Refer to the *AUDIX Documentation Guide*, 585-300-010, for the latest issue numbers and release-specific information.

Security Measures

Fraudulent long-distance calls may be placed through your voice mail and auto attendant system if proper security precautions are not implemented.

The following actions help secure your system from unauthorized use.

- For both AUDIX and DEFINITY AUDIX:
  - To help prevent exchange of information through unassigned Voice Mail, remove any unused or unassigned mail.
  - Secure the system lines that serve AUDIX and control where calls can be placed. Pay special attention to assigning restrictive COR, COS, and FRL to the station lines and trunks serving AUDIX. Use switch CDR reports to determine if the station lines connected to AUDIX are being used for calls that are normally not within your sphere of business.
  - Change default passwords on voice mailboxes immediately after installation and use random numbers for passwords.
  - Require passwords at least five digits long for user mailboxes.
  - Change all system administration passwords to alphanumeric codes.
- For AUDIX only:
  - To help prevent having billable calls placed from unauthorized transfers outside of the system, turn on the Enhanced Call Transfer (ECT) feature. This feature is available in later AUDIX systems that connect to an AT&T digital switch. ECT performs call transfer over the data link between AUDIX and the DEFINITY switch. (The destination extensions must be administered on the switch dial plan.)

- For DEFINITY AUDIX only:
  - — The administrator determines whether to allow transfer only to another AUDIX subscriber or to any extension of the correct extension length (that is, the number of digits for extensions administered through the DEFINITY switch.) For example, you may be configured to support the four-digit plan, the five-digit plan, etc. The most secure approach, which is the default, is to only allow transfers to other AUDIX subscribers. If the administrator decides to allow transfers to any extension, then you should administer the COR on the AUDIX ports to prevent calls outside the PBX.
- Both AUDIX and DEFINITY AUDIX provide a maintenance/administration port. To prevent unauthorized access to this port, a remote port security device provides an added layer of security.

Consult the *GBCS Products Security Handbook,* 555-025-600, for additional steps to secure your system and to obtain periodic information about security developments.

Considerations

For DEFINITY AUDIX, you must have five contiguous slots in the switch carrier.

In addition to supporting multiple AUDIX systems, the DEFINITY system can have multiple hunt groups associated with a single AUDIX system. This allows partitioning of the voice ports into different hunt groups and different coverage paths to cover different voice ports. Thus voice ports can be reserved for particular users or groups of users (for example, those that use the particular coverage path).

The following features do not use coverage paths:

- Transfer into AUDIX with the feature access code or with the GOTO AUDIX button.
- Return Call doesn't use the coverage path. When a user is in message display mode, and the user is displaying the message that says `Call AUDIX`, the CALL-DISP button is used.

For a review of considerations related to AUDIX, see the planning documents:

- *AUDIX System Description*, (the external unit), 585-300-905.
- *Planning for DEFINITY AUDIX System*, (the circuit pack system), 585-300-904.

Interactions

The following features interact with the AUDIX Interface feature.

NOTE:
For further details about features that interact with an external AUDIX system, see the *AUDIX Feature Descriptions*, 585-305-203; for further details about features that interact with the DEFINITY AUDIX system, see the *DEFINITY AUDIX System Feature Descriptions*, 585-300-206.

- Abbreviated Dialing (For AUDIX only)

  The FAC for Transfer Into AUDIX may be programmed into an abbreviated dialing button.

- Attendant Conference

  An attendant that has split a call can conference the call with AUDIX by dialing the Transfer Into AUDIX access code. The attendant presses Release to drop out of the conference call.

- ACD (for external AUDIX only)

  A hunt group can be administered as an AUDIX ACD split. AUDIX traffic measurements are then available utilizing the ACD Call Management System. Login occurs when AUDIX signals the switch to make a voice port available for AUDIX service and logout occurs when AUDIX signals the switch to disable the port.

  The AUDIX adjunct and ACD CMS must be connected to the same switch. If the AUDIX in the DCS feature is active, a CMS located on a switch other than the host switch (AUDIX location) does not provide measurements for the AUDIX ports.

  Because AUDIX frequently takes voice ports in and out of service for maintenance testing, high login activity may be seen for the AUDIX split in measurement reports.

  On CMS reports that display an agent's login identifier, AUDIX voice ports always show a login identifier that is the same as the extension, even if login identifiers are not administered on the switch.

- Call Coverage

  When a coverage call successfully completes to AUDIX or is routed from a remote switch to the host switch because of coverage, the principal is dropped from the call (no temporary bridge appearance is maintained).

  Coverage calls from a remote switch that fail to reach AUDIX as a coverage point cannot be returned to the original coverage path on the remote switch.

  Call Transfer Out of AUDIX interacts with Call Coverage as listed in Table 3-11.

Table 3-11. Transfer Out of AUDIX (Enhanced) and Coverage Interactions

Source	Transfer Destination	Coverage Type
External	Local Station Remote Station (DCS) Remote Station (ISDN)	Internal External Internal
Internal (local)	Local Station Remote Station (DCS) Remote Station (ISDN)	Internal Internal Internal
Internal (remote)	Local Station Remote Station (DCS) Remote Station (ISDN)	Internal Internal Internal

- Call Forwarding

  An AUDIX user can forward calls to a remote AUDIX hunt group or to the host AUDIX hunt group.

  The system administrator must correctly administer the AUDIX destination for the remote AUDIX hunt group.

- Call Transfer

  A call transfer out of AUDIX can be to a UDP extension. If the destination extension is a UDP extension on a remote switch, the call is treated as a direct call. Additional trunks are used for calls transferred between DCS nodes.

  Calls may be transferred into AUDIX by users handling redirected calls for principals who are AUDIX subscribers.

- Class of Restriction (COR)

  A high FRL assigned in the COR allows callers to transfer to long-distance numbers. To prevent toll fraud, assign a low FRL to the AUDIX system ports.

- Facility Test Call

  Unauthorized calls may be placed using the facility test call access code. To prevent this fraud, remove the facility test call access code.

- DCS Leave Word Calling (external unit only)

  In a DCS network, the called party may be on a different switch than the calling party. If the DCS link is down, attempts to store LWC messages are denied and intercept tone is returned. Leave Word Cancel requests are always denied for principals with AUDIX LWC; in some instances, the request may appear to be activated when it actually is not (see Leave Word Calling).

- Leave Word Calling (LWC)

  For AUDIX, the system administrator has the option of indicating that a principal's LWC messages are kept by AUDIX. This means that an LWC message left for a principal causes the extension of the calling and called parties to be reported to AUDIX. The principal can retrieve the message by calling AUDIX. The principal cannot retrieve the message using other retrieval methods (station display, demand print, Message Center agent, or synthesized voice), but is notified of the existence of AUDIX messages via these methods.

  If the administrator assigns a principal's LWC to another messaging service, AUDIX can still report the existence of waiting LWC messages for the principal, but not the message content. This means that an LWC message left for a principal causes an indication of a waiting LWC message to be sent to AUDIX. The principal can retrieve the message using other retrieval methods (station display, demand print, Message Center agent, or synthesized voice). However, the principal is still notified of the existence of AUDIX messages.

  If the data link between the system and AUDIX is down, attempts to activate LWC for an AUDIX-covered principal are denied and the reorder tone is returned.

  If a caller attempts to cancel a LWC message sent to AUDIX, the caller receives an intercept tone if the called party is on the same switch. If the called party is on another switch in the DCS network, then the caller receives a confirmation tone as long as the DCS data link to the called party's switch is operational, *even though the message is not actually canceled*.

- Message Waiting Lamp (MWL) Activation/Deactivation

  The MWL interactions are the same whether the switch is a host switch or a remote switch. If a message is left for a principal on AUDIX, the switch lights the principal's MWL when AUDIX tells it there is an AUDIX message.

  If the principal retrieves the message, the switch extinguishes the AUDIX MWL only if the combined status of LWC, Message Center Service (MCS), and AUDIX indicate there are no more messages.

- PCOL

  A PCOL may not be covered by AUDIX.

- Queuing

  Queuing maybe utilized with the AUDIX hunt group. This allows callers to wait for an available AUDIX system port.

- Ringback Queuing

  On direct calls to the remote AUDIX, where all trunks to the host AUDIX are busy, a busy tone is returned. On coverage calls, if all trunks to the DCS host AUDIX are busy, AUDIX is treated as a busy coverage point. If there are coverage points after AUDIX, then the call terminates there; otherwise, the call remains at the principal. In summary, Ringback Queuing does not apply to AUDIX calls.

- Single-Digit Dialing and Mixed Station Numbering

  AUDIX is designed for use with a Uniform Dial Plan. It supports only one extension number length (three-, four-, or five-digit) that is used by AUDIX subscribers. Single-Digit and Mixed Station Numbering cannot be used. However, nothing prohibits connecting a switch to AUDIX that provides these features, as long as all AUDIX subscribers have the same extension number length.

- Temporary Bridged Appearance

  Stations that normally have a temporary bridged appearance with their coverage point do not if the coverage point is AUDIX.

- UCD (for DEFINITY AUDIX only)

  For DEFINITY AUDIX, a hunt group must be administered using a Uniform Call Distribution (UCD) split.

- Voice (Synthesized) Message Retrieval

  If your system has the TN725B Speech Synthesis circuit pack, and if LWC Activation is enabled, then messages are retrieved from two locations:

  — LWC messages sent by pressing the LWC button on a voice terminal are retrieved via Voice Message Synthesized Retrieval.

  — All other messages are retrieved via AUDIX.

  If your system does not have a TN725B Speech Synthesis circuit pack, and if LWC Activation is enabled, then LWC messages sent by pressing the LWC button on a voice terminal are not retrievable by a nondisplay set.

Administration

The AUDIX Switch Interface is administered by the System Manager. For more information, refer to:

- *Switch Administration Guide for AUDIX Voice Messaging*, 585-305-505,

  for further details on AUDIX switch administration when AUDIX is configured external to the switch
- *Switch Administration for DEFINITY AUDIX System R1.0*, 585-300-509, and *DEFINITY AUDIX System R1.0 Administration*, 585-300-507,

  for DEFINITY AUDIX
- *AUDIX System Description*, 585-305-201,

  for AUDIX

Hardware and Software Requirements

For DEFINITY AUDIX, you must have five contiguous slots in the switch carrier. See the DEFINITY AUDIX documentation for further details.

The data link for the external AUDIX (either the one- or two-cabinet configuration) varies depending on the type of switch.

- An AUDIX system connected to a DEFINITY Generic 3r switch uses the TN577 Packet Gateway (PGATE) circuit pack. This board provides X.25 connectivity to support external adjuncts such as AUDIX and DCS nodes.
- An AUDIX system connected to a Generic 3i switch uses the TN765 Processor Interface (PI) circuit pack. This board has one Electronics Industries Association (EIA) port which allows direct access to one of its four data links. The EIA port is the recommended interface for the system.

Refer to the *AUDIX System Description* manual, 585-305-201, for more information on AUDIX hardware and software requirements.

For DCS AUDIX, DCS software is required.

Authorization Codes

Feature Availability

This is an optional feature available with all G3 releases.

Description

Provides the means for extending control of system users' calling privileges and security for remote access callers.

The "Authorization Codes" feature is optional, is closely linked to the "Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs)" feature, and can be used with the "Automatic Route Selection (ARS)", "Automatic Alternate Routing (AAR)", and "Remote Access (with Security Measures)" features, as well as with incoming trunk calls.

Authorization codes may be used for any or all of the following reasons:

- To allow a calling user to override the FRL assigned to the originating station or trunk
- To restrict individual incoming tie trunks and remote access trunks from accessing an outgoing trunk
- To identify certain calls on CDR records for cost-allocation purposes
- To provide additional security control for the system

When an authorization code is dialed, the FRL assigned to the extension number, attendant console, incoming trunk group, or remote access trunk group being used for the call is replaced by the FRL assigned to the authorization code. The new FRL functions the same as the one it replaces; however, the new FRL may represent greater or lesser calling privileges than the FRL that it replaces. Access to any given facility depends on the restrictions associated with the authorization code FRL.

For example, a supervisor may be at a desk of another user and want to make a call that is not normally allowed by the FRL assigned to that extension. The supervisor, however, can still make the call by dialing an authorization code that has been assigned an FRL that is not restricted from making that type call.

For security reasons, authorization codes range from four to seven digits. The number of digits in the codes must be a fixed length for a particular switch.

Incoming trunk groups within a system may be administered to always require an authorization code. The system applies recall dial tone to a call when the user must dial an authorization code. If the user dials a correct authorization code within 10 seconds (inter-digit time-out), the call completes as dialed. If the user does not dial an authorization code or dials an incorrect authorization code, the call routes to the attendant, or routes to intercept tone, depending on system administration.

Normally, DID trunks should not require authorization codes. However, it can be done and care should be taken when administering DID trunks to require an authorization code, because different type calls could terminate at different endpoints, and requiring an authorization code could be confusing to the caller.

A Cancellation of Authorization Code Request (CACR) digit may be administered. The CACR digit cancels the 10-second interval between dialing. When the CACR digit is dialed, the call immediately routes according to system administration. (Incoming trunk calls receive intercept treatment or go to the attendant.) Other calls receive intercept treatment unless the user's FRL is high enough to route the call. A CACR digit from an off-premises extension over DID/Tie trunks use DID/Tie trunk intercept treatment. Internal calls receive intercept tone.

CAUTION:

Do not program passwords or authorization codes onto auto dial buttons. Display telephones display the programmed buttons, providing internal abusers access to the auto dial buttons to originate unauthorized calls. If passwords or authorization codes must be programmed onto auto dial buttons, use the ~s (suppress) character to prevent the codes from being displayed.

AAR and ARS Calls

Each authorization code is assigned a COR that contains an associated FRL. Within a system, access privileges are determined by the FRL assigned to the facility where the call is originated. When an AAR/ARS call is dialed, the system allows or denies the call based on the FRL of the originating station. COR is used to restrict internal or non-AAR/ARS calls.

Authorization codes are given to individual users and provide a method of specifying the level of calling privileges for that user regardless of the originating facility. Once an authorization code is required and dialed on an AAR/ARS call, the FRL assigned to the authorization code replaces the originating FRL and controls and defines the user's privileges.

An AAR or ARS call originated by a system user or routed over an incoming tie trunk may require a dialed authorization code to continue routing.

Extreme care should be taken when administering authorization codes, so that a user does not have to dial the authorization code more than once. For example, if a user makes an AAR or ARS call and the user's FRL is not high enough to access any of the trunks in the routing pattern, the system prompts the user for an authorization code. If the FRL assigned to the authorization code is high enough to access the next trunk group in the routing pattern, the user is not prompted to dial the code again. If the call is routed through another PBX, the user may be required to dial an authorization code again. This type of situation can be avoided through careful administration.

When an authorization code is required on some, but not all, trunk groups, the system prompts for an authorization code when the originating FRL is not adequate to access the next available trunk group in the routing pattern.

Remote Access Calls

When a remote access caller dials the assigned remote access number and establishes a connection to the system, the system may request the caller to dial an authorization code and/or a barrier code. The authorization code defines the caller's calling privileges within the system.

If entry of an authorization code is required, it applies to all remote access trunk groups in the system. If a remote access user must dial an authorization code to gain access to the system facilities, an authorization code is not requested again even if the user places a call that routes through the "Automatic Route Selection (ARS)" or "Automatic Alternate Routing (AAR)" feature.

The system may be administered for a Time-out to Attendant option. This option routes a remote access call to the attendant if the user fails to dial within 10 seconds after receiving the system request for an authorization code. Also, the remote access user can dial the CACR digit code, if administered, which cancels the 10-second time-out interval. In this case, the call routes immediately to the attendant. If an off-premises user on a DID/tie trunk cancels an Authorization Code, DID/tie intercept treatment is received.

Considerations

From a remote location, all authorization codes as well as all barrier codes (if required) are normally entered using touch-tone dialing. However, rotary dialing may be used in some cases, depending on where the authorization code is forced and how the trunks are administered. A user with a rotary dial telephone can also dial the LDN for access to the attendant or, after dialing the remote access number, wait 10 seconds for Time-out to Attendant. In either case, the attendant must extend the incoming call.

The "Authorization Codes" feature is entirely in addition to, and in no way limits, other methods of call control such as Toll Restriction, Miscellaneous Trunk Restriction, and Outward Restriction.

For security reasons, Authorization codes must be assigned randomly. This also makes it difficult for one user to guess the authorization code assigned to another user.

A CACR digit code, if administered, can be either the # symbol or the digit 1. The # symbol is used when the tandem and main switches are DEFINITY system

Generic 1 and Generic 3i. If a System 85, DIMENSION PBX, DEFINITY Generic 1 switch, or DEFINITY system Generic 3i switch is part of the network, then the digit 1 is used as the CACR digit code. If the digit one is used as the CACR digit code, then it cannot also be used as the first digit of an authorization code.

If the Time-out to Attendant option is not administered and if a user dials the CACR digit code instead of an authorization code, the system assumes that an invalid authorization code was dialed and routes the call to intercept tone.

Calling privileges are affected by the "Authorization Codes" feature as follows:

- For incoming trunk calls, where an authorization code is required due to administration on the ‘Trunk Group’ form, the authorization code does not change the privileges of the user in any way.
- For outgoing calls, where the FRL of the user is insufficient for accessing the routing pattern preference assigned by AAR/ARS, the authorization code changes the FRL of the user only. The FRL used is the one assigned to the COR that is associated with the authorization code entered. No other data assigned to that COR is assigned to the user.
- For remote access calls, where the user is required to enter an authorization code, the user is assigned the COR of the dialed authorization code, with all connected data, such as the FRL. This COR overrides the COR assigned to a barrier code, if a barrier code is also required.

Interactions

The following features interact with the "Authorization Codes" feature.

- AAR/ARS Partitioning

  Since PGNs are assigned according to COR and Authorization Codes can change a COR, PGNs can be changed on incoming Remote Access calls by the use of Authorization Codes. On originating calls, the user’s COR determines the PGN.

- COR and FRL

  When an internal system user dials an authorization code on an AAR/ARS call, the FRL associated with the authorization code overrides the FRL assigned to the system user.

  When a remote access user dials an authorization code, the associated COR determines the caller’s access privileges to the system’s features and services.

- Forced Entry of Account Codes and CDR

  On the 94A LSU (no longer supported) and 3B2 CDRU 18-word records, the authorization code is output only if the administered account code length is less than six digits in length. On the 59-character record, the authorization code is never recorded.

When an authorization code is required after the destination address is dialed, that code is recorded. Thus, all unauthorized attempts to dial an invalid authorization code is recorded, and a pattern of such calls can be traced using the CDR printouts.

Administration

The use of authorization codes is optional. However, if authorization codes are used, the following items must be administered by the System Manager:

- Authorization Code Parameters
  - — Enable the "Authorization Codes" feature
  - — Authorization code length — Can be from four to seven digits, and all authorization codes must be the same length
  - — CACR digit — Choice is the digit 1 or the # symbol
  - — Whether or not the Time-out to Attendant option is used
  - — The authorization codes themselves — This is a list of all authorization codes and their associated CORs. As many as 5,000 codes may be used. Authorization codes should be selected randomly and cannot begin with the digit 1 if the digit 1 is used as the CACR digit code.
- Remote Access
  - — Whether or not an authorization code is required on a remote access call
  - — Whether or not the system applies recall dial tone to request that an authorization code be dialed
- AAR and ARS
  - — If possible, assign COR FRLs and Routing Pattern FRLs so that no more than one authorization code is required when making an AAR/ARS call.
- Trunk Groups
  - — Whether or not each incoming or two-way trunk group requires an authorization code for incoming calls on that trunk group to complete to their destination.

Hardware and Software Requirements

No additional hardware is required.

Auto Start and Don’t Split

Feature Availability

This feature is available with G3i-Global, G3rV1, G3V2, and later releases, and is not available with G3vsV1/G3sV1, or G3iV1.

Description

The "Auto Start and Don’t Split" feature allows the attendant to initiate a phone call by depressing any button on the dial keypad without having to first press the Start Button, thus reducing the number of buttons that attendant console users need to use to handle calls. This is a system-wide feature. If the attendant is on an active call and presses digits on the keypad, the system automatically splits the call and begins dialing the next call. When the "Auto Start and Don’t Split" feature is enabled, the Start button is disabled and end-to-end signaling is not allowed. That is, digits pressed on the keypad are always interpreted and are not passed to another device, such as a voice mail service.

Auto Start is temporarily deactivated when the Don’t Split button is pressed and remains deactivated until the Don’t Split button is depressed again or the current call is terminated. This allows end-to-end signaling: When the Don’t split is pressed, digits pressed on the console keypad are heard by the parties on the call. For example, use Don’t split if you need to send touch-tones to the far end to pick up answering machine messages.

To extend an active call to another extension, begin dialing the digits of the other extension. The active call is automatically put on hold. Once the called party answers, press Release to extend the call.

To send touch-tones on an active call, press Don’t Split. The call remains active. Press the keypad digits; the tones are sent to the far end. To deactivate Don’t Split, press Cancel.

Considerations

When Auto Start is enabled and an attendant dials an AAR number where the min and max in the AAR analysis table are not equal, then the attendant must dial a ***“#”*** after the digit string or the call is not processed.

Interactions

Release, Forced Release, and Cancel deactivate the "Auto Start and Don’t Split" feature.

The "Auto Start and Don’t Split" feature is deactivated when the ‘Don’t Split’ feature is activated and reactivated upon deactivation of ‘Don’t Split’.

If the system is using the "CDR Account Code Dialing Forced Entry of Account Codes" feature, Auto Start and Don’t Split cannot be activated. If VIS (Visually Impaired Service) is activated or deactivated while Don’t Split was enabled, then Don’t Split is deactivated.

Administration

This feature is administered on a system-wide basis on the ‘System Parameters Features’ form by enabling the ‘‘Auto Start’’ field. The Don’t Split button is administered on each ‘Attendant’ form.

Hardware and Software Requirements

No special hardware or software is required for this feature.

Auto-Available Split (AAS)

Feature Availability

This feature is available with G3rV1, G3V2, and later releases.

Description

AAS provides a way for members of an ACD *split* (Automatic Call Distribution group members: usually found in call centers) to be in a continuously AUTO-IN work mode. Although not restricted to such, this feature is intended to be used for splits containing only nonhuman members (for example, recorders or Voice Response Units). Its principal value is in bringing ACD members back into AUTO-IN work mode after a system restart.

Considerations

AUDIX does not require the "Auto-Available Split (AAS)" feature since it uses BX.25 messages to automatically activate its ACD agent ports after a PBX restart. The "Auto-Available Split (AAS)" feature is primarily intended for nonBX.25 and nonASAI PBX adjuncts, which require extra help in getting their PBX ports back on-line after a restart.

Interactions

AAS is not intended for any agent port hardware that can effect a change of its work mode state since a request to move to any state other than AUTO-IN is denied; however, administration of such sets is not blocked.

ACD Splits

Standard operation for ACD Splits remains the same. The major difference is in the handling of work modes. For non-Auto-Available splits, when an agent logs in, they are immediately placed in the AUX-WORK mode and must generate a buttonpush (or FAC) requesting a change to the AUTO-IN or MANUAL-IN work mode in order to receive calls. For Auto-Available splits, the agent log in operation and work mode change requests are handled differently. An agent is automatically logged in under the following circumstances:

- CMS successfully completes an Agent Move request into an Auto-Available split.
- A previously maintenance-busied-out port (which is defined as an agent in an Auto-Available split) is released.
- The system (or a particular process) reinitializes and requires a relogging in of agents.

- Administrator administers a split as AAS = ***y*** (Hunt Group Screen).
- Administrator administers an agent into a previously defined AAS split.

Agents are automatically logged out under the following circumstances:

- The adjunct successfully completes an Agent Move request out of an Auto-Available split.
- The Auto-Available agent's port is unavailable due to a periodic or system technician-initiated maintenance busy request.
- Administrator administers a split as AAS = ***n***.
- Administrator removes an agent from a previously defined AAS split.

Once an agent is logged into an Auto-Available split, it is immediately moved to the AUTO-IN work mode. After an agent is logged into an Auto-Available split, subsequent requests to change out of the AUTO-IN work mode are denied (in other words, if appropriate, an intercept tone is returned to the user and/or the lamp associated with the work mode request receives denial flutter).

Auto-Answer

The 'Auto-Answer' feature was originally implemented for the benefit of human agents. Since AAS is intended for nonhuman agents, administering an Auto-Answer terminal as a member of an AAS is not recommended or supported; unexpected/undesirable feature interaction may result. Currently, if a nonanalog terminal is administered to be Auto-Answer and that terminal is logged into a split then when the terminal goes on-hook (or pulls the handset) then the terminal is logged out. For analog terminals administered to be auto-answer and the terminal is logged into a split, the agent must dial the ACD logout FAC in order to be logged out. If this terminal is a member of an AAS split, a logout FAC entry is denied. The agent is logged out by removing the terminal from the split when it is not active on a call or by busying-out the terminal. For an agent in an Auto-Available split, if that agent's terminal is also defined as Auto-Answer, then going off-hook logs the agent's station into any AAS of which it is a member. Going on-hook logs the agent's station out of any AAS of which it is a member. However, the agent must first push an administrable RELEASE button. However, the agent is first put in AUX-Work until the agent pushes the administered RELEASE button on nonanalog sets or generates a disconnect on analog sets. This delay in putting the agent in Auto-IN work mode allows the set to go off-hook to place a personal or emergency call prior to receiving any ACD calls that may be in queue.

Agent Logout

Any logout request initiated from an agent in an Auto-Available split is denied (that is, if appropriate, intercept tone is returned to the user and/or the light associated with the work mode request receives denial flutter).

Group Administration

See “Hunt Group Administration” in *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653. When ACD has been optioned for the system and the “ACD” field on the “Hunt Group” form has been set to ***y***, then a new “Auto-Available” field appears.

CMS Notifications

The "Auto-Available Split (AAS)" feature notifies CMS (Call Management System) of any login, logout, or change into the AUTO-IN work mode on a per agent basis. An AAS agent is identified to CMS with a LoginID equivalent to the agent’s administered extension in a non-EAS environment. With EAS, the AAS LoginID and port are assigned on the “LoginID” form.

Adjunct CMS Move Agent

Move agent requests initiated by an Adjunct CMS simulate the normal administrative requests to remove the member from one split then add the member to another split. As such, CMS requests are subject to the same end validation checks that are provided any time these administrative changes are issued via the G3-MT. However, it is possible to move (that is, remove then add) a member of an Auto-Available split while that member is LOGGED IN (subject to the end validation outlined in Hunt Group Administration).

Administration

Hunt Group Administration

When ACD has been optioned for the system and the “ACD” field on the "Hunt Group" form has been set to ***y***, then a new “AAS” field appears. For more information, see the feature description for Hunt Group Administration.

Hardware/Software Requirements

No special hardware is required to support this feature.

Automatic Alternate Routing (AAR)

Feature Availability

This optional feature is available with G3s/G3vs PBP, G3i, and G3r, and is not available with G3s/G3vs ABP.

Description

Provides alternative routing choices for private on-network calls. With Automatic Alternate Routing (AAR), the system automatically selects the most desirable (normally the least expensive) route over various trunking facilities for private network calls. AAR also provides digit modification to allow on-network calls to route through the public network when an on-network route is not available.

The private network of PBXs that utilizes the "Automatic Alternate Routing (AAR)" feature is called an ETN. An ETN is a hierarchical network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations. An ETN consists of tandem switches, the intertandem tie trunks that interconnect them, the access or bypass tie trunks from a tandem switch to a main switch, and the capability to control call routing over these facilities.

Within an ETN, each switching facility is identified by a unique private network office code. Private network office codes may be one to eight digits in length. Throughout the rest of this description, the private network office code will simply be referred to as the *office code*.

ETN addresses for DCS or UDP destinations are limited to a seven-digit format. This means that the location code part for UDP/DCS is a three-digit code of the ‘RNX’ form and the extension number is a four-digit number in the XXXX format (along with limitations that the UDP/DCS number cannot start with a 0). Note that 5-digit UDP extensions are supported. For other destinations, ETN addresses are not limited to the seven-digit RNX format.

The principal use of AAR is to provide routing of private network calls. Private network calls are those calls that originate and terminate at a customer location without accessing the public network. The normal scenario is as follows: The calling party dials the AAR access code followed by an on-network number. AAR then selects the route for the call and performs any necessary digit manipulation. AAR selects the most desirable route for the call. If the first choice route is not available, another route is chosen automatically.

To use AAR, the user dials the AAR access code and the called number. Feature operation is completely transparent to the user. The AAR access code is normally the digit 8. Normally, the called number is a private network number. However, it may also be a public network number, a service code, an IDDD number, an operator code 0 (or any other digit assigned to the operator), or a CDOS (Customer Dialed/Operator Serviced) number (0+ or 01+ the number).

Private network (on-network) numbers are handled by the "Automatic Alternate Routing (AAR)" feature. An on-network number can be changed into a public network direct-distance dialing number, a CDOS number, or an IDDD number by, administering the *ars* call-type for such numbers.

The private network location codes may match public network central office codes. Therefore, the only way to determine the intended network for seven-digit calls is by the dialed AAR or ARS access code or by specific administration of the *ars* call-type on the AAR Analysis' form.

AAR and Subnet Trunking provide a convenient means to place IDDD calls to a frequently called foreign city. Such calls route as far as possible over the private network before exiting the network. The office code is, of course, reserved to represent a particular country and city. At the final on-network switch, the office code is deleted. The international prefix code 011 (in the US, 00 in most of Europe, and so on), the country code, and the city code are inserted. The inserted digits plus the last four digits of the originally dialed number constitute the IDDD number. "Subnet Trunking", which also has ARS applications, is discussed elsewhere in this chapter.

Similar to the IDDD case, certain domestic calls may reach a point on the network where they can route no further because tie trunks to the next switch are busy or none are provided. In this case, the office code can be deleted and the appropriate public network code inserted. Calls of this type route off-network via a central office. The central office may be connected to either an ETN tandem or main switch. Toll charges, if any, are from the final ETN switch to the destination.

Each office code can point to any one of several Routing Patterns, numbered 1 through a maximum limit for your system. More than one office code can point to the same pattern. A blank pattern provides intercept treatment and pattern 254 (640 for G3r) is the default for all office codes. Routing Patterns are shared with ARS. Access to a route within the pattern is controlled by FRL assignments. "Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs)" are fully described elsewhere in this chapter. For outgoing ISDN calls, route selection is dependent on Bearer Capability Class (BCC), FRL, and type of facility.

The system may serve as an ETN tandem switch. This distinction as a tandem switch is important with respect to the routing of certain calls. As a tandem switch, the system can access or be accessed by Intertandem Tie Trunks to/from other tandem switches and/or Access Tie Trunks to/from ETN main switches. Traveling Class Marks (TCMs) are appended to numbers outpulsed on tandem trunks. (TCMs represent the originating user's FRL.) The system can also access Bypass Tie Trunks to an ETN main switch.

AAR Dialing

AAR begins when a user dials the AAR access code (normally the digit 8), followed by the number to be called.

As soon as the user dials the AAR access code, the system checks to see if the user's voice terminal extension has been Origination Restricted or Outward Restricted by its assigned COS. The system also checks to see if the user has a Controlled Restriction of either Outward or Total. If any of these restrictions apply, intercept treatment is applied to the call. Otherwise, the AAR call continues and the user can enter the number to be called.

A second dial tone may or may not be heard after the AAR access code is dialed, depending on the system administration.

Inter-Digit Time-out

The system uses a short inter-digit timer and a long inter-digit timer during the dialing process. Normally, a 10-second inter-digit timer is used between each digit for the user to continue dialing. If the digits dialed so far point to a valid destination, but there is a similar string of digits which is of different length, the short three-second inter-digit timer will be started. If dialing does not continue before the timer expires, it is assumed that no more digits will follow, and # is appended by the system to indicate end of dialing. To override the timer for faster call processing, the originator may dial # to indicate end of dialing.

A 10-second long inter-digit timer is used when the digits dialed so far are not a valid destination. But more digits may be required, and a 10-second timer is allowed for dialing the next digit. Time out of this timer results in Intercept tone to the caller if it is not a valid number. To indicate the end of dialing for valid strings with a lesser number of digits, the user may, however, dial # on any of these calls to cancel the time-out interval and indicate end of dialing.

When no length ambiguity exists and all digits are collected, the call is routed, and no timer or # is required.

Digit Conversion

Once the AAR access code and the called number are dialed, the dialed number is compared to entries in the "Matching Pattern" fields of the AAR Digit Conversion Table screen. If all or part of the dialed number matches one of the Matching Patterns on the screen, the dialed number is replaced by a new number from the "Replacement String" field on the screen. This new number is then used to route the call, the call becomes an ARS call, and is routed using the ARS Analysis Table. This function may be used to route specific dialed number strings to a different number, intercept, and so on. The Digit Conversion Table is only used once per call.

The purpose of AAR digit conversion is simply to convert private network numbers to public network numbers. This allows the system to change specific dialed digits to a public network number and route some calls via ARS. Also, unauthorized private network calls can be routed to an attendant or receive intercept treatment. In G3r, conversion can be to an extension, AAR, or ARS.

Time of Day Routing

The Time of Day Plan Number of the calling party is used to make the choice of an associated Time of Day Routing screen. After this, for G3s/G3vs and G3i, an AAR call passes through AAR digit conversion. For G3r, the opposite is true, as PGN chooses the pattern after the routing analysis is successful. On this screen, a Routing Plan Number (RPN), which is identical to the Partition Group Number, is identified based on the time of day. This plan is then used to select the AAR Analysis Table screen, discussed later in this feature description, which will determine how the call is routed.

If Time of Day Routing is not assigned, the user’s PGN is used to select the ARS Analysis screen.

See the "AAR/ARS Partitioning" and "Time of Day Routing" features for more information.

AAR Analysis

After an AAR call passes through AAR Digit Conversion and Time of Day Routing, AAR Analysis is performed based on the Time of Day Routing Plan Number or (if Time of Day Routing is not assigned) the user’s PGN.

The system uses AAR Analysis to compare the dialed number with entries in an AAR Analysis Table. When the system finds a dialed string entry in the table that matches the dialed number, the AAR Analysis Table maps the dialed number to a specific Routing Pattern (discussed later) and Call Type. The selected Routing Pattern and Call Type are then used to route the call. If the “Call Type” field on the AAR Analysis Table for a digit string is `ars`, call processing crosses over to ARS and the call is processed as an ARS call. The ‘AAR Analysis Table Screen’ form also shows the minimum and maximum number of digits required for digit analysis of each dialed number.

Dialed string entries may contain the letter ***x*** or ***X***, which is used as a wildcard character. This wildcard character matches any of the digits 0 through 9. For example, a dialed string entry of ***3x*** applies to all calls beginning with 30 through 39. This wildcard makes it possible for traditional three-digit RNXs to be represented in several ways in the AAR Analysis Table. For example, RNXs 200 through 299 can be assigned to the AAR Analysis Table in either of the following ways:

Dialed String	Min. # of Digits	Max. # of Digits
2	7	7
	or	
20	7	7
21	7	7
22	7	7
...	...	...
29	7	7
	or	
2xx	7	7
	or	
20x	7	7
21x	7	7
22x	7	7
...	...	...
29x	7	7

It is possible that some numbers may overlap other numbers. For example, the AAR Analysis Table may have dialed string entries of ***645*** and ***6452***. In this case, for example, the number 645-2045 will be routed according to the 6452 entry (the longest dialed string). In G3r, analysis can specify a node number, not a pattern (the pattern is obtained from the node number routing table).

When the UDP is used, the three-digit RNX dial string representation must be used on the AAR Analysis Table to match the administration in the UDP table.

Possible Call Types in the AAR Analysis Table are as follows:

- Regular AAR call (the only one for G3r)
- Crossover to ARS call (administered as a digit conversion in G3r)
- Attendant (indicates that the call will be terminated to remote attendant) (G3i only).

- Home ETN address (indicates that the call should be terminated locally on the home switch instead of routing to another ETN node. If the UDP is administered, then that dial plan is used to convert the number to a local extension number. Otherwise, the location code is deleted from the dialed number, and the remaining digits are used to route the call to a local extension). This is administered as a digit conversion in G3r.
- AAR calls that should be sent to the local attendant group. This is administered as a digit conversion in G3r.

Some special dialing patterns are automatically mapped to a specific Routing Pattern and Call Type. These special dialing patterns are preassigned and require no administration. However, they can be changed by the system administrator. See the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for a list of these default translations.

Normally, the “Route Pat” (routing pattern) field on the Analysis Table screen contains a routing pattern number. However, this field may instead contain a Remote Home Numbering Plan Area (RHNPA) table number. When an AAR Analysis table points to an RHNPA table, the next three dialed digits are compared with codes in the selected RHNPA table. Each code on the table is then mapped to a specific routing pattern number.

In summary, AAR Digit Analysis is merely a method of selecting a routing pattern. The routing pattern may be selected in several ways:

- It may be selected directly from the AAR Analysis table.
- The AAR Analysis table may first have to select an RHNPA table which will in turn select the routing pattern.
- For G3r only, it may be selected from the node number routing table.

Routing Patterns

The digit translations performed on an AAR call by the AAR Analysis and RHNPA tables cause a specific Routing Pattern to be selected for the call. More than one combination of dialed digits can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with AAR, digit translation should always point to a Routing Pattern. If calls to some numbers are to be denied, this should be handled by FRL assignment, not by intercept on the codes. "Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs)" are discussed elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of trunk groups that can be used for the call. Trunk group access is controlled by FRLs. The digit manipulation necessary to route the call is controlled by the "Subnet Trunking" feature. See the "Subnet Trunking" feature. Otherwise, the digit string to be outpulsed is as dialed by the user or as converted by AAR Digit Conversion.

Considerations

AAR provides efficient use of private network facilities.

Routing patterns and RHNPA tables are shared between partitions.

If a customer changes AAR routing assignments, it is the customer's responsibility to notify the RSC network designer and the SCO technician of the changes in order to receive their continued support.

If a system is the last ETN tandem switch for a main ETN switch that has no tie trunks, but has DID trunks, then digit deletion/insertion can be used to route calls to the ETN main switch.

Internal memory resources used for AAR Digit Analysis are shared by "Automatic Route Selection (ARS)", "Automatic Alternate Routing (AAR)", Digit Conversion, and Toll Analysis features. A "Percent Full" field on the ARS and AAR Digit Analysis screens indicates how many of these resources have been used.

Interactions

The following features interact with the "Automatic Alternate Routing (AAR)" feature.

- ARS

  ARS and AAR can access the same trunk groups and share the same Routing Patterns and RHNPAs. Also, AAR calls can be administered to cross over to ARS via digit analysis and digit conversion (or, for G3r, ARS to AAR).

- Abbreviated Dialing

  FRL checking is bypassed on an AAR call made via a privileged Abbreviated Dialing Group List.

- Attendant Control of Trunk Group Access

  Attendant control of a trunk group, in effect, removes a trunk group from the Routing Pattern. A controlled trunk group is never accessed by AAR.

- Authorization Codes

  An AAR or ARS call originated by a system user or routed over an incoming tie trunk may require a dialed authorization code to continue routing. If authorization codes are required for an incoming trunk call, then an authorization code must be dialed even if the originating FRL was adequate to complete the call. The authorization code's FRL is used in preference selection.

- CAS

  A CAS Attendant may extend a call out of a Branch PBX by use of AAR. The call is extended over an RLT by dialing the appropriate feature access code and number. The call is routed as determined by AAR administration at the Branch PBX.

- Toll Restriction

  Toll Restriction is not checked on AAR calls, unless they cross over to ARS or digit conversion occurs.

- Controlled Restriction, Origination Restriction, and Outward Restriction

  These features prohibit access to the "Automatic Alternate Routing (AAR)" feature.

- Miscellaneous Trunk Restrictions

  Miscellaneous Restrictions are not checked on AAR calls.

- Ringback Queuing

  Ringback Queuing can be used on AAR calls originated at the switch that provides the queuing. Incoming tie trunk calls will not queue on an outgoing trunk group.

  If a multi-appearance voice terminal user has an Automatic Callback button, makes an AAR call, and all trunks are busy, Ringback Queuing is activated automatically.

- CDR Account Codes

  An CDR Account Code may be required for an AAR call if it crosses over and becomes an ARS call via AAR Analysis or AAR Digit Conversion.

- CDR

  An AAR call using a trunk group marked for CDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number.

  Subnet Trunking does not affect CDR. The dialed digits are recorded, not the outpulsed digits.

  The originating FRL associated with the call is recorded. However, if 15-digit CDR account codes are used, the FRL value is overwritten.

  If CDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group. If an CDR account code is to be dialed with an AAR call, it must be dialed before the AAR access code is dialed.

- Voice Terminal Display

  The voice terminal display shows the dialed digits (not outpulsed digits). The called-party shown on the display is that of the trunk group actually used. The “Miscellaneous Call Identification” field displays the Trunk Access Code (tac) associated with the trunk used for the outgoing call.

An ISDN-BRI station may format these display fields differently and the timing of display updates may be different.

- UDP

  When a UDP number is dialed (four or five digits), the routing software initially converts the dialed number to a seven-digit format. The location code in the UDP table equates to a three-digit RNX plus four digits. In a four-digit UDP, the number created is the RNX plus the four extension digits originally dialed. In a five-digit UDP, the number created is the RNX plus the last four extension digits dialed.

  UDP destinations are limited to a seven-digit format. This means that the location code part for UDP/DCS is a three-digit code of the 'RNX' form and the extension number is a 4-digit number in the XXXX format (along with limitations that the UDP/DCS number cannot start with a zero).

  When the UDP is used, the three-digit RNX dial string representation must be used on the AAR Analysis Table to match the administration in the UDP table.

Administration

AAR is initially assigned on a per-system basis by an AT&T service technician. After the feature is activated, the following items are administered by either the System Manager or the service technician:

- AAR Access Code (one to three digits)
- AAR Analysis Table (one per PGN)
- AAR digit conversion table Up to 32 RHNPA Tables
- Routing Patterns
- FRLs — Assigned via Class of Restriction to each originating facility, authorization codes, and barrier codes
- Trunk Groups to be used with AAR
- Whether or not the system returns dial tone after the AAR FAC is dialed on trunk calls

Hardware and Software Requirements

AAR may require additional tie trunks. These additions are, however, cost effective when compared to the other alternatives for call routing.

AAR is provided as a part of the optional Private Networking software.

Optional ARS/AAR Digit Conversion software is required in G3s/G3vs and G3i. This can be established in the system when ARS and either UDP or Private Networking are ordered. Note that this is not optional for G3r.

Automatic Call Distribution (ACD)

Feature Availability

This optional feature is available with all G3 releases.

Description

Provides automatic connection of incoming calls to specific splits (hunt groups). Calls to a specific split are automatically distributed among the agents (hunt group members) assigned to that split. ACD data, transmitted from the switch to the CMS or BCMS, is used to generate various reports on the status of ACD agents, splits, and trunks.

An ACD split is simply a hunt group that is designed for use wherever a high volume of similar calls are received. ACD split call handling features can be measured by BCMS/CMS.

Members of a split are called agents. An agent can be a voice terminal extension or individual attendant extension. A voice terminal or individual attendant can be an agent in one or more splits. However, at any one time, an agent can be logged into a maximum of three or four ACD splits. (Non-EAS agents can only be logged into one split if that split is administered for Multiple Call Handling.)

In addition to the agents, a split supervisor can be assigned to each split. The split supervisor can listen in on agent calls, monitor the split queue status (discussed later) via queue warning buttons (see "Queue Status Indications" feature) and can assist agents on ACD calls. Although split supervisors can assist agents on ACD calls, the supervisors themselves do not normally receive ACD calls unless they are also members of the split. The request for assistance comes from the agents. An agent can request supervisory assistance by pressing an Assist button or by dialing the assist feature access code and the split number.

Call Distribution

An ACD split can use one of three methods to select an idle available line appearance, terminal, or console. The three distribution methods are described below.

Direct Department Calling

If a split is administered for direct hunting (DDC), an incoming call rings the first available extension number in the administered sequence. If the first split agent in the sequence is not available, the call routes to the next split agent with all call appearances idle, and so on. In other words, incoming calls always try to complete at the first split agent in the administered sequence. Therefore, the calls are not evenly distributed among the split agents.

Uniform Call Distribution

Uniform Call Distribution (UCD) uses the Most-Idle Agent (MIA) algorithm to route calls. The MIA algorithm creates a queue of agents who are available to receive calls. An incoming call is routed to the agent who has waited the longest time since completing an ACD call for that split. An agent who receives a call is placed on the bottom of the queue for that split. However, the agent remains in the MIA queue for any other splits/skills they are logged into. UCD ensures that calls to a split/skill are distributed evenly among agents logged into that split.

Expert Agent Distribution

Expert Agent Distribution (EAD) is only available with EAS. EAD also uses the MIA algorithm to route calls. The only difference is that EAD maintains a queue of idle agents by primary and secondary skill type. When agents with primary skills become idle, they enter the MIA queue in front of secondary agents. When calls are routed to an EAD skill, the call goes to the most-idle primary agent. If no primary agent is available, the call is routed to a secondary agent.

Split Queuing and Announcements

The ACD feature can be enhanced via the "Call Prompting" feature and/or "Call Vectoring" feature (see the appropriate feature sections). This information on Queuing announcements and priority applies to ACD without Call Vectoring or Call Prompting.

A queue can be established for an ACD split. When all agents are active on a call or in After Call Work, the queue allows incoming calls to wait for an idle terminal. If an agent becomes available while an incoming call is in the split queue, the call is automatically connected to the available agent.

NOTE:
Multiple Call Handling allows an agent to receive an ACD call while other calls are active or held at the agent's station. See the "Multiple Call Handling" feature for more information.

For non-vector-controlled splits, calls will not queue to a split that has no agents logged in, all logged-in agents in the Aux Work mode, or no queue slots available. A busy signal is returned to the caller unless a call has come in via an automatic-in central office facility, in which case the caller hears ringback from the CO, and the system continues trying to place the call in queue.

Two announcements can be assigned to each split. The second announcement can be administered so that it will repeat itself.

When an incoming call is directed to an ACD split, the call, depending on the administration of the split, will either try to access a split agent or will automatically be connected to the first announcement (Forced First Announcement), if available.

Forced First Announcement

The first announcement delay interval (0 to 99 seconds) indicates how long a call will remain in queue before the call is connected to the first announcement. If this interval is set to 0 seconds, the incoming call will automatically be connected to the first announcement, if available. The result is a forced first announcement, and the call will not attempt to access an agent until after the first announcement is heard.

NOTE:
If a call forwards from a dummy split to a destination that has forced first announcement administered, the caller will hear the first announcement if it is not forced. The caller will not hear the first announcement if it is forced. The caller will hear a forced first announcement on the dummy split. Also, if a call reaches a split via the "Call Coverage" feature, the first announcement (forced or delay) will not be provided to the covered call.

When a forced first announcement is assigned, the system tries to connect the incoming call to the first announcement, with the results being one of the following:

- If the first announcement is available, the caller receives audible ringing followed by the first announcement. The system then tries to connect the call to an agent.
- If the announcement is busy and has no queue, the system will wait 10 seconds and then try to access the announcement again.
- If the announcement is busy and has a queue, one of the following happens:
  - If the queue is full, the system will wait 10 seconds and then try to access the announcement again.
  - If the queue is not full, the call enters the announcement queue and the caller receives audible ringing until the first announcement is heard. The system then tries to connect the call to an agent.
- If the announcement is not busy, but is still unavailable (it might have been deleted), then the system tries to connect the call to an agent.

After a forced first announcement, the caller always hears ringback until the call is answered or until it is connected to a second delay announcement. After a first or second delay announcement, the caller hears music on hold (if administered).

Entering the Queue

When a forced first announcement is not assigned, the system will try to connect an incoming call to an available agent. If an agent is available, the call is connected to the agent. If all agents in the split are active (either on an ACD call or in ACW mode), the call enters the split queue. If there is no queue assigned, or if no queue slots are available and the incoming facility is a CO trunk, the caller will hear ringing. The system will continue trying to put the call in queue until a queue slot becomes available, or until the call is abandoned. If a split queue is not assigned, if the queue is full, if there are no agents logged in, or if all the logged in agents are in AUX, and the incoming facility is a digit-oriented facility (digits are being sent to the PBX as in DID, incoming wink, or immediate tie trunks), the caller receives busy tone or the call is redirected by the "Intraflow and Interflow" feature (discussed later) associated with the "Call Coverage" and "Call Forwarding All Calls" features.

NOTE:
Central office trunk (non-DID) calls receive ringback from the central office, so the PBX cannot provide a busy signal to these callers. The system will keep trying to put such calls into queue until successful or until the caller abandons.

First Announcement

After a call enters a split queue, the caller receives audible ringing and the first announcement delay interval begins. (If there is no first announcement, the second announcement delay interval begins. If there is no second announcement, the call remains in queue until answered or removed from the queue.) If an agent becomes available during the first announcement delay interval, the call is connected to the available agent. Otherwise, the first announcement delay interval expires and the system tries to connect the incoming call to the first announcement, with the result being one of the following:

- If the first announcement is available, the caller receives audible ringing followed by the first announcement.
- If the announcement is busy and has no queue, the caller receives audible ringing and the first announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
- If the announcement is busy and has a queue, one of the following happens:
  - — If the queue is full, the caller receives audible ringing and the first announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
  - — If the queue is not full, the call enters the announcement queue and the caller receives audible ringing until the first announcement is heard. The system then tries to connect the call to an agent.

- If the announcement is not busy, but is still unavailable (it might have been deleted), the second announcement delay interval begins and the system attempts to connect the call to the second announcement. If there is no second announcement, the call will remain in queue until answered or removed from the queue.

  After the first delay announcement, the caller will hear music on hold (if administered).

Second Announcement

After the first announcement has completed, the second announcement delay interval begins and the caller hears music (only if the first announcement is not a forced first announcement, in which case the caller hears ringing), if provided. (If there is no second announcement, the call remains in queue until answered or removed from the queue.) If an agent becomes available during the second announcement delay interval, the call is connected to the available agent. Otherwise, the second announcement delay interval expires and the system tries to connect the incoming call to the second announcement, with the result being one of the following:

- If the second announcement is available, the caller receives audible ringing or music followed by the second announcement.
- If the announcement is busy and has no queue, the caller receives audible ringing and the second announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
- If the announcement is busy and has a queue, one of the following happens:
  - If the queue is full, the caller receives audible ringing (only if the first announcement has not been heard) and the second announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
  - If the queue is not full, the call enters the announcement queue and the caller receives audible ringing (only if the first announcement has not been heard) until the second announcement is heard. The system then tries to connect the call to an agent.
- If the announcement is not busy, but is still unavailable (it might have been deleted), the call will remain in queue until answered or removed from the queue.

After the second announcement is heard, the caller hears music (if provided) or silence (if music is not provided), and one of the following occurs:

- If the split has been administered so that the second announcement is repeated, the system will attempt to connect the call to the second announcement after the delay expires.

- If the split has been administered so that the second announcement is not repeated, the call will remain in queue until answered or removed from the queue.

Forced Disconnect

At times, it may be desired to connect an incoming call directly to an announcement and then disconnect the call after the announcement has completed. This can be accomplished two ways:

- The incoming destination can be administered as an announcement extension. This way the calling party will hear the announcement and be disconnected. Also, the call is never queued for a split because it goes directly to the announcement.
- An announcement extension can be administered as a point in a split's coverage path. This way, calls that have been in the queue for a long period of time are forced to go directly to the announcement and are then disconnected.

Announcement Rules

The following rules govern which announcements a caller hears:

1. Calls coming directly into a split will always hear a forced first announcement if assigned, regardless of subsequent treatment such as Call Coverage, Call Forwarding, Night Service, or busy signal. If these calls queue for a sufficient period of time, they will hear delay first and second announcements.
2. Calls that reach a split via Call Coverage from another split or a station will NOT receive a forced first or delay first announcement at the destination split. They will hear a delay second announcement if administered and if the delay interval is met. The assumption is the caller has heard a first announcement at the original destination or that a call redirected from a station should not receive the first announcement, if administered.
3. Calls that reach a split via Call Forwarding from another split or a station WILL receive delay first and second announcements at the destination split if administered and the delay interval is met. These calls will receive a forced first announcement at the original split (if administered) but will NOT receive a forced first announcement at the ''forwarded-to'' split.

Intraflow and Interflow

The "Intraflow and Interflow" feature allows splits to be redirected to other destinations on the system. This is accomplished via the "Call Forwarding All Calls" or "Call Coverage" features. CMS receives a track count on forwarded calls that intraflow via Call Coverage. CMS is notified only if the call goes into the queue before redirecting. Splits can be assigned coverage paths. Also, a split can be a part of a coverage path. Thus, the "Call Coverage" feature can be used to redirect ACD calls from one split to another split according to the coverage path's redirection criteria. For instance, a split's coverage path can be administered so that incoming ACD calls are automatically redirected to another split during busy or unanswered conditions.

If Intraflow via Call Coverage is provided, the Coverage Don't Answer Interval (1 to 99 ringing cycles) associated with Call Coverage may begin when the call enters the split queue. The Coverage Don't Answer Interval does not begin until after the forced first announcement completes (if assigned). If the Coverage Don't Answer Interval expires before either of the two announcement delay intervals expires, the call is redirected to coverage. If no coverage point is available to handle the call, the call remains in queue and may then be connected to a delay announcement. If either of the announcement delay intervals expires before the Coverage Don't Answer Interval, the call is connected to a recorded announcement, if available, but the call will still go to coverage after the announcement.

The "Intraflow and Interflow" feature allows ACD calls to be redirected from one split to a split on another switch. This is accomplished by forwarding calls to an off-premises location via the "Call Forwarding All Calls" feature.

For a detailed description of the "Call Forwarding All Calls" feature and the "Intraflow and Interflow" feature, consult the appropriate feature section.

Queue Status Indications

The system provides queue status indications for ACD calls based on the number of calls in queue and time in queue. These indications are provided via lamps assigned to the terminals or consoles of split agents or supervisors. In addition, an auxiliary warning lamp can be provided to track queue status based on time in queue and another for number of calls in queue. Also, display-equipped voice terminals and consoles can display the time in queue of a split's oldest call and the number of calls in that split's queue. For more detailed information, consult the "Queue Status Indications" feature.

Priority Queuing

Priority Queuing allows calls with increased priority to be queued ahead of calls with normal priority. Priority Queuing can be provided two ways:

- A calling party's COR can be assigned Priority Queuing.
- An ACD split can be assigned Priority on Intraflow. This allows calls from the split, when intraflowed (via the "Call Coverage" feature, but not via the "Call Forwarding All Calls" feature) into another split, to be queued ahead of nonpriority calls already queued in the other split.

Agent Call Handling

"Agent Call Handling" is a separate feature that includes the various agent functions and operations. For details, see the "Agent Call Handling" feature description in this chapter. The following is a brief summary of the Agent Call

Handling functions and operations. The information below applies generally to traditional ACD. See the "Expert Agent Selection (EAS)" feature for additional EAS procedures.

NOTE:
Parts of this section describe when agents are available to receive ACD calls. Multiple Call Handling modifies the situations under which an agent can receive an ACD call. See the "Multiple Call Handling" feature for more information.

- Agent Log-in and Log-out — An agent is required to log in before he or she is able to receive ACD calls. The agent may or may not be required to enter a personal identification number, depending on administration. An agent performs the following steps to log in:
  1. Dial the login Feature Access Code (FAC).
  2. Enter the 2-digit (3-digit for G3r) split ID number.
  3. Enter the agent ID if IDs are administered.

  In addition, an agent can log out to let the system know that he or she is unavailable for ACD calls. To log out, the agent performs the following steps:
  1. Dial the logout FAC.
  2. Dial the two-digit or three-digit (G3r) split ID number.
- ACD Call Work Modes
  - Auxiliary Work — An agent can enter the Auxiliary Work mode when he or she is doing non-ACD activities such as taking a break or going to lunch but wants the relevant time tracked by BCMS/CMS. This makes the agent unavailable for ACD calls for that split. Entering the AUX work mode in one split does not affect the agent's status in other splits. The agent is not in the Most Idle Agent queue while in AUX work mode.
  - After Call Work — An agent can enter the ACW mode to perform ACD-related activities when needed. For example, an agent may need to fill out a form as a result of an ACD call. The agent can enter the ACW mode to fill out the form. The agent is unavailable for ACD calls from any split while in the ACW mode (the agent is placed in the AUX work mode for other splits). The agent is in the Most Idle Agent queue, but he or she is unavailable while in ACW.
  - Auto-In or Manual-In — An agent can enter either the Auto-In mode or the Manual-In mode to become available for ACD calls.

    When an agent enters the Auto-In mode, he or she, upon disconnecting from an ACD call, automatically becomes available for answering new ACD calls.

When an agent enters the Manual-In mode, he or she, upon disconnecting from an ACD call, enters the After Call Work mode for that split, and is not available for ACD calls. The agent must then manually reenter either the Auto-In mode or Manual-In mode to become available for ACD calls.

An agent may be required to enter a Stroke Count or Call Work Code when in the Manual Mode. For details on this interaction, see the ''Forced Entry of Stroke Counts and Call Work Codes'' later in this ACD description.

- Agent Answering Options
  - Automatic Answer All Calls — An agent with Automatic Answer All Calls will be connected directly to an incoming call if the station is idle.

    If an ACD split/skill call or direct agent call is directed to an idle agent with Automatic Answer All Calls then the agent will hear Zip Tone in the headset or handset and be connected directly to the call with no audible ringing. Agents in this situation must be aware that Zip Tone is the only signal they will receive when they are connected to this call. In this context the agent is idle if there are no calls on the station.

    If a non-ACD call is directed to an idle agent with Automatic Answer All Calls then the agent terminal will receive one cycle of audible ringing. At the same time the agent will hear Incoming Call Identification Tone in the headset or handset and be connected directly to the call. In this context the agent is idle if there are no calls on the station or all calls are on hold.
  - Automatic Answer ACD Calls Only — An agent with Automatic Answer ACD Calls Only will be connected directly to an incoming ACD split/skill call or direct agent call if the station is idle. In this context the agent is idle if there are no calls on the station.

    Non-ACD calls directed to an Multi-Appearance Voice Terminals with Automatic Answer ACD Calls Only will ring until answered, redirected or abandoned. The ringing treatment applied to these calls depends on the Voice Terminal Alerting Option administered for this agent.

    If an Analog Voice Terminal with Automatic Answer ACD Calls Only is off-hook and idle, then only ACD split/skill calls and direct agent calls will be directed to this agent. Non-ACD calls to this agent will receive busy treatment. However, if this analog agent is active on an ACD split/skill call or direct agent call and a non-ACD call is directed to the agent, the agent will hear Call Waiting Tone if appropriate. The agent may then answer the waiting call by going on-hook or flashing the switch-hook and dialing the Hold/Unhold Feature Access Code.

- Automatic Answer No Calls — Automatic Answer No Calls is also known as Manual Answer. When a call is directed to an agent the agent hears ringing, and then goes off-hook to answer the incoming call. The ringing treatment applied to the agent terminal is determined by the Voice Terminal Alerting Option administered for the agent.
- Headset Use Recommended — It is recommended this feature be used with a headset. In this case, the agent hears zip tone through the headset and is then automatically connected to the call. (If the incoming trunk group is data restricted, the zip tone is not heard. If the agent's extension is data restricted, zip tone is not heard. Therefore, trunk groups terminating to auto answer positions and auto answer agent positions should not be assigned data restriction.)

  Although not recommended, the automatic answering option can also be used with a handset or speakerphone. The feature works the same as with a headset, except the agent must be off-hook in order to receive the call. Zip tone, in this case, is heard through the handset or speakerphone.

  Please see the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for more information about administration.

■ Multiple Call Handling (MCH) — Multiple Call Handling allows an agent to be interrupted with an additional ACD call either after putting a call on hold, or when the agent is active on another ACD call. With MCH, the agent can either receive additional calls only when he or she requests one (G3V3 and later releases). Or, an additional call will automatically alert by ringing at the agent's terminal (G3V4 and later releases). Calls can be received up to the limit of non-restricted line appearances on the terminal.

■ Voice Terminal Alerting Options — The ringing treatment applied to an agent's voice terminal is specified by the Voice Terminal Alerting Option administered for the agent. This option is used to determine whether the alerting call receives: no audible ringing, a single ring cycle or continuous ringing. For details, see the "Voice Terminal Alerting Options" feature description in this chapter.

■ Agent Request for Supervisor Assistance — Agents can request assistance (whether on an active ACD call or not) from the split supervisor by using the Assist button or the assist feature access code.

■ ACD Call Disconnecting — An agent can be disconnected from an ACD call in either of three ways. The agent can press a Release or Drop button, the call can be dropped by the calling party, or the agent without the automatic answering option can go on-hook. If the agent presses the Drop button, he or she will receive dial tone and be unavailable for calls. The Drop button is not recommended for disconnecting calls.

CMS

CMS is an optional adjunct to the system that collects and processes ACD data. CMS uses this data to generate various reports on the status of agents, measured splits, measured trunks and, measured VDNs and vectors. These reports can be stored for later use or can be displayed on a terminal for real-time information.

For information on CMS for G3r and G3i, see *G3 Call Management System Administration*, 585-215-511. For information on CMS for G3i, see the *3B Call Management System Administration*, 585-215-504.

BCMS

The "Basic Call Management System (BCMS)" feature provides real-time and historical reports that assist a customer in managing individual agents, ACD splits (hunt groups), trunk groups and VDNs. These reports, provided by the system, are a subset of those available on the CMS adjunct. BCMS reports can be accessed and displayed on the Management Terminal, or printed on demand on the printer associated with the Manager I terminal or G3 Management Terminal. In addition, the historical reports can be scheduled to print on the system printer. For a detailed description, see the "Basic Call Management System (BCMS)" feature.

Abandoned Call Search

The "Abandoned Call Search" feature (G3V3 and later releases only) is used to identify abandoned calls on ground start, CO, FX, and WATS trunks. When the calling party on an ACD call abandons (drops) the call while waiting to be connected to an agent, the call is not connected to the agent, and the call is reported to the BCMS or CMS as being abandoned. For a detailed description, see the "Abandoned Call Search" feature.

Service Observing

Split supervisors (or other specified users) can use the "Service Observing" feature to train new agents and to observe in-progress ACD calls. The supervisor can observe in either a listen-only or a listen/talk mode. An optional warning tone can be administered to let the agents know that someone is observing the call. For more details on this feature, see the "Service Observing" feature.

NOTE:
The use of "Service Observing" feature may be subject to federal, state, or local laws, rules, or regulations and may be prohibited pursuant to the laws, rules, or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable law, rules, and regulations before using these features.

Direct Agent Calling

Direct Agent Calling causes a call to a particular ACD agent to be treated as an ACD call. Direct agent calling requires ASAI or EAS. Whether or not ASAI is turned on, Direct Agent Calling must be invoked by a calling endpoint with Direct Agent in its COR calling a station with Direct Agent in its COR.

Agents may receive zip tone, ring, or ring/ping when these calls are delivered. Agents may have ‘‘After Call Work’’ associated with these calls. The CMS and the BCMS correctly measure these calls as ACD calls.

Delivery of Direct Agent Calls

If the agent receiving the direct agent call is available to answer an ACD call in the associated split, the direct agent call is delivered to the agent. Zip tone is applied if the agent is in the automatic answer mode.

If the receiving agent is not available to answer an ACD call, (for example, the agent is busy on a call, in the After Call Work mode, or in the Auxiliary Work Mode), the receiving agent is notified with a ring-ping if the agent has a multifunction voice terminal or is on-hook. If the receiving agent has a single line voice terminal and is not available, the receiving agent will hear call waiting tone (even when the "Call Waiting Termination" feature is not assigned) if the agent is off-hook. The ring-ping or call waiting tone is given only once per call when the call is queued. The currently lit work mode button lamp for the associated split on the receiving agent’s voice terminal flash, indicates a direct agent call is waiting. Flashing starts when the call queues and stops when all direct agent calls leave the queue (answered, abandoned, or sent to coverage).

Direct agent calls are queued and served in a first-in first-out order before any non-Direct Agent Call (including priority calls). Therefore, when an agent becomes available, the switch first checks for any direct agent calls before serving normal ACD calls in queue.

Direct agent calls follow the receiving agent’s coverage and call forwarding, if activated. Once the call goes to coverage or is forwarded, the call is no longer treated as a direct agent call. CMS is informed that the call has been forwarded.

Answering a Direct Agent Call

The receiving agent answers a direct agent call by becoming available in the split with which the direct agent call is associated. While on a direct agent call, the agent becomes unavailable to subsequent ACD calls.

NOTE:
For initialized ASAI or user dialed EAS, both originating and called party CORs, need to be set for direct agent dialing.

If the receiving agent logs off by unplugging the headset, the agent may still answer a direct agent call in queue by logging back in and becoming available. Agents who have direct agent calls waiting will be denied if they attempt to log-off using a feature access code. If the agent is in the MAN-IN mode or has pushed the After Call Work (ACW) button while on a Direct Agent Call, the agent will go to ACW.

Vector-Controlled Splits

For detailed information on vector-controlled splits, see ‘‘ACD Split/Hunt Group Operation with Call Vectoring’’ in the "Call Vectoring" feature. Vector controlled splits/skills (that is, ACD hunt group with vector = ***y***) can be called directly via the split/skill extension (instead of calling a VDN mapped to a vector that will terminate the call to a vector controlled split/skill); however, the calls will not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.

Agent Sizing

Agent Sizing provides a maximum limit of logged-in ACD agents. For more information on Agent Sizing, see ‘‘Agent Sizing’’ in the "Agent Call Handling" feature.

Stroke Counts

Stroke Counts provide ACD agents with the ability to record customer-defined events on a per-call basis when the CMS is active. For details on the Stroke Counts function, see the "Agent Call Handling" feature description elsewhere in this manual.

Call Work Codes

Call Work Codes allow ACD agents to enter up to 16 digits for an ACD call to record the occurrence of customer-defined events (such as account codes, social security numbers, or phone numbers). For details on the Call Work Codes function, see the "Agent Call Handling" feature description elsewhere in this manual.

Forced Entry of Stroke Counts and Call Work Codes

An agent is always allowed to enter a Stroke Count and/or Call Work Code for an ACD call. However, each split can be administered so agents in that split are forced to complete a Stroke Count and/or a Call Work Code entry before becoming available for another call using Manual-In mode. The rest of this discussion assumes this has been done.

An agent can enter the Stroke Count and/or Call Work Code while on the call, or while in the ACW mode after the call releases. After a call has been released by an agent in the Manual-In mode, the agent automatically enters the ACW mode. The agent is not permitted to return to the Manual-In mode until a Stroke Count or a Call Work Code is completed. If the Manual-In button is depressed before a

Stroke Count or a Call Work Code has been completed, the Manual-In lamp flutter. If the Manual-In FAC is used before a Stroke Count or a Call Work Code has been completed, intercept tone is given. (However, the agent can subvert the Forced Entry mechanism by going into AUX first or by entering (*) only for forced entry of call work codes.)

Once a Stroke Count or a Call Work Code entry is completed, pressing the Manual-In button (or FAC) returns to Manual-In mode, and lights the Manual-In lamp.

ACD agents with an attendant console or multi-appearance voice terminals can enter Stroke Counts or Call Work Codes.

An Agent is permitted to be logged into 3 or 4 splits (R3/R3V2 CMS) at the same time. Any of these splits may have the Forced Entry option active. A transition into the Aux-Work mode in any split will remove the Forced Entry requirement for all other splits.

The ACD feature must be enabled on the 'System Parameters Customer Options' form. The Call Work Code feature may also be enabled on this screen. If Call Work Code is not selected, the Forced Entry capability applies only to Stroke Counts.

Considerations

ACD is particularly useful whenever a department or answering group receives a high volume of calls of the same type (for example, a catalog ordering department). Members of the department or answering group can be assigned to an ACD split. Call completion time is minimized and, since calls go directly to the split, attendant assistance is not required.

For a complete list of the ACD parameters please see Appendix A, "System Parameters".

A voice terminal or individual attendant can be an agent in one or more splits. However, an agent cannot be logged into more than 4 splits simultaneously. If an agent is assigned to more than one split, each assignment applies to the maximum number of agents. When computing the number of agents measured by BCMS, count one agent as one agent regardless of the number of splits that the agent will be logged into. For CMS, count one agent for each agent in each split measured by CMS; one agent logged into three splits counts as three agents.

Announcements can be analog, aux trunk (G3r only), DS1 (G3V4 and later releases) or integrated. Integrated announcements use the 16-channel announcement board and queuing is based on whether or not one of the 16 channels is available. When a channel becomes available, any of the announcements on the board can be accessed. Therefore, a caller may be in queue for an announcement (because a channel is not available), even though that announcement is not being used. The maximum queue length for all digital announcements is shown in the Capacities Table. Queues for analog and aux trunk announcements are on a per-announcement basis. The capability to install multiple Integrated Announcement boards is available with G3V4 and later releases.

If a delay announcement is used, answer supervision is sent to the distant office when the caller is connected to the announcement. Charging for the call, if applicable, begins when answer supervision is returned.

Calls incoming on a non-DID trunk group can route to an ACD split instead of to an attendant. Calls incoming on any non-DID trunk group can have only one primary destination; therefore, the trunk group must be dedicated to the ACD split.

Agents using multiappearance voice terminals can receive only one ACD call at a time unless Multiple Call Handling is administered. Without MCH, a voice terminal is available for an ACD call only if all call appearances are idle. The agent may, however, receive non-ACD calls while active on an ACD call.

Leave Word Calling messages can be stored for an ACD split and can be retrieved by a member of the ACD split, a covering user of the split, or a systemwide message retriever. The "Voice Terminal Display" feature and proper authorization must be assigned to the message retriever. Also, a remote Automatic Message Waiting lamp can be assigned to a split agent to provide a visual indication that a message has been stored for the split. The status lamp associated with this button informs the user that at least one message has been left for the split.

Each ACD split and each individual agent is assigned a COR. Miscellaneous Restrictions can be used to prohibit selected users from accessing certain splits. Either Miscellaneous Restrictions or restrictions assigned through the COR can be used to prohibit the agents from being accessed individually. Unless such restrictions are administered, each agent can be accessed individually as well as through the split.

CMS measurements may be inaccurate on calls to splits that intraflow to the attendant group.

If an agent becomes available while a caller is listening to an announcement (other than a forced first announcement), the call is removed from the announcement and is connected to the available agent.

For MEGACOM 800 Service with DNIS over a wink/wink-tie trunk, if all agents are logged out or in the AUX-Work mode, incoming MEGACOM telecommunications service calls receive a busy signal if no coverage path is provided (unlike other automatic-in trunk groups which receive ringback from the central office).

When a CO call enters a full ACD split queue, there may be a difference in the switch measurement and the CMS measurement. This is because it is a CO call.

The switch measurement will indicate the maximum number of calls allowed in the queue. The CMS measurement will indicate all the calls in the ACD split queue plus any call on the CO trunk waiting to terminate on the ACD split.

CO switches will usually drop calls which remain unanswered after a period of two to three minutes. Therefore, if an incoming CO call queues to a split without hearing an announcement or music, and the caller listens to CO ringback for two to three minutes, the call will be dropped by the CO.

If an ACD split extension is assigned as the incoming destination of a trunk group, and that split's extension is later changed, the trunk group's incoming destination must also be changed to a valid extension.

Agents should not be used for hunt group calls and ACD split calls simultaneously. Otherwise, all of the calls from one split (either ACD or hunt group) will be answered first. For example, if the ACD calls are answered first, none of the hunt group calls will be answered until all of the ACD calls are answered.

The oldest call waiting termination is only supported for agents who are servicing ACD calls only.

Interactions

The following features interact with the "Automatic Call Distribution (ACD)" feature:

- Attendant Call Waiting

  An attendant can originate or extend a call to an ACD split. Attendant Call Waiting cannot be used on such calls. However, such calls can enter the split queue, if provided.

- Attendant Intrusion

  Attendant Intrusion will not work with ACD split extensions since an ACD extension has many agent extensions. Therefore, it would not be possible for the switch to determine which agent extension to intrude upon.

- Automatic Callback

  Automatic Callback calls cannot be activated toward an ACD split.

- Call Coverage

  Calls can redirect to or from an ACD split.

  A vector-controlled split cannot be assigned a coverage path.

  For a call to an ACD split to be redirected to Call Coverage on the busy criterion, one of the following conditions must exist:

  — Each agent in the split must be active on at least one call appearance and the queue, if there is one, must be full

— No agents are logged in

— All agents are in AUX work mode

If the queue is not full, a call will enter the queue when at least one agent is on an ACD call or in ACW mode. Queued calls remain in queue for a time interval equal to the Coverage Don't Answer Interval before redirecting to coverage. If any agent in the split becomes idle, the call directs to that voice terminal.

Calls that redirect on the ‘‘don't answer’’ criterion are reported to BCMS/CMS as intraflowed calls. Calls that redirect on the ‘‘busy’’ coverage criterion are not reported to BCMS/CMS as intraflowed calls. Calls can be redirected to another ACD split via Call Coverage to activate the "Intraflow and Interflow" feature.

If a call is queued for an ACD split and redirects via call coverage directly to an announcement, the call will be dropped upon completion of the announcement.

When a call is redirected via Call Coverage to an ACD split, the calling party will not hear a forced first announcement or a first delay announcement at the covering split, if administered. The redirected call will receive a second delay announcement only.

Calls to a split that are directed to an agent's voice terminal will not follow the agent's call coverage path. Activating Send All Calls for an agent terminal will not affect the distribution of ACD calls. An ACD call directed to an agent's station will follow the split's call coverage path if the specified "don't answer" interval is met at the agent's set.

- Call Forwarding All Calls

  When activated for an individual extension, the ACD functions of the individual extension are not affected.

  When activated for the split extension, calls directed to the split are forwarded away from the split. No announcements (other than a forced first announcement, if administered) associated with that split are connected to the call. The system reports to the BCMS/CMS that the call is queued on the split and then reports to the CMS that the call has been removed from the queue and forwarded.

  Calls can be forwarded to an off-premises destination to activate the "Intraflow and Interflow" feature. Destinations may be domestic or international numbers that can be reached via the public switched network.

  Calls can be forwarded to destinations outside your PBX (that is, domestic and international phone numbers on the public-switched telephone network).

  On calls forwarded to an ACD split, the caller will hear the forwarded-to split's first and second delay announcement(s), if assigned. A forced first announcement at the forwarded-to split will not be delivered.

- Data Call Setup (to or from a member of an ACD split)

  Voice Terminal Dialing or Data Terminal (Keyboard) Dialing can be used on calls to an ACD split.

- Data Restriction

  If the trunk group used for an ACD call has Data Restriction activated, agents with automatic answer activated will not hear the zip tone that is normally heard.

- DDC and UCD

  Before the System Manager changes a ‘‘hunt group’’ from ACD to non-ACD (DDC or UCD), all agents in that hunt group must be logged out.

  When the System Manager changes a hunt group from ACD to non-ACD (DDC or UCD), all agents in that hunt group are placed in a ‘‘hunt group busy’’ state by the system software. If any voice terminals in the hunt group have an Auxiliary Work button, the lamp associated with that button will light. In order to become available for calls, the agent can press the Auxiliary Work button. Voice terminals without Auxiliary Work buttons can dial the Hunt Group Busy Deactivation feature access code followed by the hunt group number to be able to receive calls.

- DCS

  ACD is not a DCS-transparent feature. CMS cannot measure ACD splits on a DCS network as though they were one switch. Agents for a split must be all on the same switch. If a call to an ACD split is forwarded to a split at another DCS node, the caller will not hear the forced first announcement of the forwarded-to split.

  If an ACD split is in night service, with a split at another DCS node as the night service destination, a call to the first split will be connected to the first forced announcement of the split serving as the night service destination.

- Dial Intercom

  An agent with origination and termination restriction can receive ACD calls and can make and receive Dial Intercom calls.

- EAS

  The interactions of EAS and ACD are closely knit. Please see the "Expert Agent Selection (EAS)" feature section for information on using EAS with ACD.

- Forced ACD Calls

  Multiple Call Handling and Forced ACD Call features are mutually exclusive. Forced ACD Call features are *not* available in G3V4 and later versions.

- Hold

  If an agent puts an ACD call on Hold, information is reported to the CMS via the Personal Call Tracking Feature. Therefore, the CMS records the amount of time the agent is actually talking on the call.

- Individual Attendant Access

  Individual attendant extensions can be assigned to ACD splits. Unlike voice terminal users, individual attendants can answer ACD calls as long as there is an idle call appearance and no other ACD call is on the console.

- Internal Automatic Answer (IAA)

  Internal calls placed to an ACD split number are eligible for IAA.

  IAA and ACD Automatic Answer cannot be administered simultaneously on the same station.

- Multi-Appearance Preselection and Preference

  All assigned call appearances must be idle before an ACD call is directed to a voice terminal.

- Multiple Call Handling

  "Multiple Call Handling" and Forced ACD Call features are mutually exclusive. Forced ACD Call features are available beginning with G3V4 and later releases.

- Night Service — Hunt Group

  When Hunt Group Night Service is activated for a split and the night-service destination is a hunt group, the caller will hear the first forced announcement for the original split, if administered. The call is then redirected to the night service destination hunt group. When an agent in the night service hunt group becomes available, the call goes to that agent. If all agents in the night destination hunt group are busy, the caller will hear the following, if assigned: forced or delayed first announcement, ringback, music-on-hold or silence, and a second announcement.

- Priority Calling

  A priority call directed to an ACD split is treated the same as a nonpriority call, except that the distinctive three-burst ringing is heard (if three bursts have been administered for priority calls).

  A call made to an ACD split from a user or trunk group with a COR that has priority queuing is inserted ahead of normal priority calls in the split queue. However, if the call intraflows to another split without priority queuing, it is queued as a normal priority call in the covering split's queue.

- CDR

  When a CO caller enters a full ACD split queue, CDR and the CMS may show different measurements. CDR measurements indicate the maximum number of calls allowed in the queue, whereas the CMS measurements indicate all calls in the queue plus any call on the CO trunk waiting to enter the split queue.

- Terminating Extension Group

  A Terminating Extension Group cannot be a member of an ACD split.

- Restriction — Origination and Termination

  An agent with Origination and Termination Restriction can receive ACD calls and use the assist feature.

- Termination Restriction (COR)

  A station that is in a COR with Termination Restriction can receive ACD calls.

- Transfer

  Calls cannot be transferred to a busy split. The transfer operation will fail and the transferring party will be re-connected to the call. If the transferring party depresses the TRANSFER button, dials the hunt group extension number, and then disconnects (and the split is busy), the call will be disconnected.

- Vector Disconnect Timing

  This timer can be administered on the System Parameters Form ("Vector Disconnect Timing" line). When disconnect supervision is not available, a call may stay up even *after* the called party hangs up. To prevent this, the timeout may be set (from 1 to 240 minutes) to force-disconnect the call. Vector Disconnect Timing features are available in G3V4 and later versions.

- Voice Terminal Display.

  On calls dialed directly to an ACD split extension number, the calling party's identity (trunk name or user name) and the ACD split's identity (split name) are displayed at the called extension.

Administration

ACD is administered by the System Manager. The following items can be administered for each ACD split (hunt group):

- Split extension number, name, and type of hunting. The type of hunting is administered as DDC (Direct), UCD (Most-Idle Agent), or EAD (Expert Agent Selection).
- Whether or not it is an ACD split.
- Whether the hunt group is a skill hunt group.

- Whether or not the split is adjunct-controlled.
- Whether or not each split is vector-controlled. If a split is vector-controlled, announcement attributes, night service destination, intraflow, coverage path, and message information cannot be administered for the split.
- Whether split measurements are to be internal, external, both, or none.
- When Multiple Call Handling is enabled, if the split/skill has a MCH type of ‘‘none,’’ ‘‘on-request,’’ “one-forced,” “one-per-skill’’ or “many-forced.”
- First announcement extension.
- First announcement delay interval.
- Second delay announcement extension.
- Second delay announcement interval.
- Whether or not the second delay announcement is recurring.
- Night service destination.
- Whether or not calls redirected by Intraflow have priority over other calls.
- Inflow threshold (0 to 999 seconds). If the oldest call in queue has been in queue for this amount of time, the split will not accept any redirected calls.
- Split supervisor extension.
- Split coverage path.
- Class of Restriction.
- Four-Digit security code.
- Type of Message Center the split serves as (AUDIX, LWC reception, AUDIX name, Message Server name, or blank).
- Whether or not the split is served by a queue.
- Queue length (1 to 200 calls for G3i, 1 to 999 calls for G3r).
- Queue Warning Threshold for number of calls (1 to 200 calls for G3i,

  1 to 999 calls for G3r).
- Queue Warning Threshold for time in queue of oldest call (0 to 999 seconds).
- Port Number assigned to auxiliary queue warning lamp (based on number of calls).
- Port Number assigned to auxiliary queue warning lamp (based on time in queue of oldest call).
- Group Members (extension numbers).
- Whether or not agents in the split are required to enter Stroke Counts and/or Call Work Codes.
- Redirection on No Answer (RONA).

Hardware and Software Requirements

Each auxiliary queue warning level lamp requires one port on a TN742 or TN746B Analog Line circuit pack. An indicator lamp may be used as a queue warning level lamp. The lamp operates on ringing voltage and can be mounted at a location convenient to the split.

A reload indicator can be optioned on a button to indicate when a system reload has completed so agents know when calls begin again. Each delay announcement requires either one port on an Integrated Announcement circuit pack or external announcement equipment and one port on a TN742 or TN746B Analog Line circuit pack or a port on an Auxiliary Trunk pack.

NOTE:
There are 16 ports available for listening to any one of the announcements. In addition, any announcement may be played over one or more physical channels on the integrated announcement board. External Announcement Units may also require a port on an Analog Line circuit pack for remote recording capabilities. No more than four external announcements should be assigned on the TN742 ports since it can ring only four ports at a time.

G3V4 and later releases allow for the installation of multiple Integrated Announcement packs (TN750C) in a system. See Appendix A, "System Parameters" for the number of boards that can be installed in each system.

If music is to be heard after an announcement, a music source and a port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) is required. Music sources are not provided by the system.

If a CMS is to be used, CMS hardware is required.

The MERLIN 731x series hybrid voice terminals are not recommended for ACD agent application because in heavy traffic conditions display information may be lost. Furthermore, display functionality is limited to only calling and called information.

Automatic Callback

Feature Availability

Automatic Callback is available with all G3 releases.

Description

Allows internal users who placed a call to a busy or unanswered internal voice terminal to be called back automatically when the called voice terminal becomes available.

A single-line voice terminal user activates this feature by pressing the Recall button or flashing the switchhook and then dialing the Automatic Callback access code. Only one call can be activated at a time by a single-line user.

A multiappearance voice terminal user can activate Automatic Callback for the number of Automatic Callback buttons assigned to the terminal. After placing a call to a voice terminal that is busy or that is not answered, the caller simply presses an idle Automatic Callback button and hangs up.

When activated, the system monitors the called voice terminal. When the called voice terminal becomes available to receive a call, the system then originates the Automatic Callback call. A busy voice terminal becomes available when the user hangs up after completing the current call. An unanswered voice terminal becomes available after it is used for another call and is then hung up.

When the called voice terminal becomes available, the system originates the Automatic Callback call and the calling party receives three-burst ringing (the number of bursts is administrable). The calling party then lifts the handset and the called party receives the same ringing provided on the original call. The ringing at the called voice terminal occurs immediately after the calling voice terminal user lifts the handset.

If the calling voice terminal user answers an Automatic Callback call, and for some reason the called extension cannot accept a new call, the calling user hears confirmation tone and then silence. The call still is queued.

Considerations

The maximum number of callback calls that the system can process at one is shown in Appendix A, "System Parameters".

An Automatic Callback request is canceled for any of the following reasons:

- The called party is not available within 30 minutes.

- The calling party does not answer the callback call within the administered interval (two to nine ringing cycles).
- The calling party decides not to wait and presses the same Automatic Callback button a second time (multiappearance voice terminal) or dials the Automatic Callback cancellation code (single-line voice terminal).

Automatic Callback eliminates the need for voice terminal users to continually redial busy or unanswered calls to internal voice terminals. Instead, the user simply activates Automatic Callback. The system then calls the user back when the called voice terminal becomes available.

Automatic Callback is administered to individual voice terminals by their COS and cannot be assigned to the attendant(s).

Multiappearance voice terminals must have an Automatic Callback button to activate the feature.

Interactions

The following features interact with the "Automatic Callback" feature.

- Attendant Intrusion
- Attendant Call Waiting and Call Waiting Termination

  If the "Automatic Callback" feature is activated to or from a single-line voice terminal, the "Call Waiting Termination" features are denied.
- Bridged Call Appearance

  Automatic Callback calls cannot originate from a bridged call appearance. When a call is originated from a primary extension number, the return call notification rings at all bridged call appearances.
- Busy Verification
- Call Coverage

  Automatic Callback calls do not redirect to coverage.
- Call Waiting
- Call Pickup

  A group member cannot answer a callback call for another group member.
- Call Forwarding

  Automatic Callback cannot be activated toward a voice terminal that has Call Forwarding activated. However, if Automatic Callback was activated before the called voice terminal user activated Call Forwarding, the callback call attempt is redirected toward the forwarded-to party.
- Conference and Transfer

- Internal Automatic Answer (IAA)

  Automatic Callback calls are not answered automatically via the "Individual Attendant Access (IAA)" feature.

- DCS
- Ringback Queuing

  An Automatic Callback button is used to activate the "Ringback Queuing" feature.

- Voice Terminal Display
- Hold
- Auto and Dial ICOM

Voice terminals with the following features cannot activate the "Automatic Callback" feature:

- Hot Line Service
- Manual Originating Line Service
- Restriction — Origination

Automatic Callback cannot be activated to the following:

- The attendant console group
- A voice terminal assigned Termination Restriction
- A voice terminal when using Automatic Call Distribution
- An extension with Automatic Callback already activated toward it
- A data terminal (or data module)
- A Direct Department Calling group
- A Uniform Call Distribution group
- A Terminating Extension Group
- A VDN Extension

Administration

The System Manager assigns Automatic Callback to individual voice terminals by their COS. The following items also require administration:

- No Answer Time-Out Interval (number of times the callback call rings before it is canceled). This interval is assigned on a per-system basis.
- Feature Access Codes — For activating and deactivating Automatic Callback.
- Automatic Callback Buttons — For multiappearance voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Automatic Circuit Assurance (ACA)

Feature Availability

Automatic Circuit Assurance is available with all Generic 3 releases.

Description

Assists users in identifying possible trunk malfunctions. The system maintains a record of the performance of individual trunks relative to short and long holding time calls. The system automatically initiates a referral call to an attendant or display-equipped voice terminal user when a possible failure is detected.

Holding time is the elapsed time from the time a trunk is accessed to the time a trunk is released. When the "Automatic Circuit Assurance (ACA)" feature is enabled by the System Manager, the system measures the holding time of each call.

A short holding time limit and a long holding time limit are preset by the System Manager for each trunk group. The short holding time limit can be from 0 to 160 seconds. The long holding time limit can be from 0 to 10 hours. The measured holding time for each call is compared to the preset limits for the trunk group being used.

A short holding time counter and a long holding time counter associated with each trunk group member are kept by the system. When the measured holding time of a call is compared to the preset limits, these counters are incremented or decremented as follows:

- Measured holding time less than short holding time limit — Short holding time counter is incremented.
- Measured holding time greater than short holding time limit and less than long holding time limit — Short holding time counter is decremented.
- Measured holding time greater than long holding time limit — Long holding time counter is incremented.

The short holding time counter is constantly compared to a preset threshold. This threshold can be from 0 to 30 and is set by the System Manager. The threshold for the long holding time counter is always 1. Each time a counter reaches a preset threshold, two things occur as soon as the system clock reaches the next hour or the call is dropped:

- If ACA referral has been activated by an attendant or voice terminal user, a referral call is sent by the system to a designated attendant console or display-equipped voice terminal.
- An entry is made in an audit trail which stores information on the occurrence.

When ACA is enabled by the System Manager, the ACA measurements are made and the audit trail is updated each time a preset counter threshold is reached. However, in order for a referral call to be sent, ACA referral must be activated. ACA referral is activated whenever an attendant or user presses an ACA button. When this is done, the system can send referral calls to the destination specified by the System Manager.

The referral call destination can be the attendant console group, a specific attendant console, a display-equipped voice terminal, or, if Voice Message Retrieval is provided, a non-display voice terminal. The information appearing on the display identifies the call as an ACA call, identifies the trunk group access code and the trunk group member number, and shows the reason for referral (short or long holding time). When the call is answered, this information is displayed and remains displayed until the call is released.

Each time a counter threshold is reached, a record of the information is stored in the audit trail. The audit trail records are available to the System Manager. Each record contains the following information:

- Time and Date of occurrence
- Trunk group number, trunk access code, and trunk group member
- Type of referral (short or long holding time)

If the referral call destination does not answer the call within three minutes, the call times out and this information is entered in the audit trail. The audit trail is examined once each hour. If any entries indicate a referral call was not completed, the call is tried again.

ACA can be enabled or disabled for the entire system by the System Manager. The System Manager can also enable or disable ACA for each individual trunk group. When ACA is disabled, ACA measurements are not made.

Two extensions must be assigned for the purpose of letting the referral call destination identify the type of ACA call (short or long holding time). The two extensions are assigned as a short holding time origination extension and a long holding time origination extension. These extension numbers do not require hardware circuit packs.

As an illustration of how ACA functions, assume the following:

- The ACA is enabled for the entire system.
- The ACA referral destination is extension 389.
- The ACA long holding time origination extension is 423.
- The long holding time limit for trunk group 3 (trunk access code is 9) is one hour.
- The ACA referral is activated.

With the above information, assume a call is made on a trunk in trunk group 3 and the call lasts more than one hour. Then, the threshold for the long holding time counter is reached, a referral call is made to extension 389, the display reflects a long holding time call, and the information is entered in the audit trail.

The referral destination can then have the operation of the trunk checked and taken out of service if defective.

Considerations

The "Automatic Circuit Assurance (ACA)" feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time. Some types of trunk failures cause people to shorten their calls. For example, an excessive number of short calls may indicate a noisy trunk. Similarly, a trunk that remains busy for an abnormally long time may be permanently busy due to a trunk fault. This feature takes advantage of these characteristics to identify possibly defective trunks. Once the trunk has been identified as possibly being defective, the "Busy Verification of Terminals and Trunks" feature can be used to check the trunk.

The audit trail contains a maximum of 64 records at any one time. The oldest information is overwritten by the newest information.

Measurements are not made on personal central office lines, out-of-service trunks, or trunks undergoing maintenance testing.

If ACA referral calls are sent off the PBX generating the referral, the display information indicating the failed trunk will be lost even if the referral call is made over a DCS network.

Interactions

The following features interact with the "Automatic Circuit Assurance (ACA)" feature.

- CAS

  When CAS is activated, the referral call destination must be on the local switch. A referral destination of ***0*** is interpreted as the local attendant, if one exists.

  The CAS attendant cannot activate or deactivate ACA referral calls at a branch location.

- Internal Automatic Answer (IAA)

  Calls generated by the "Automatic Circuit Assurance (ACA)" feature are not eligible for IAA.

- Night Service — Night Station Service

Referral calls will not be placed if the system is in the Night Service mode.

- Visually Impaired Attendant Service

  Automatic Circuit Assurance applies for both Station and Attendant users. However:

  — If the attendant presses the DISPLAY STATUS button and an incoming ACA call has not been answered, then `Automatic Circuit Assurance` is voiced to the visually-impaired attendant.

  — If the attendant presses the DISPLAY STATUS button and the incoming ACA call has been answered, then `Automatic Circuit Assurance` and the extension number assigned to the phantom ACA call are simultaneously voiced to the visually-impaired attendant.

  — If the switch is equipped with a voice synthesis board, ACA referral calls are accompanied by an audible message identifying the type of ACA infraction encountered. The message is “Automatic circuit assurance <long> or <short> holding time threshold has been exceeded for trunk group <#> member number <#>. The member number here is actually the Trunk Access Code (TAC).

Administration

ACA is administered by the System Manager. The following items require administration:

- Whether ACA is enabled or disabled (per system)
- Short holding time origination extension (per system). Assigned name must reflect short holding time nomenclature.
- Long holding time origination extension (per system). Assigned name must reflect long holding time nomenclature.
- Referral destination (per system)
- Whether ACA is assigned (per trunk group)
- Short holding time limit (per trunk group)
- Long holding time limit (per trunk group)
- Threshold for short holding time counter (per trunk group)
- ACA activate/deactivate button on attendant console or voice terminal (one per system)

Administer these items on the ‘‘Feature Related Parameter’’, ‘‘Attendant Console’’, ‘‘Trunk Group’’, and ‘‘Station’’ forms.

Hardware and Software Requirements

A TN725 Speech Synthesizer circuit pack is required if the referral destination is not a display-equipped voice terminal. A TN457 Speech Synthesizer circuit pack is required to hear ACA messages is English, and a TN433 Speech Synthesizer circuit pack is required to hear ACA messages in Italian.

No additional software is required.

Automatic Incoming Call Display

Feature Availability

Display of Incoming Calls is available with all Generic 3 releases.

Description

Provides display-equipped voice terminal users who are already active on a call with the identity of a second or subsequent caller. The identity is automatically displayed on the terminal's alphanumeric terminal for 30 seconds. The displays (which are administrable) can be in English, French, Italian, or Spanish. This feature applies when an incoming call terminates at a user's voice terminal while the user is active on another call appearance. The information displayed on the current call is replaced by the identity of the incoming call. The identity of the incoming call normally remains displayed for 30 seconds unless there is another incoming call, the user hangs up, or the calling party hangs up. After 30 seconds, the display returns to the current call information. With the CALLMASTER terminal, the display goes blank after 30 seconds.

A third or subsequent incoming call overwrites the information displayed on the previous call and restarts the 30-second interval. In any case, the most recent call to terminate at the user's voice terminal is the call identified by the display.

If the party whose identity is currently being displayed hangs up, the display returns to the current call information. If the user hangs up on the current call before the 30-second interval expires on the incoming call, the display is cleared.

The information displayed on the current call is not replaced by the identity of the incoming call if the called user is in the process of dialing the current call or if the Outgoing Display Option is not administered to the trunk group being used.

Considerations

The "Automatic Incoming Call Display" feature lets certain users, while active on one call, know the identity of another incoming caller. This is done without the use of an Inspect button. By knowing who is calling, the user can handle the calls accordingly.

The incoming call must terminate at the user's voice terminal in order to be displayed. Calls forwarded to another extension are not displayed.

The display must be in the normal mode to display the identity of the incoming call. The Automatic Incoming Call Display does not apply under the following conditions:

- The called station user is not off-hook on a call.
- The called user's display is turned OFF.
- The called user's display is in any other mode other than normal mode.
- The called station is being dialed.

Interactions

This feature enhances the "Voice Terminal Display" feature by providing automatic identification of incoming calls. The same incoming call information can be provided by putting the display in the inspect mode; however, this is not automatic and must be done manually for each call.

When a station is optioned for PC/PBX, then, depending on the set type, the call appearance designation (a=, b=, c=, d=) may be omitted from the displayed information in order for the *find* feature of the Call Log in the PC/PBX Connection software to work properly.

Administration

Unless you want to administer a language other than English for the displays, no administration is required. See the "Administrable Language Displays" feature for a description of multi-language displays.

Hardware and Software Requirements

Requires a 515 BCT, a display-equipped voice terminal, or a voice terminal capable of displaying information through an attached data terminal. No additional software is required.

Automatic Route Selection (ARS)

Feature Availability

This feature is optional with all Generic 3 releases.

Description

Routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

ARS provides a choice of routes for any given public network call. The following types of trunk groups can be accessed by ARS:

- **Local central office** — Used for local calls and to provide access to a long-distance carrier. Access to the long-distance carrier can be provided either automatically by the central office or by a carrier access code (where supported).
- **Foreign exchange** — Used to emulate local calling in an area not served by the local central office. Like the local central office, the foreign exchange office provides a choice of long-distance carriers.
- **WATS** — Used to provide calling to predefined geographic areas at a rate based on expected usage.
- **Tie trunks** — Used to provide access to an ETN, or to an EPSCS or CCSA office. (In some cases, it is preferable to allow a private network to handle the routing of calls destined for the public network.)
- **ISDN-PRI** — Used for calls over an ISDN and provides users access to a variety of switched nodal services such as MEGACOM telecommunications service, INWATS, and ACCUNET digital service and allows access to other inter-exchange carriers or private networks.

A variety of numbers can be called using ARS, including 7-digit numbers, 10-digit numbers, International Direct Distance Dialing (IDDD) numbers, service codes, Customer-Dialed Operator-Serviced (CDOS) numbers (for example, in the United States, 0+ or 01+), and Inter-Exchange Carrier (IXC) numbers.

ARS is particularly useful when one or more long-distance carriers and WATS are provided. The system selects the most preferred (normally least expensive) route for the call. Long-distance carrier code dialing is not required on routes selected by the system. Long-distance carrier codes are assigned in translations to best benefit the customer on any given call. These codes are inserted as needed to guarantee automatic carrier selection.

The system may serve as an ETN tandem switch. In this case, the system can access or be accessed by Intertandem Tie Trunks to/from other tandem switches and/or Access Tie Trunks to/from ETN main switches. The system can also access Bypass Tie Trunks to an ETN main switch. This distinction as a tandem switch is important with respect to the routing of certain calls.

ARS Dialing

ARS begins when a user dials the ARS access code (normally the digit 9), followed by the number to be called.

As soon as the user dials the ARS access code, the system checks to see if the user's voice terminal extension has been Origination Restricted or Outward Restricted by its assigned COR. The system also checks to see if the user has a Controlled Restriction of either Outward or Total. If any of these restrictions apply, intercept treatment is applied to the call. Otherwise, the ARS call continues and the user can enter the number to be called.

A second dial tone may or may not be heard after the ARS access code is dialed, depending on the system administration.

Inter-Digit Timeout

The system uses a short inter-digit timer and a long inter-digit timer during the dialing process. Normally, a 10-second inter-digit timer is used between each digit for the user to continue dialing. If the digits dialed so far point to a valid destination, but there is a similar string of digits which is of different length, the short three-second inter-digit timer will be started. If dialing does not continue before the timer expires, it is assumed that no more digits will follow, and # is appended by the system to indicate end of dialing. To override the timer for faster call processing, the originator may dial # to indicate end of dialing.

A 10-second long inter-digit timer is used to wait for another digit when the digits dialed so far are not a valid destination. Timeout of this timer results in Intercept tone to the caller.

When no length ambiguity exists and all digits are collected, the call is routed, and no timer or # is required.

Special Dialing Patterns

The system recognizes certain dialing patterns on outgoing calls and routes these calls accordingly. The descriptions of these dialing patterns reflect the system defaults as used in the United States. Other countries may require different administration of these values. The following dialing patterns are recognized:

- **DDD Calls With Prefix Digit 1 Required**

  The user may or may not be required to dial a 1 before dialing a seven- or 10-digit number, depending on the system's dial plan administration. There are two cases where the digit 1 must be dialed:

— Some metropolitan areas are so densely populated that there simply are not enough traditional central office codes. Therefore, it is possible that some NPA codes, also called *area codes* may also serve as CO codes. In this case, the digit 1 must be dialed if a 10-digit call is intended. The first digit tells the system whether to route the call as a seven-digit call within the home NPA (1 not dialed) or as a 10-digit call to another NPA (1 dialed). In this case, the dial plan should be administered so that the user is required to dial 1 for 10-digit calls. (This is unnecessary in G3r.)

— Digit 1 dialing may also be required in areas near an NPA boundary. In these areas, certain calls to the adjacent NPA may be local calls rather than toll calls. However, central office codes may be duplicated in the home and adjacent NPAs. Also a CO code in the home NPA may be a toll call. Therefore, if the digit 1 is not required on certain adjacent NPA local calls, then it must be dialed on the home NPA seven-digit toll calls so the system can differentiate between the intended destinations.

- **DDD Calls with Prefix Digit 1 Not Required**

  The first digit following the ARS access code may or may not be a 1. In systems where the 1 prefix is dialed, but not required (as administered on the 'Dial Plan' form), dialing the 1 prefix before a 10-digit call is optional and the prefix will be ignored.

- **IDDD Calls**

  IDDD numbers consist of a Country Code and a National Number. The National Number is simply the number used when calling within the country. The Country Code can be from one to three digits in length. In the NANP the National Number is 10 digits in length. The Country Code and National Number together cannot exceed 12 digits. In the NANP, international numbers are recognized by special prefix codes:

  — **011** — Indicates that the caller is making a station paid direct international call. The Country Code and National Number follow the 011 prefix.

  — **01** — Indicates the caller desires operator assistance on an international call, such as person-to-person, credit card, collect call, and so on. The Country Code and National Number follow the 01 prefix.

- **Operator Assistance Calls**

  The first digit following the ARS access code is a 0. If a 0 is dialed by itself to access an operator, a special inter-digit time-out occurs, the route for dial 0 calls is selected and a 0# is outpulsed. If the user dials another 0, the route for 00 is selected and a 00# is outpulsed. The call is routed to the toll operator (if one exists) instead of the local operator in this case.

- **Operator Assisted and International Calls**

  The first digits following the ARS access code are 0 (operator) or 00 (toll operator) optionally followed by a 10-digit DDD number, or 01 or 010 (international operator) for international dialing followed by international destination address digits. Because of the variable number of digits required on these calls, an inter-digit time-out is used to recognize end of dialing.

- **Special Service Codes**

  The first three digits following the ARS access code are of the form X11 (where X = 0 through 9) with or without dialing the 1 prefix digit. This is called a service code. These are recognized as complete addresses if no further digits are dialed, and are routed to the appropriate facility. If it is administered with a length of three to seven digits, the inter-digit time-out determines whether the call is a 3- or 7-digit call. For example, if the user dials ***911***, the call will route to the police/emergency operator; if the user dials ***811-XXXX***, the call will be translated as a seven-digit call for the repair bureau corresponding to the last four digits (***811*** is a service code for repair). In any case, the call is routed based on the first three digits (***X11***) for these special services. If the first three digits after the prefix digit (if any) are not in the form X11, further processing is required to route the call.

- **Calls Dialed with Inter-Exchange Carrier (IXC) Access**

  The first digits following the ARS access code are an IXC Access Code. The access code may be followed by a DDD or an IDDD number. This gives the user control over which carrier or facilities should be used for routing the call. The call is routed based on the administration of the IXC prefixes in the ARS Digit Analysis Table (and Page 2 of the "IXC" form) discussed later in this chapter. For G3r, the system may ignore the IXC code when routing if it cannot find a route using the IXC code.

The system supports access to three general IXC arrangements which are commonly referred to as Feature Groups A, B, and D:

- Feature Group A access dialing is of the form NXX-XXXX (where N is any digit 2 through 9, and X is any digit from 0 through 9) and may be followed by a Personal Identification Number (PIN) (for example, 800-XXXX).
- Feature Group B access dialing is of the form 950-0XXX or 950-1XXX (where X is any digit from 0 through 9) and may be followed by a PIN.
- Feature Group D access dialing is of the form 10XXX (where X is any digit from 0 through 9). As of 1993, equal access codes of the form 101XXXX will be permitted.

From a caller's perspective, the major differences between use of the various groups are:

- Access to Feature Groups A and B requires the dialing of seven digits, whereas access to Feature Group D requires just five digits.

- Single-stage dialing is supported for access to Feature

  Group D, whereas access to Feature Groups A and B requires two-stage dialing. Two-stage dialing means that there is a pause for dial tone between the two groups of dialed digits.
- No customer identification digits are required for access to Feature

  Group D.
- A touch-tone telephone is required to enter a PIN code when accessing Feature Group A or B. A rotary or touch-tone telephone may be used with Feature Group D.

Digit Conversion

When the ARS access code and the called number are dialed, the dialed number is compared to entries in the Matching Pattern fields of the ARS Digit Conversion Table screen. If all or part of the dialed number matches one of the Matching Patterns on the form, the matching part of the dialed number is replaced by a new number from the “Replace” field on the form. This new number is then used to route the call. The new number will route the call via AAR over a private network. If no corresponding entry is found in the AAR Digit Analysis table, the ARS Digit Analysis table is searched for a match with the new modified number and routed accordingly. (An intercept tone is supplied if a match is not found and the call fails.)

In G3r, digit analysis precedes toll analysis. Digit analysis implies both route and conversion. The algorithm for deciding how to analyze a call is as follows:

- number of digits
- most digits matched
- explicit digits over wildcards (from left to right)
- conversion nodes over route nodes

The dialed number is compared with digit string entries on the ‘System’s Toll Analysis’ form for a relationship with the system’s Restricted Call List or Unrestricted Call List. If the dialed number matches a number assigned to a Restricted Call List, the call may not be allowed to complete. Also, if the user is toll restricted and the dialed number matches a number on the Toll List, the call may not be allowed to complete. Calls that are not allowed to complete receive intercept treatment when allowed by the unrestricted call list.

The primary purpose of ARS digit conversion is to convert public network numbers (to be routed with ARS) to private network numbers (to be routed with AAR).

NOTE:
You can convert back to another AAS number or an extension, where the extension most likely is a UUP.

This can save toll charges and allow users to route calls on the customer's private network facilities. Also, unauthorized public network calls can be blocked and routed to an attendant or receive intercept treatment via digit conversion.

Table 3-12 shows several ARS Digit Conversion examples.

The following conditions are assumed for the examples: ARS Access Code = 9, AAR Access Code = 8, Home RNX (Private Network Office Code) = 222, Prefix 1 is required on all long-distance DDD calls, Dashes (-) shown in the table are for readability only. (G3r analysis is more flexible because multiple conversions are permitted.)

Table 3-12. ARS Digit Conversion Examples

Operation	Actual Digits Dialed	Matching Pattern	Replacement String	Modified Address	Notes
DDD call to ETN	9-1-303-538-1345	1-303-538	362	362-1345	The call will be routed via AAR on the route selected for RNX 362.
Long-distance call to presubscribed carrier	9-10222	10222+DDD	(blank)	(blank)	The call will be routed as dialed with the DDD number over the customer's network facilities.
Terminating a local DDD call to an internal station	1-201-957-5567 or 957-5567	1-201-957-5 or 957-5	222-5	222-5567.	The call goes to the home RNX 222, Extension 5567
Unauthorized call to intercept treatment	9-1-212-976-1616	1-XXX-976	#	(blank)	The "#" signifies the end of dialing. Any digits dialed after 976 are ignored by ARS. The user will receive intercept treatment.
International calls to an attendant	9-011-91-672530	011-91	222-0111#	222-0111	The call is routed to local switch (RNX 222), then to the attendant (222-0111). This method may also be used to block unauthorized IDDD calls. The call can be routed to an announcement by replacing 0111 with an announcement extension.
International call from certain European countries needing dial tone detection	0-00-XXXXXXX	00	+00+	00+XXXX	The first 0 denotes ARS, the second pair of zeroes denotes an international call, the pluses denote "wait," for dial tone detection

NOTE:
The dialed digits are matched to the Matching Pattern that most closely matches the dialed number. For example, if the dialed string is 957-1234 and matching patterns

957-1 and 957-123 are in the table, the match is on pattern 957-123. The call will be routed as dialed.

Time of Day Routing

After an ARS call passes through ARS digit conversion (with no Matching Pattern found) and toll analysis allows the call, the Time of Day Plan Number of the calling party is used to make the choice of an associated 'Time of Day Routing' form. On this form, a RPN is identified based on the time of day that is identified to the Partition Group Number. This plan is then used to select the specific partition of the 'ARS Digit Analysis' form, discussed later in this chapter, which will determine how the call is routed.

If Time of Day Routing is not assigned and partitioning is enabled, the user's PGN is used to select the specific 'ARS Digit Analysis' form.

See the "AAR/ARS Partitioning" and "Time of Day Routing" features for more information. G3r searches analysis and conversion tables simultaneously.

ARS Digit Analysis

The following applies to G3vs/G3s, G3i, and G3i-Global. ARS calls that pass through ARS Digit Conversion, Toll Analysis, and Time of Day Routing, are analyzed based on the Time of Day Routing Plan Number or (if Time of Day Routing is not assigned) the user's PGN.

The system uses ARS Digit Analysis to compare the dialed number with entries in an ARS Digit Analysis Table. When the system finds a Dialed String entry in the table that matches the dialed number, the ARS Digit Analysis Table maps the dialed number to a specific Routing Pattern, discussed later in this chapter, and Call Type. The selected Routing Pattern will then be used to route the call. The ARS Digit Analysis Table screen also shows the minimum and maximum number of trailing digits required for digit analysis of each dialed number. G3r supports node number routing.

Possible Call Types in the ARS Digit Analysis Table are as follows:

- 10-digit call within North America
- 7-digit call within North America
- International call (in the United States, the international prefix is 011)
- International operator
- Operator-assisted call (0+)
- Service call (such as 811 for repair or 911 for emergency)
- National numbers within a country (used outside North America)
- Unknown call

Some special dialing patterns are automatically mapped to a specific Routing Pattern and Call Type. They can, however, be changed by the system administrator and probably will be changed for PBXs used outside North America. See *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for a list of these default translations.

Normally, the “Route Pat” (routing pattern) field on the ARS Digit Analysis Table screen contains a routing pattern number. However, this field may instead contain a Remote Home Numbering Plan Area (RHNPA) table number (r1 through r32). An RHNPA is simply a concentrator for up to 1,000 calls. Calls are routed to these tables by ARS/AAR Digit Analysis when an ARS Digit Analysis table points to an RHNPA table. The next three dialed digits (the code) are compared with the codes in the selected RHNPA table. Each code on the table is then mapped to a specific routing pattern number (one through 254).

The RHNPA tables allow up to 1000 codes to be handled by one entry in the Digit Analysis table.

In summary, Digit Analysis is merely a method of selecting a routing pattern. The routing pattern may be selected in two ways:

- It may be selected directly from the Digit Analysis table.
- The Digit Analysis table may first have to select an RHNPA table which will in turn select the routing pattern.

Routing Patterns

The digit translations performed on an ARS call by the Digit Analysis and RHNPA tables cause a specific Routing Pattern to be selected for the call. More than one combination of dialed digits can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with ARS, digit translation should always point to a Routing Pattern. This way, calls to unassigned office codes will be intercepted by the central office, not by the system. By allowing the unassigned codes to be intercepted by the central office, the System Manager does not have to keep track of which office codes are in service. If calls to some codes are to be denied, this should be handled by FRL assignment, not by intercept on the codes. "Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs)" are discussed elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of trunk groups that can be used for the call. Trunk group access is controlled by FRLs. If access to the public network is through a main switch (an Access trunk group is selected for the call), then the call will route through the main switch to one of the public network offices serving the main switch. The digit manipulation necessary to route the call is controlled by the "Subnet Trunking" feature. See the "Subnet Trunking" feature. Otherwise, the digit string to be outpulsed is controlled by ARS. ARS digit manipulation is called code conversion. Code conversion is used to determine whether or not to outpulse the digit 1 on toll calls and whether to insert, keep, or delete the NPA on toll calls.

The following paragraphs describe how the switch decides what digits to outpulse in specific situations.

Digit 1 Outpulsing

The digit 1 may or may not be required at the public network office to which the call will be routing. (If 1 is dialed on 7-digit calls at a stand-alone system (non-ETN), the 1 is outpulsed by the system.) In the other cases, the 1 outpulsing requirements are indicated in the system. Since any given call may have a choice of routes, some of which may require a 1 and some of which may not, this indication is associated with each route. Five choices are available and are identified in translations by a Prefix Mark. Digit 1 outpulsing only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table. The values and meanings of the Prefix Marks are as follows:

- Prefix Mark 0 — Suppress a user-dialed Prefix digit 1 for 10-digit FNPA calls, but leave a user-dialed Prefix digit 1 for the following types of calls:
  - 10-digit calls that are not administered as FNPA or HNPA types in the ARS Routing Table.
  - 7-digit HNPA calls
- Prefix Mark 1 — Send a 1 on 10-digit calls, but not on 7-digit calls.
- Prefix Mark 2 — Send a 1 on all toll calls (for example, all 10-digit calls and 7-digit toll calls).
- Prefix Mark 3 — Send a 1 on all toll calls and keep or insert the NPA to ensure that all toll calls are 10-digit calls. Note that a user-dialed Prefix digit 1 for a 7-digit call makes it a toll call and, hence, NPA is also inserted in this case.
- Prefix Mark 4 — Always suppress a user-dialed Prefix digit 1.

NOTE:
This capability is required, for example, when routing ISDN-PRI calls to an AT&T 4ESS. If the prefix digit 1 were not suppressed, then the 4ESS would reach calls.

Which of the five possible treatments of the 1 prefix digit should be administered on a given route is based on the characteristics of the distant office. Prefix Mark 0 is prevents the system from sending a 1 prefix digit for 10-digit FNPA calls. However, the system leaves a user-dialed prefix digit 1 for 7-digit HNPA calls and 10-digit calls that are not administered as FNPA or HNPA types in the ARS Routing Table.

Prefix Mark 1 causes the system to send a 1 prefix on all 10-digit FNPA calls.

With Prefix Marks 2 and 3, the decision is based on whether the call is a toll call. Toll Lists are provided in the system to furnish this information. A Toll List simply indicates if the office code associated with the call constitutes a toll call from the interconnecting office (not from the local system). Up to 32 Toll Lists are provided. The applicable list number, if any, for the call is given in the Routing Pattern.

Prefix Marks are only applicable on 7- or 10-digit DDD public network calls. Requirements for outpulsing a 1 are specified via Prefix Marks and go into effect when the call accesses is outpulsed. Digit 1 outpulsing only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table.

NPA Deletion and Insertion

Each public network route in the ARS Routing Pattern contains an indication of the NPA of the distant end of the trunk group. If this NPA is the same as the NPA associated with the call, the NPA is deleted prior to outpulsing unless the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

NPA deletion and insertion only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table.

The NPA is inserted on 7-digit calls if the distant NPA is different from the home NPA or if the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

The preceding paragraphs describe NPA deletion or insertion when the call is an ARS 7- or 10-digit DDD call. An ARS call accessing a tandem trunk is another example of NPA insertion. If the call is a 7-digit ARS call, the system inserts the home NPA before sending the call to the tandem trunk. Therefore, all ARS calls accessing a tandem trunk are 10-digit calls. Whether or not the digit 1 is sent on a tandem call is determined by the prefix rules. This enables the system to distinguish between ARS calls and the 7-digit on-network calls.

IDDD and Service Code Dialing

IDDD calls other than those using Subnet Trunking need not be modified before outpulsing. Since international numbers can be of variable length, the system awaits a dialing time-out before processing the call. The US ARS default dialing time-out is three seconds for the 0 and 1 prefix digits, but is 10 seconds for the called number. In other countries, the three second timer will apply to all numbers administered as valid dialed destinations that also happen to be sub-strings of a longer valid dialed destination. The calling party can speed up call processing by dialing the end-of-dialing digit # after the called number. Receipt of this digit cancels the remaining time-out interval. The system always outpulses the # digit for use by the distant switch, whether dialed by the calling party or not (unless it is an ISDN trunk or it is suppressed on the ‘Trunk’ form).

Subnet Trunking is not required for service codes. If the prefix digit 1 is dialed before the code, it is outpulsed.

ARS can provide individual Routing Patterns for each type of call. An ARS call can be processed via the RHNPA table. This is particularly useful on international calls, since the RHNPA table can be used on the country code. Thus, call routing can be determined according to the called country, rather than handling all international calls alike.

Operator and Operator-Assisted Calls

Calls to an operator (0 by itself) with 0 as the attendant access code, require a three-second time-out or dialing of the # digit before the call is processed. Operator-assisted calls (0 plus a 7- or 10-digit number) require 10-digit dialing if the call is within a home NPA and there are office codes within the home NPA which look like NPAs. (On directly dialed calls, this distinction was made by prefix digit 1 dialing.) All other dialing is the same as direct dialing.

Operator-assisted calls, like IDDD calls, can be routed on the first three digits of the called number. Through the use of Subnet Trunking, this means that different long-distance carriers can be selected for different calls.

These examples are for the US and will differ in other countries.

Considerations

ARS provides the most-preferred usage of public network facilities available at a system.

Routing Patterns, toll lists, and RHNPA tables are shared between partitions.

Internal memory resources used for ARS Digit Analysis are shared by "Automatic Route Selection (ARS)", "Automatic Alternate Routing (AAR)", Digit Conversion, and Toll Analysis features. A “Percent Full” field on the ‘ARS’ and ‘AAR Digit Analysis’ screens indicate how many of these resources have been used.

If a customer changes ARS routing assignments, it is the customer’s responsibility to notify the RSC network designer and the SCO technician of the changes in order to receive their continued support.

Interactions

The following features interact with the "Automatic Route Selection (ARS)" feature.

- AAR

  ARS and AAR can access the same trunk groups and share the same Routing Patterns, toll lists, and RHNPA tables. ARS calls may be converted to AAR calls.

- ARS

  When an ISDN/BRI station dials sufficient digits to route a call, but could route differently if additional digits were dialed, the station will not recognize the Conference or Transfer buttons. The user must delay dialing for three seconds or dial a # to indicate the call can be routed based on the digits already dialed. The Conference or Transfer buttons are then recognized and the operation is completed by the switch.

- Abbreviated Dialing

  FRL checking is bypassed on an ARS call made via a privileged Abbreviated Dialing Group List.

- Attendant Control of Trunk Group Access

  Attendant control of a trunk group, in effect, removes the trunk group from the Routing Pattern. The trunk group is never accessed by the "Automatic Route Selection (ARS)" feature. ARS calls do not route to the attendant.

- CAS

  A CAS Attendant may extend a call out of a Branch PBX by use of ARS. The call is extended over an RLT by dialing the appropriate feature access code and number. The call is routed as determined by ARS administration at the Branch PBX.

- Toll Restriction

  Toll Restriction is checked on ARS calls.

- Controlled Restriction, Origination Restriction, and Outward Restriction

  These features prohibit access to the "Automatic Route Selection (ARS)" feature.

- Forced Entry of Account Codes

  Prefix marks and other digits inserted from routing patterns will not be used in determining whether a call is a toll call. See the Forced Entry of Account Codes feature for more information.

- GRS

  The "Generalized Route Selection (GRS)" feature works with ARS to provide call routing over the appropriate trunking facilities. Routing is determined by the type of call being made. With GRS, calls may be routed differently than they would with just ARS. For details on GRS, see the "Generalized Route Selection (GRS)" feature description elsewhere in this chapter.

- Miscellaneous Trunk Restrictions

  Miscellaneous Restrictions are not checked on ARS calls.

- PCOL

  A trunk assigned as a PCOL may not be assigned to any ARS routing pattern.

- Ringback Queuing

  Ringback Queuing can be used on ARS calls originated at the switch that provides the queuing. Incoming tie trunk calls will not queue on an outgoing trunk group.

If a multi-appearance voice terminal user has an Automatic Callback button, makes an ARS call, and all trunks are busy, Ringback Queuing is activated automatically.

- CDR

  An ARS call using a trunk group marked for CDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number. Subnet Trunking does not affect CDR.

  G3r can record outpulsed digits. If CDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group. If an CDR account code is to be dialed with an ARS call, it must be dialed before the ARS access code is dialed.

- Voice Terminal Display

  The voice terminal display shows the dialed digits (not outpulsed digits). The called-party shown on the display is that of the trunk group actually used. The “Miscellaneous Call Identification” field on the display will show ARS.

  An ISDN-BRI station may format these display fields differently, and the timing of display updates may be different.

Administration

ARS is initially assigned on a per-system basis by an AT&T service technician. After the feature is activated, the following items are administered by either the System Manager or the service technician:

- ARS Access Code 1 (one to three digits)
- ARS Access Code 2 (one to three digits)
- ARS Digit Analysis Table (1 per PGN)
- ARS Digit Conversion Table
- Up to 32 RHNPA Tables
- Up to 32 Toll Lists
- FRLs — Assigned via COR to each originating facility.
- Up to 254 ARS Routing Patterns
- Trunk Groups to be used with ARS
- Whether or not the system returns dial tone after the ARS FAC is dialed on trunk calls

Hardware and Software Requirements

ARS may be used on a stand-alone system or may be an integral part of a private network. No additional hardware is required for a stand-alone system.

A private network may require additional tie trunks and tone detector circuit packs such as TN744C-Tone Detector/Call Classifier, TN748C-Tone Detector, TN420C-Tone Detector, or TN2182-Tone Clock/Detector/Generator. TN420C, TN744C, and TN2182 support A-law. These additions are, however, cost effective when compared to the alternatives for call routing.

Optional ARS software and ARS/AAR Digit Conversion software is required in G3vs/G3s and G3i. For G3r, ARS/AAR Digit Conversion is included in ARS. Also, for G3r, ‘‘Node’’ (Node Number Routing) may be included.

Automatic Transmission Measurement System (ATMS)

Feature Availability

Automatic Transmission Measurement System is available with Generic 3rV1 and all Generic V2 and later releases.

Description

The Automatic Transmission Measurement System (ATMS) provides for voice and data trunk facilities to be measured for satisfactory transmission performance. The performance of the trunks are evaluated according to measurements produced by a series of analog tests and are compared against user defined threshold values. The reporting mechanism provides for a way to display the measurements produced by these tests. This feature requires knowledge of traffic engineering terms and concepts in order to obtain useful information. See the *DEFINITY Communications System Generic 3 Version 4 Traffic Reports*, 555-230-511, for additional information.

The measurement report contains data on trunk signal loss, noise, signaling return loss, and echo return loss. The purpose of the report is to provide measurement data to help customers determine the quality of their trunk lines.

These measurements are produced by a set of analog trunk tests generated on the TN771B (or later version) board. The tests are initiated by a maintenance demand test or by a set of scheduled tests. The biggest part of these measurements is collected through scheduled testing during system quiet hours (hours where the traffic volume is low).

Each trunk test performed by the system stores the results of the test in a database. The trunk measurements in this database reflect the current state of each trunk at the time of its last test. If trunks are being tested on a regular basis, trunk measurements should not be any more than two to three weeks old.

The measurement data can be output to the G3-MT printer, to the system printer, or a G3-MA. If a printer is attached, then the data is printed directly to the printer. If a G3-MA is attached to the port, then the data is collected by the personal computer for off-line data analysis.

The commands used to generate ATMS measurements are described in the Implementation guide. Further information is available in the Reports guide and G3-MA manual.

Considerations

The number of pages generated for each report are dependent upon the selection criteria and the number of outgoing trunks in the system. Approximately 10 measurements can be listed per page on the G3-MT screen or about 50 measurements can be listed on a printer.

The entire set of measurements for all test cases is usually be polled by an adjunct for historical reporting. By default all measurements are printed. Ranging and filtering is available to limit the amount of information printed. (Ranging and filtering are not supported in G3iV1, V2, and later releases.)

ATMS reports allow a user to review individual trunk measurements from the most recent test entries. The number of transmission parameters that can be measured depends on the type (Terminating Test Line) and the transmission test executed on that. For example, some TTLs allow you to measure loss in one direction at one frequency only, while other TTLs allow you to measure up to 18 different parameters (such as the TN771B circuit pack). The kinds of measurements that can be reported on different TTLs with ATMS are:

- Two-way transmission loss at 404 Hz, 1004 Hz, and 2804 Hz
- Near-end and far-end C-message and C-notched noise
- Near-end and far-end signaling and echo return loss
- Central office 100-type TTLs
- One-way transmission loss at 1004 Hz
- Near-end C-message noise
- Near-end signaling and echo return loss
- Central office 102-type TTLs
- One-way transmission loss at 1004 Hz
- Central office 105-type TTLs
- Two-way transmission loss at 404 Hz, 1004 Hz, and 2804 Hz
- Near-end and far-end C-message and C-notched noise
- Near-end and far-end signaling and echo return loss

Terminating Test Lines

A terminating test line is an extension over a DID or tie trunk that users call into in order to generate a random set of far-end measurements (see bullets above for examples of measurements) about the facility used to make the call. The originating switch uses the far-end measurements to compare with the near-end measurements that it obtains to determine the quality of the facility (per customer set thresholds).

Interactions

The report interacts with the trunk threshold values defined on the ‘Trunk Group’ form. The marginal values defined on that form are used to highlight measurements on the G3-MT report which are out of tolerance. The unacceptable values are used to identify (by blinking) the trunk members on the G3-MT report which are unusable. These mechanisms help a user to quickly identify out of tolerance or unusable trunks.

Administration

The "Automatic Transmission Measurement System (ATMS)" feature must be turned *on* in the ‘System Customer Options’ form in order to access these reports. The extension assigned to the TN771B or later must be set to a COR that allows access to Facility Test Calls.

Hardware/Software Requirements

The TN771B or later version circuit pack is required in order for ATMS to function. The customer must acquire the license for ATMS to be activated on the switch. The G3-MA is convenient but not required.

Automatic Wakeup

Feature Availability

Automatic Wakeup is available with all Generic 3 releases.

Description

Allows attendants, front desk users, and guests to request that a wakeup call be placed automatically to a certain extension number at a later time. Wakeup requests may be placed from five minutes to 23 hours and 55 minutes in advance of the wakeup call.

When a wakeup call is placed and answered, the system can provide a recorded announcement, speech synthesis announcement, music, or simply silence.

All wakeup times entered into the system are rounded to the nearest five minutes. For example, a requested time of 6:58 a.m. would be stored in the system as 7:00 a.m. Time validity checks are based on the rounded figure.

Wakeup calls are placed within two and one half minutes of the requested time, and are never rerouted, forwarded, or sent to coverage. Prior to placing the wakeup call, the system overrides Do Not Disturb for the extension, if applicable.

If a wakeup call attempt is not answered or if the extension is busy, the system will try two more times at five-minute intervals. If the call is not completed after the three attempts, the system can leave a LWC message for a designated extension, if administered. In addition, the system maintains a complete record of all wakeup call activity for the past 24 hours.

Touch-tone dialing is required for a wakeup request to be entered. Users with rotary dial terminals must call the attendant to request a wakeup call.

The "Automatic Wakeup" feature can be activated either by dialing the FAC or by pressing the Automatic Wakeup Entry button. If the FAC is used, the system provides voice prompting. If the Automatic Wakeup Entry button is used, the system provides display prompting.

- Voice Prompting

  A guest can enter his or her own wakeup call request; however, the request can be entered *only* for the extension number where the call is originated.

  After the user dials the Automatic Wakeup FAC, the system generates voice prompts (through the use of a Speech Synthesizer circuit pack). These prompts tell the user when to enter information and what information is needed. The touch-tone buttons are used to enter the required information, and military and standard time are accepted. The user must dial the Automatic Wakeup FAC again to change or delete a wakeup request.

If invalid entries are made, a standard message is generated to notify the user of the error. The system then repeats the original prompt for input. If an invalid input occurs on the second try, the system informs the user to dial the attendant for assistance.

- Display Prompting

  Display prompting is provided to attendants, front desk users, and to other users with display-equipped voice terminals. Front desk users are administered with console permission COS and can perform the same actions as the attendant. Other users can enter a wakeup request only for the extension number where they are originating the call. The attendant presses the Automatic Wakeup Entry button to activate the feature. If the attendant is on an active call with a system user, the user's extension number will be displayed as the default extension by pressing the pound sign. If the displayed extension number is not the extension number of the user desiring the wakeup call, the attendant can change it. Display prompting continues until the attendant has entered all necessary information and the request for the wakeup call is confirmed.

  If a condition exists that the system does not accept the wakeup request, the system displays the reason for denial. Wakeup requests may be denied for one of the following reasons:

  — Too Soon — Indicates that the requested wakeup time is within the current five-minute wakeup interval.

  — System Full — Indicates that the maximum number of system wakeup calls has been reached.

  — Interval Full — Indicates that the maximum number of wakeup calls in any 15-minute interval has been reached.

  The attendant can change or cancel a wakeup call request at any time.

When the system places a wakeup call, one of the following will occur:

- Extension Is Busy — The wakeup call will be placed again later.
- No Answer — The system will apply ringing for 30 seconds. If the call is not answered, the system will try again later.
- Ringing Blockage — If four or more ports on the same Analog circuit pack are already ringing, the system will wait 16 seconds and try again. If the second attempt is blocked, the call is considered to have failed and the system will wait five minutes before trying again.

- Call Is Answered — When a wakeup call is answered, the guest hears music, a recorded announcement (from the Speech Synthesis circuit pack, the Audichron® Recorder/Announcer, the TN750B circuit pack, or silence, according to system administration.
- System Reset — Indicates that a system reset level 1 or system reset level 2 occurred while the system attempted to place the wakeup call. Calls affected by these conditions are treated as other wakeup attempts.

If a wakeup call was not completed because of a busy, no answer, ringing blockage, or system reset, the system attempts to place the call two more times at five-minute intervals. If the call is not completed after the three attempts, the system leaves an LWC message for the designated extension.

A special extension, called the Wakeup Messages Extension, must be administered exclusively for receiving failed wakeup call LWC messages. When such a message is retrieved, the display shows the date, time, and extension number for the failed wakeup call attempt.

An Automatic Message Waiting (AMW) button and associated lamp can be assigned to attendant consoles or front desk terminals. The number associated with the button can be the Wakeup Messages extension. The AMW lamp lights when a failed wakeup message is waiting. The user may retrieve the message by invoking Coverage Message Retrieval on the wakeup message extension. When the button associated with the AMW lamp is pressed, the console or terminal is placed in the Coverage Retrieval mode. The user then retrieves the failed wakeup call attempt messages. Only attendants and specified voice terminal users can retrieve and delete the failed wakeup messages.

The system maintains an audit trail record of wakeup call activity for the past 24 hours. Not all records are saved for the past 24 hours. The wakeup call buffer can only hold a number of records equal to the maximum number of stations administrable on the switch. For example, if the switch is a G3V4s, a maximum of 200 stations is administrable, therefore, only 200 Automatic Wakeup records can be stored.

The System Manager can request that wakeup events be displayed at the Management Terminal, or printed at a designated printer. If the system has a journal printer, wakeup events are printed as they occur.

The audit trail record contains the following information:

- Type of event, such as:
  - Request — A new wakeup call request has been made.
  - Change — The time has been changed on an existing wakeup call request.
  - Cancel — A wakeup request has been canceled. This event can be caused by a user request, a front desk request, or a room check out.

- — Move To — The wakeup request for this room has been moved to another room. This event occurs when the PMS sends a room change message.
- — Move From — The wakeup request for another room has been moved from that room to this room. This event occurs when the PMS sends a room change message.
- — Move-Cancel — A wakeup request from another room has replaced the request for this room. This event occurs when the PMS sends a room change request.
- — Swap — A room swap has occurred and at least one of the rooms had a wakeup request. Wakeup calls are swapped when a room swap is performed. A journal entry is made for each room. If the room has received a wakeup call as the result of the swap, the time of the call is provided in the entry. If the room has lost a wakeup call as the result of the swap (and has not received another), the time is not present in the entry.
- — Completed — The wakeup call was completed successfully.
- — Not Completed — The wakeup call failed (not answered, busy, and so on.)
- — Skip — The wakeup call was skipped. This event occurs if the system time is advanced past the requested time of a wakeup call.

- Time of the event
- Extension number receiving the call
- Time of the wakeup request
- Extension number (or 0 for the attendant) where the event took place
- Number of call attempts that were placed
- An indication of why a wakeup call attempt failed

In addition, all wakeup time changes are recorded. This record shows the original time requested and the changed time. The audit trail record is not backed up and all wakeup data is lost if a system failure occurs.

The following reports can be scheduled for printing on a daily basis:

- Wakeup Activity Report — This report summarizes the wakeup activity for each extension that had any wakeup activity over the past 24 hours.
- Wakeup Summary Report — This report gives an hour-by-hour summary of the number of scheduled wakeup calls, the number of wakeup calls that were completed, and a list of extensions to which wakeup calls were attempted but not completed during that hour. The report covers all Automatic Wakeup events for each hour over the past 24-hour period.

G3V3 offers an additional announcement type, integrated. This type uses the TN750B or TN750 C circuit pack, which is available on previous releases for announcement types other than Automatic Wakeup. This type of announcement operated similarly to the Audichron adjunct; however, it offers a less costly solution to voice-synthesized announcements because it is part of the existing system. The integrated type enables several new features, including multiple announcements.

The ability to use multiple announcements provides the benefit of having customized announcements for different purposes. For example, for the convenience of your foreign guests, you can use wakeup announcements in a variety of languages. Additionally, because you are able to use a vector directory number (VDN), as a wakeup announcement extension. With VDNs and multiple announcements, a hotel administrator can choose as the announcement extension a VDN that reaches one announcement if the system clock is less than 12:00 and another if the system clock is greater than 12:00. The hotel guest could then hear `good morning` before noon and `good evening` after noon. Or, a business customer could choose as the announcement extension a VDN that points to an extension assigned to a Quorum Bridge, with the ‘‘wakeup’’ time being a scheduled teleconference time. When the wakeup call comes through, the customer is automatically connected to the teleconference bridge.

Also, you may administer this type of announcement to repeat. To enable repeating announcements, enter the announcement type ***integ-rep*** on the ‘Recorded Announcement’ form. With the repeating integrated message functionality, the announcement keeps repeating from when the first guest (of a group of guests receiving the same wakeup announcement at the same time) goes off-hook until the last guest goes on-hook.

If the announcement type is either an externally-recorded announcement or is integrated-repeating, you may administer the wakeup call queue for ‘‘barge-in.’’ Barge-in means that the guest receiving the wakeup call hears the announcement as soon as he or she is off-hook, even if the announcement is not at the beginning — that is, the wakeup call, when it is answered by the guest, ‘‘barges-in’’ to the announcement. This provides the capability of many users being bridged onto the same announcement port, eliminating the need for a separate port for each wakeup call. for additional information, see the "Recorded Announcement" feature.

Considerations

The "Automatic Wakeup" feature lessens the attendant’s work load since each user can activate the feature and request his or her own wakeup call. In addition, the system places the wakeup calls automatically.

The voice and display prompting assures the user that his or her request is confirmed. Also, the audit trail record information assures the staff that users will not miss their wakeup calls.

The following items should also be considered:

- Verification of Wakeup Announcements — A special access code can be administered for the attendant or front desk users to verify that wakeup announcements are operating properly.
- If an announcement resource is not available or is not operating properly when a wakeup call is placed, the user still receives the call but hears silence instead of an announcement.
- A time change entered at Management Terminal may cause some calls to be skipped. Moving the system clock ahead will skip the calls scheduled during the skipped interval; moving the clock back a maximum of one and one half hours has no effect on wakeup calls. If an initial call attempt was made before the time change, the retry call attempts will still be placed.
- Once a wakeup call request has been completed, skipped, or failed after three attempts, the request is deleted from the system. A record of the call request is, however, maintained in the audit trail record.
- One wakeup request at any one time is allowed per extension number.
- Wakeup requests may be entered from five minutes to 23 hours and 55 minutes in advance of the actual wakeup call. If the requested wakeup hour entered is 0 or from 13 to 23, the system assumes military time. If the requested wakeup hour is from 1 to 12, the system prompts the user to enter 2 (for a.m.) or 7 (for p.m.) to indicate morning or evening.
- Up to 10 attendant consoles and/or front desk terminals may be in the wakeup display mode at any one time.
- The number of available speech synthesis ports is the only limit to the number of users entering wakeup call requests at the same time. If overflow occurs, such calls are routed to the attendant or to the specially administered Wakeup VoiceSynthesis-Unavailable Extension.

  Speech synthesis for Automatic Wakeup is available with US speech synthesis boards (United Kingdom customers can use the US board for speech synthesis if the American accent is acceptable for their application). Italian and the United Kingdom boards do not support Automatic Wakeup.
- Wakeup call attempts are not rerouted, forwarded, or sent to coverage.
- An RS-Alert lamp on the attendant console or a station would enable you to know when a reset system levels three to five has occurred. Resets at these levels will erase any AWU requests from the system. When the RS-Alert lamp is lit, check the AWU journal printer to see if there were any outstanding requests. During a reset system 3, the AWU journal printer will notify you of the reset only if there were wakeup requests in the system when the reset occurred. This report will include the time of the reset. Reset system 4 and reset system 5 are always journaled, since the system cannot determine whether or not there were wakeup requests prior to the reset.

Interactions

The following features interact with the "Automatic Wakeup" feature.

- Attendant or Voice Terminal Display

  If the console or terminal is in the Automatic Wakeup mode and the user presses another display mode button, the Wakeup mode is aborted and the wakeup request is not entered, changed, or deleted.

- Do Not Disturb

  If Do Not Disturb is active at a voice terminal, the "Automatic Wakeup" feature deactivates Do Not Disturb for that terminal, and then the system places the wakeup call.

- PMS Interface

  A Check-Out request will cancel an active wakeup call request for the guest room. Also, a Room Change/Room Swap request through the PMS will cause a wakeup request to be changed or swapped.

- Voice Synthesis Board

  Auto Wakeup competes with the following features for use of the Speech Synthesis board.

  — Do Not Disturb

  — Voice Message Retrieval

  — Visually Impaired Attendant Service

Administration

The "Automatic Wakeup" feature is administered by the System Manager. The following items require administration:

- Wakeup call announcement type — Choose one of the following: music, external recorded announcement, voice synthesis announcement, integrated, multiple integrated, or silence.
- Length of time to leave a voice terminal connected to the announcement.
- Extension number to receive LWC messages for failed wakeup call attempts.
- Extension number to receive wakeup call attempts when voice synthesis prompting is not available.
- Automatic Wakeup Entry button (per attendant console or display-equipped terminal).
- Feature access code for voice prompting.
- Special access code for the attendant to verify that speech synthesis announcements are operating properly.

- Hospitality-Related System Parameters assignments:
  - Extension number for the wakeup log printer, if a journal printer is used
  - The time for the scheduled Wakeup Activity Report to be printed
  - The time for the scheduled Wakeup Summary Report to be printed
  - Integrated Announcement Extension if the Announcement Type is integrated
  - Default Announcement Extension if the Announcement Type is multiple-integrated

(G3V3 and later releases) If you are going to use the integrated announcement types, you must administer the following:

- The “G3V3 Options” field on the ‘System-Parameters Customer-Options’ form (contact your AT&T support representative for assistance).
- The “G3V3 Enhanced Hospitality” field on the ‘System-Parameters Customer-Options’ form (contact your AT&T support representative for assistance).
- The extensions used for the announcements, administered on the ‘Announcements/Audio Sources’ form.
- Repeating announcements and barge-in queues, administered on the ‘Announcements/Audio Sources’ form.

Hardware and Software Requirements

If voice prompting is used, a TN725 Speech Synthesizer circuit pack is required. Each circuit pack has four ports to provide voice prompting. If speech synthesis is selected for wakeup call announcements, two ports must be reserved for wakeup announcements.

If recorded announcements are used, a model HQD 614B Recorder/Announcer manufactured by the Audichron Company is required. Each Recorder/Announcer requires four auxiliary trunk ports which must be on the same TN763B circuit pack.

With hospitality features such as "Automatic Wakeup", it is recommended that a journal printer be used with switch configurations larger than 800 stations. The journal printer requires an MPDM and a port on a Digital Line circuit pack or an ADU and a port on a Data Line circuit pack.

No additional software is required.

Integrated announcement types require the TN750B recorded announcements circuit pack. Multiple integrated announcements require the TN750C circuit packs. See the "Recorded Announcement" feature for details.

Basic Call Management System (BCMS)

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides real-time and historical reports to assist customers in managing individual agents, ACD splits (hunt groups), VDNs, and trunk groups. These reports are a subset of those available on the CMS adjunct. BCMS reports can be accessed and displayed on the Management Terminal or printed on demand on the printer associated with the Management Terminal. In addition, the historical reports can be scheduled to print on the system printer.

The BCMS report feature collects and displays information pertaining to individual agents (based on the agent's extension), ACD splits, trunk groups, and VDNs. Data is stored by hour or half hour for 25 time intervals (includes current time interval). Daily summary data are also calculated and stored for seven days.

The following reports are available with the BCMS:

- Real Time Reports
  - Split Status
  - System Status
  - VDN Status
- Historical Reports
  - Agent
  - Agent Summary
  - Split
  - Split Summary
  - Trunk Group
  - Trunk Group Summary
  - VDN
  - VDN Summary

The reports can be displayed and/or printed both locally and remotely. Locally, the reports can be accessed by the ACD administrator from the Management Terminal. Customers with multiple premises may wish to centralize the measurements data evaluation and access the switch data remotely. Reports can also be scheduled to print on the Report Scheduler system printer.

An example of each BCMS report follows, along with a brief description of the data in the report. More detailed information on these reports can be found in *DEFINITY Communications System Generic 3 Basic Call Management System (BCMS) Operations*, 555-230-704.

VuStats, available in G3V3 and later releases, provides call measurements for voice terminals equipped with displays. This feature can be used in conjunction with R3 CMS or BCMS to supplement the number of individuals receiving real-time information about call performance. This feature can also be used without BCMS. (Refer to ‘‘Considerations’’ at the end of this section.)

NOTE:
The screens and calculations for some of the field values differ between G3V1, V2, V3, and V4.

Also, if the EAS feature is optioned on the switch, the word ‘‘split’’ on menus and screens actually prints as ‘‘skill’’ since the concept of skills and skill groups replaces splits in an EAS environment.

Acceptable Service Level

Before using BCMS, you must understand the concept of Acceptable Service Level and then set the “Acceptable Service Level” field on various forms.

Acceptable Service Level is the desired time to answer for a given VDN or hunt group. Timing for a call begins when the call encounters a VDN or enters a hunt group queue. If the number of seconds to answer the call is equal to or less than the administered acceptable service level for the VDN or hunt group, the call is recorded as acceptable.

Percent Within Service Level

A service level can be administered for each hunt group or VDN, if the customer option has been set to ‘‘y’’ and if the hunt group or VDN is measured by BCMS. The service level is the amount of time (number of seconds) allowed the switch to answer calls (from 0 to 9999).

To calculate the percentage of calls within the acceptable service level, BCMS divides the number of acceptable calls by the calls offered.

For hunt groups, BCMS calculates the Percent Within Service Level as follows:

$$\%\ IN\ SERV\ LEVL = \frac{accepted * 100}{ACDcalls + abandons + outflows + dequeued}$$

where

accepted — Is the number of calls answered for which the queue time was less than or equal to the administered service level for the split.

dequeued — Is the number of calls that encountered the split's queue, but were NOT answered, abandoned, or outflowed. This occurs with multiple split queuing.

For VDNs, BCMS calculates the Percent Within Service Level as follows:

$$\% \ IN \ SERV \ LEVL \ = \ \frac{accepted \ * \ 100}{calls \ offered}$$

where

accepted — Is the number of answered calls (*num ans*) for which the answer time was less than or equal to the administered service level for the VDN. *num ans* here refers to the data item on the form of the same name.

calls offered — Is the total number of completed calls that accessed the VDN during the interval.

BCMS Real-Time Reports

BCMS provides three real-time reports:

- **BCMS Split Status Report**
- **BCMS System Status Report**
- **BCMS VDN Status Report**

The BCMS Split Status Report provides the current (real-time) status and cumulative measurement data for those agents assigned to the split you specify. The BCMS System Status Report provides current (real-time) status information for either all BCMS splits or selected splits. The BCMS VDN Status Report provides the current (real-time) status and cumulative measurement data for VDNs monitored by BCMS.

You may generate these reports using the `monitor` command. These reports display data accrued since the last interval boundary. The time intervals may be in one-hour or half-hour increments. (To select the desired increment, access the Feature-Related System Parameters screen and enter `hour` or `half-hour` in the "Measurement Interval" field.

There are three `monitor` commands, one to print each real-time report:

- `bcms split`
- `bcms system`
- `bcms vdn`

The `bcms split` command generates the BCMS Split Status Report. The `bcms system` command generates the BCMS System Status Report. The `bcms vdn` command generates the BCMS VDN Status Report.

Whenever a status report is displayed on the G3 Management Terminal, it is updated automatically approximately every 30 seconds. You can immediately update the on-screen status report by pressing UPDATE. To cancel the `monitor` command and return to the command prompt, press CANCEL. If the status report consists of more than one page, press NEXTPAGE to display any subsequent pages and PREVPAGE to display any previous pages.

If you incorrectly enter the command, or if the qualifier is not applicable or cannot be measured, a descriptive error message appears on the message line, located on the bottom of the screen. Usually, the error message descriptions provide enough information about the problem so that you will not need to research it. However, if you require more information about the error message, press HELP. Some examples of error messages are listed below:

- *"??" invalid report type for specified time or day*
- *?? number of BCMS measured agents exceeds maximum*
- *Split not measured by BCMS*

BCMS Split Status Report

The BCMS Split Status Report provides the current (real-time) status and cumulative measurement data for those agents assigned to the split you specify. This report displays data accrued since the last interval boundary. For example, if the interval is set for hourly, and you issue the command to display the BCMS Split Status Report at 11:10 a.m., the report displays the data accrued since 11:00 a.m. Although this report is updated approximately every 30 seconds, you can immediately update the information on the screen by pressing UPDATE. At the beginning of the next interval, the report resets. Screen 3-8 shows the BCMS Split Status Report.

```
monitor bcms split 30
                         BCMS SPLIT (AGENT) STATUS

        Split: 30                              Date: 12:13 pm MON MAY 15, 1995
   Split Name: headquarters
Calls Waiting:    5                         Acceptable Service Level: xxx
  Oldest Call:  1:39                          % Within Service Level: xxx

Staffed: 7   Avail: 1   ACD: 1   ACW: 1   AUX: 1   Extn Calls: 2   Other: 1

                                                    ACD    EXT IN  EXT OUT
AGENT NAME        LOGIN ID   EXT    STATE      TIME  CALLS  CALLS   CALLS

Agent 1            32191     12345  Avail    12:00      0       0        0
Agent 2            32192     12346  ACD      12:04      1       0        0
Agent 3            32193     12347  ACW      12:12      3       0        0
Agent 4            32194     12348  AUX      11:30      0       0        0
Agent 5            32195     12349  Ext In   12:08      1       2        0
Agent 6            32196     12350  Ext Out  12:10      0       0        1
Agent 7            32197     12351  Other    11:58      0       0        0
$                  32198     12352  INIT     00:00      0       0        0
```

Screen 3-8. BCMS Split Status Report Screen

* An asterisk precedes the ‘‘Call Waiting’’ field if any of the calls are Direct Agent calls.

& The ‘‘LOGIN ID’’ column is empty if the BCMS login system parameter is set to ‘‘no.’’

$ If name is not administered, this column is blank for the agent.

"'Split" is displayed as "Skill" when EAS is optioned.

Report Headers, Abbreviations, and Their Definitions

The header information at the top of each page includes the command entered to generate the report, the page number and the total number of pages in the report, the title of the report, and the time and date the report was generated. If there are more than nine agents in the split, the remaining agent information appears on subsequent pages.

Split — The split number specified with the command line.

NOTE:
With BCMS, splits do not have to be numbered from 1, and split numbers do not have to be consecutive.

Split Name — The administered name of the split. This name usually describes the purpose or service of the split (for example, sales, service, or help line). If no name exists, BCMS displays the split extension (for example, EXT 65222).

NOTE:
The split name is limited to a maximum of 11 characters. If you enter more than 11 characters, the additional characters are not printed on the System Printer.

Calls Waiting — The number of calls currently queued and calls ringing at an agent's phone. If any of the calls in the queue are Direct Agent calls, an asterisk appears before the value in this field. The Glossary describes the Direct Agent feature.

Oldest Call — The number of minutes and seconds that the oldest call in queue has been waiting to be answered. This includes calls ringing at an agent's phone.

Acceptable Service Level — The desired time to answer for a given hunt group or VDN. Timing for a call begins when the call enters the hunt group queue.

% Within Service Level — The percentage of calls answered within the administered service level. This field is blank if no calls have been recorded for this time interval or if there is no ''Acceptable Service Level'' administered on the Hunt Group form.

Staffed — The number of agents currently logged into the split.

Avail — The number of agents in this split currently available to receive an ACD call. In order to be counted as being available, agents must either be in the Auto-In or Manual-In work mode. Refer to the Glossary for a description of work modes (or ACD work modes). If the agent is on another split's call or is performing After Call Work for another split, the agent is not considered available and is not recorded here. If a call is ringing at the agent's phone or a call is on hold, the agent is not considered available unless "Multiple Call Handling" is active and the agent selects AI/MI with a call on hold.

ACD — The number of agents who are currently on an ACD call for this split. This value also includes Direct Agent calls and those agents who are currently on ACD calls that flowed in from another split.

ACW — The number of agents in this split who are currently in ACW mode for this split. Refer to the Glossary for a description of after call work (ACW) mode. If an agent is in ACW mode for another split, the agent is included in the Other state count for this split. Also, if an agent is on a call while in ACW mode, the agent appears in the Extn state count, and not in the ACW state.

AUX — The number of agents in this split who are currently in the AUX work mode for this split. If an agent is answering a call from another split or is in ACW work mode for another split, that agent is not considered in AUX work mode for this split and is not included in this number. The agent is included in the Other state count.

Extn — The number of agents in this split who are currently on non-ACD calls. These non-ACD calls may be either incoming (direct to the extension) or outgoing (direct from the extension). Those agents receiving or making extension calls while in Avail, ACW, or AUX work mode is recorded as being on extension calls.

Other — The number of agents in this split who:

- Are on a call from another split
- Are in ACW work mode for another split
- Have placed a call on HOLD and made no other state selections
- Have a call ringing at their voice terminals
- Are dialing a number (to place a call or activate a feature)

All of the agents in the Other state are unavailable for ACD calls.

AGENT — The name of the agent. Generally, this is the agent's first or last name. However, if no name is administered on the station form, this field is left blank. When the field is blank, the data can be identified by the extension.

LOGIN ID — The BCMS login ID(s) (taken from the BCMS Login ID form or EAS Login form) for which you requested the report. This column does not appear if BCMS logins are not optioned.

EXT — The 2-, 3-, 4-, or 5-digit extension number for the agent.

STATE — The current work state for the agent. Possible work states are Avail, ACD, ACW, AUX, Extn, and Other. (The sum of the time the agent spends in the possible work states is the agent's ''staffed'' time.) Unstaffed agents do not appear on the report. When the system time is changed, agents are in the INIT state. Each agent remains in the INIT state until he or she takes a call or pushes a work button.

NOTE:
Refer to the Glossary for a description of the term "work state".

TIME — The 24-hour clock time that the agent entered this work state.

ACD CALLS — The number of ACD calls that the agent has completed since the beginning of the current interval. This value includes any calls that flowed in from other splits. (Calls in process are not counted until they are completed.)

EXT IN CALLS — The number of non-ACD calls that the agent has received (incoming) since the beginning of the current interval. (Calls in process are not counted until they are completed.) The maximum value is 255.

EXT OUT CALLS — The number of non-ACD calls that the agent has made (outgoing) since the beginning of the current interval. (Calls in process are not counted until they are completed.) The maximum value is 255.

BCMS System Status Report

The BCMS System Status Report provides current (real-time) status information for either all BCMS splits or selected BCMS splits. This report displays data accrued since the last interval boundary. For example, if the interval is set to hour, and you issue the command to display the BCMS System Status Report at 11:10 a.m., the report displays the data accrued since 11:00 a.m. Although this report is updated approximately every 30 seconds, you can immediately update the information on the screen by pressing UPDATE. This report is reset at the beginning of the time interval (for example, hour or half-hour). Screen 3-9 shows the BCMS System Status Report.

NOTE:
When analyzing this report, keep the following things in mind:

- All averages are for completed calls only.
- A completed call may span more than one time interval. ACD calls that are in process (have not terminated) are counted in the time interval in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time interval, but terminates in the 11:00 to 12:00 time interval, the data for this call is counted in the 11:00 to 12:00 time interval.
- Asterisks indicate that the maximum for the associated field has been exceeded.

```
 monitor bcms system
                           BCMS SYSTEM STATUS
                                         Date:  12:53 MON MAY 15, 1995

                           AVG                 AVG          AVG   AVG   % IN
             CALLS OLDEST SPEED AVAIL ABAND ABAND ACD     TALK  AFTER SERV
SPLIT NAME   WAIT  CALL   ANS   AGENT CALLS TIME  CALLS  TIME  CALL  LEVL

Service          3   1:03   :45     0     3   :30    20  2:30   1:25   85
EXT 4000         5    :33   :15     0    11   :45    36  1:32    :35   91
```

Screen 3-9. BCMS System Status Report Screen

& Split name is not administered — default is EXT xxxx, where xxxx is the extension administered for the split.

"SPLIT" is displayed as "SKILL" when EAS is optioned.

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number and the total number of pages in the report, the title of the report, and the time and date the report was generated. If more than 14 splits are being measured by BCMS, the remaining splits are displayed on multiple pages.

SPLIT NAME — The name of the split (for example, sales, service, or help line). If no name exists, the split extension (for example, EXT 12345) is displayed.

CALLS WAIT — The number of calls in the split's queue that are currently waiting to be answered and calls ringing at an agent's phone. If any of the calls in the queue are Direct Agent calls, an asterisk appears before this field. Consult the Glossary for a description of the Direct Agent feature.

OLDEST CALL — The number of minutes and seconds the oldest call in queue has been waiting to be answered. This includes calls ringing at an agent's phone.

AVG SPEED ANS — The average amount of time it takes before the calls are being answered. This value includes time waiting in the queue and time ringing at the agent's voice terminal. The calculation is:

$$AVG\ SPEED\ ANS = \frac{Sum\ of\ Each\ Completed\ Call's\ Time\ In\ Queue\ +\ Time\ Ringing}{The\ Total\ Number\ of\ ACD\ Calls\ Answered}$$

NOTE:
Keep the following things in mind:

- Calls that flow in from other split(s) do not include "time in queue" from the other splits in this calculation. Also, the AVG SPEED ANS does not include time spent listening to a forced first announcement.
- A completed call may span more than one time period. ACD calls that are in process (have not terminated) are counted in the time period in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time period, but terminates in the 11:00 to 12:00 time period, the data for this call is counted in the 11:00 to 12:00 time period.

- Asterisks indicate that the maximum for the associated field has been exceeded.

AVAIL AGENT — The number of agents in this split who are currently available to receive an ACD call directed to this split.

ABAND CALLS — The total number of ACD calls that have hung up while waiting to be answered. This includes those calls that have abandoned while in queue or while ringing. Calls that are not queued (for example, because the queue is full, the caller receives a forced first announcement and abandons during the announcement, or no agents are staffed) are not counted as abandoned for the hunt group.

AVG ABAND TIME — The average time before an ACD call abandons. This does not include any time spent in another split's queue before intraflowing to this split. The calculation is:

$$AVG\ ABAND\ TIME = \frac{Total\ Abandon\ Time}{Total\ Number\ of\ Abandoned\ Calls}$$

NOTE:
This value does not include time spent listening to a forced first announcement or calls that "abandon" while listening to a forced first announcement.

ACD CALLS — The number of ACD calls completed during the current interval. This number also includes those calls that flow in from other splits.

AVG TALK TIME — The average duration of ACD calls for each split. This calculation includes the time each agent spent talking but does not include ring time at an agent's voice terminal. The calculation is:

$$AVG\ TALK\ TIME = \frac{Total\ ACD\ Talk\ Time}{Total\ Number\ of\ ACD\ Calls\ Answered}$$

AVG AFTER CALL — the average ACW time for call-related ACW time completed by agents in this split during this time interval. Call-related ACW is the time that occurs immediately after an ACD call (that is, when an agent was in Manual mode and an ACD call ended, or when the agent presses the ACW button during an ACD call). AVG AFTER CALL does not include time spent on direct incoming or outgoing calls while in ACW or time that immediately follows an EXTN call. The calculation is:

$$AVG\ AFTER\ CALL = \frac{Total\ Call\ Related\ ACW\ Time}{Number\ of\ Call\ Related\ ACW\ Sessions}$$

NOTE:
The average is for ACW sessions, which may not correspond to the number of ACD calls either because some ACD calls did not have ACW time or because the call was recorded in another interval.

% IN SERV LEVL — The percentage of calls answered with in the administered service level for this split. Calculation is based on the following:

$$\%\ IN\ SERV\ LEVL = \frac{Accepted * 100}{ACD\ calls + Abandons + Outflows + dequeued}$$

where

accepted is calls answered whose queue time was less than or equal to the administered service level for the split.

dequeued is a call that encountered the split's queue, but which was NOT answered, abandoned, or outflowed. This occurs with multiple split queuing.

BCMS VDN Status Report

The VDN Status Report gives real-time status information for internally measured VDNs. You can monitor up to 99 VDNs at one time, however; the report can display up to 13 VDNs on a single page. Therefore, if you are monitoring 99 VDNs, the report is 6 pages long. You must specify the extensions of the VDNs you want the system to monitor. You can specify the extension in a list or in a range format. For example, `monitor bcms vdn 12345 12346 12350-12359`.

```
monitor bcms vdn 12345-12349

                      BCMS VECTOR DIRECTORY NUMBER STATUS
                                             Date: 15:30 Mon May 15, 1995

                              AVG           AVG    AVG                CALLS % IN
           CALLS OLDEST ACD   SPEED  ABAND ABAND  TALK/ CONN   FLOW  BUSY/ SERV
VDN NAME   WAIT  CALL   CALLS ANS    CALLS TIME   HOLD  CALLS  OUT   DISC  LEVL

knives         5    :25    50   :39      5   :45   2:30     0     0     24   91
EXT 12346      0    :00     0   :00      0   :00    :00     0     0      0    0
```

Screen 3-10. BCMS Vector Directory Number Status

* Indicates that the VDN name is not administered for the VDN; default extension is as shown.

Report Headers, Abbreviations, and Their Definitions

Date — The current date and time (updated every 30 seconds or when the update key is pressed).

VDN NAME — The name of the VDN being reported. If the VDN does not have a name administered, this field displays “EXT XXXXX” where "“XXXXX’’ is the VDN’s extension.

CALLS WAIT — The number of calls that encountered this VDN and have not been answered, abandoned, outflowed, or forced busy/disc. Includes calls in queues in vector processing, and ringing at an agent’s station.

OLDEST CALL — The time the oldest call currently waiting has waited in the VDN. Timing starts when the call enters the VDN.

ACD CALLS — The number of completed ACD calls answered in a BCMS-measured split. The split may have been reached via the queue-to-main, check backup, route-to, messaging split, or adjunct routing commands. Includes Direct Agent calls.

AVG SPEED ANS — The average speed of answer for ACD and connect calls that have completed for this VDN during the current period. This includes the time in vector processing, in a split’s queue, and time ringing. The calculation is:

$$AVG\ SPEED\ ANS\ =\ \frac{Total\ Answer\ Time}{Total\ ACD\ Calls\ +\ Total\ CONNect\ CALLS}$$

NOTE:
Answer time for a call is recorded when the call ends. If a call originates in interval x, is answered in interval y, and ends in interval z, the associated answer and talk times are recorded in interval z.

ABAND CALLS — The number of calls to this VDN that have abandoned before being answered during the current period. This includes VDN calls that were routed to an attendant, station, or announcement, and abandoned before being answered.

AVG ABAND TIME — The average time abandoned calls waited before abandoning during the current period. The calculation is:

$$AVG\ ABAND\ TIME\ =\ \frac{Total\ Abandon\ Time}{Total\ Calls\ Abandoned}$$

AVG TALK/HOLD — The average talk time for ACD calls completed by this VDN during the current period. This does not include ring time, but it does include any time the caller spent on Hold. The calculation is:

$$AVG\ TALK/HOLD\ =\ \frac{Total\ Talk\ Time}{ACD\ Calls}$$

CONN CALLS — The number of calls that were routed to a station (agent or non-ACD), attendant, or announcement, and were answered there.

FLOW OUT — The number of calls that were routed to another VDN or to a trunk, including successful look-ahead attempts.

CALLS BUSY/DISC — The number of calls that encountered a busy or disconnect step (and the announcement ends).

% IN SERV LEVL — The percent of calls offered that completed and were answered within the acceptable service level defined on the VDN form. The calculation is:

$$\%\ SERV\ LEVL\ =\ \frac{Acceptable\ *\ 100}{Offered}$$

‘‘Offered’’ is defined as:

acdcalls + flowout calls + abandoned + connect + busy/disc

‘‘Acceptable’’ is the number of ACD and CONNect calls that were answered within the administered service level. This field is blank if no calls were recorded for this time interval. This field is also blank if no ‘‘Acceptable Service Level’’ has been administered on the VDN form.

BCMS Historical Reports

BCMS provides eight historical reports. These reports give you information for an interval of time. You can print the reports for a period time measured in minutes or hours, or a period of time measured in days. The BCMS historical reports are:

- Agent Report
- Agent Summary Report
- Split Report
- Split Summary Report
- Trunk Group Report
- Trunk Group Summary Report
- VDN Report
- VDN Summary Report

You are able to print the historical reports using the list commands.

The `list` commands are used to display historical information for agents, splits, system, trunk groups, and VDNs. There are eight secondary list commands.

- `bcms agent`
- `bcms summary agent`
- `bcms split`
- `bcms summary split`
- `bcms trunk`
- `bcms summary trunk`
- `bcms vdn`
- `bcms summary vdn`

With these commands, you can specify:

- Whether you want the data in the reports to be displayed in hourly/half-hourly or daily intervals
- The times or days for which you wish to see data
- The system to immediately display the report on your terminal
- The system to print the report.

  If you include `print` at the end of the command, the system will immediately print the report to a slaved printer. If you include `schedule` at the end of the command, the system will allow you to schedule the report to print to the system printer immediately (immediate), at a later time (deferred), or routinely at specified times (scheduled).

NOTE:
Time interval data may be collected in half-hour or one-hour increments. (To select the desired increment, access the Feature-Related System Parameters screen and enter `half-hour` or `hour` in the “Measurement Interval” field.) The switch stores time interval data in a time database which holds a maximum of 25 intervals. Data for the 26th interval overwrites the first interval in the time database (and so on). Therefore, if the half-hour option is selected, care should be exercised to ensure that time interval reports are run while the data for the desired interval is still available in the time database. For example, if you select the half-hour option, print the report twice daily to ensure that you do not lose information.

BCMS Agent Report

The BCMS Agent Report provides traffic information for the specified agent. Depending on specifics from the command line, the information may be displayed as either a time interval or a daily summary. If neither ‘‘time’’ nor ‘‘day’’ is specified, ‘‘time’’ is the default. In this case, the report displays data accrued for the previous 24 time intervals (hour or half-hour), including data from the most recently completed time interval. To get information on the current time interval, you must use a `monitor bcms` command. Screen 3-11 shows the BCMS Agent Report — Hourly, and Screen 3-12 shows the BCMS Agent Report — Daily.

NOTE:
BCMS can track agents based on their phone numbers, or based on login IDs. If BCMS/VuStats Login IDs is optioned, BCMS tracks login IDs.

NOTE:
When analyzing this report, keep the following things in mind:

- All averages are for completed calls only.
- A completed call may span more than one time interval. ACD calls that are in process (have not terminated) are counted in the time interval in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time interval, but terminates in the 11:00 to 12:00 time interval, the data for this call is counted in the 11:00 to 12:00 time interval.
- Asterisks indicate that the maximum for the associated field has been exceeded.

```
 list bcms agent 4222 8:00
                                BCMS AGENT REPORT

Switch Name: Lab Model                         Date: 11:05 am MON MAY 15, 1995
      Agent: 4222
 Agent Name: s-jones

                     AVG    TOTAL   TOTAL   TOTAL         AVG   TOTAL     TOTAL
             ACD     TALK   AFTER   AVAIL   AUX/   EXTN   EXTN  TIME      HOLD
TIME         CALLS   TIME   CALL    TIME    OTHER  CALLS  TIME  STAFFED   TIME

 8:00- 9:00     10   1:15    7:30   25:00   10:40      1  4:00    60:00    :20
 9:00-10:00     18   1:40   18:00    4:20     :00      2  3:20    60:00   1:00
10:00-11:00     10   1:20    8:20   16:10     :00      0   :00    38:00    :10
----------- ----- ------ ------- ------- ------- ----- ----- -------- ------
SUMMARY         38   1:28   33:50   45:30   10:40      3  3:33   158:00   1:30
```

Screen 3-11. BCMS Agent Report — Hourly

NOTE:
4222 could be a login ID or an extension, depending on whether BCMS/VuStats Login IDs is administered.

```
 list bcms agent 4222 day 5/17
                                BCMS AGENT REPORT

Switch Name: Lab Model                         Date: 11:05 am MON MAY 15, 1995
      Agent: 4222
 Agent Name: s-jones

                     AVG    TOTAL   TOTAL   TOTAL         AVG   TOTAL     TOTAL
             ACD     TALK   AFTER   AVAIL   AUX/   EXTN   EXTN  TIME      HOLD
DAY          CALLS   TIME   CALL    TIME    OTHER  CALLS  TIME  STAFFED   TIME

 5/14/95       200   1:30  100:00   35:00   80:00     10  2:00   540:00   5:00
 5/13/95        38   1:28   34:12   45:30   10:40      3  3:33   158:00   1:30
----------- ----- ------ ------- ------- ------- ----- ----- -------- ------
SUMMARY        238   1:30  134:12   80:30   90:40     13  2:22   698:00   6:30
```

Screen 3-12. BCMS Agent Report — Daily

NOTE:
4222 could be a login ID or an extension.

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number of the report, the title of the report, and the time and date the report was generated. If this is a time report and there are more than 11 time intervals, this report is displayed on multiple pages. A daily summary report is displayed on the last page of the report.

AGENT NAME — The name of the agent. If no name is administered, the agent's extension is displayed in the form "EXT 65432."

TIME/DAY — The time or day interval specified in the command line.

Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the earliest time interval (either hour or half-hour). If no start time is given, the oldest time interval is the default. A stop time requires an associated start time. If no stop time is given, the last completed time interval (hour or half-hour) is the default. If no start time or stop time is given, the report displays data accrued for the previous 24 time intervals. If you specify ''day'' in the command and do not include a start day or stop day, the report displays data accrued for the previous six days and data accrued through the most recently completed interval (hour or half-hour).

ACD CALLS — The number of ACD calls answered by this agent for all splits during the reporting interval. This value includes calls that flowed in from other splits and Direct Agent calls.

AVG TALK TIME — The average duration of ACD calls for all splits the agent was logged into. This value includes time spent talking but does not include the amount of time the agent was holding an ACD call or ring time at the agent's voice terminal. The calculation is:

$$AVG\ TALK\ TIME = \frac{Total\ ACD\ Talk\ Time}{Total\ Number\ of\ ACD\ Calls\ Answered}$$

TOTAL AFTER CALL — The total amount of time that the agent spent in call-related or non-call-related ACW work states for all splits during the reporting interval. This does not include time spent on direct incoming or outgoing calls while in ACW. If an agent entered ACW in one interval, but ended ACW in another interval, the appropriate amount of ACW time is credited to each of the intervals.

TOTAL AVAIL TIME — The sum of the time that the agent was available to receive ACD calls during the current interval. During this time, the agent:

- Was in Auto-In or Manual-In work modes for at least one split

- Was not in ACW in any split
- Was not on any call or placing any call (unless MCH is active)
- Did not have ringing calls

TOTAL AUX/OTHER — The sum of the time that the agent has the AUX button pressed and is not doing anything else for any of the other splits (that is, the sum of the time that the agent is in AUX work mode for all splits). This value does not include time the agent spent on an EXTN call or in Manual-In, Auto-In, or ACW mode for another split. Note that if the agent was in Other for all logged-in splits that time is reflected here. For example, ringing calls can cause several seconds of AUX time to accrue.

For the agent report, any non-ACD call time is totaled in the AVG EXTN TIME column. Two points of contrast are:

1. The measurement TOTAL AUX/OTHER is time-interval based, rather than being call related. For example, assuming that the previously identified stipulations are met, then if the agent is in AUX from 9:55 to 10:05, five minutes is pegged in the 9:00 to 10:00 time interval and five minutes is pegged in the 10:00 to 11:00 time interval.
2. The measurement AVG EXTN TIME is call related. For example, if an agent is on a non-ACD call from 9:55 to 10:05, the call and ten minutes of EXTN time is pegged in the 10:00 to 11:00 time interval.

   Because the agent report includes some call-related items, the sum of all items for a given hour may not exactly equal 60 minutes.

EXTN CALLS — The total number of non-ACD incoming and outgoing calls for this agent during the reporting interval. Only those non-ACD calls that are originated and/or received while the agent is logged into at least one split are counted.

AVG EXTN TIME — The average amount of time that the agent spent on non-ACD calls while logged into at least one split during the reporting interval. This average does not include time when the agent was holding the EXTN call. The calculation is:

$$AVG\ EXTN\ TIME = \frac{Total\ Ext\ Time}{Total\ Number\ of\ Ext\ Calls}$$

TOTAL TIME STAFFED — The total time that the agent spent logged into at least one split during the reporting interval. Staff time is clocked for an agent who is in multiple splits as long as the agent is logged into any split. Concurrent times for each split are not totaled.

TOTAL HOLD TIME — The total time that the agent placed ACD calls on hold. This time is the "caller's hold time" and is independent of the state of the agent. TOTAL HOLD TIME does not include the hold time for non-ACD calls.

SUMMARY — The total of each of the columns that do not contain averages.

Columns that do contain averages are the total time divided by the number of calls.

BCMS Agent Summary Report

This report is similar to the BCMS Agent Report except that this report provides one line of data for each agent. You can specify one or more agents by entering agent IDs or extensions. You can not include more than 30 agents in a single report. An agent does not appear on the report if there is no data for that agent. If you specify that you want the report to include more than one time period, and the data exists for one or more, but not all of the specified times, the system uses the available data to calculate and display the one-line summary; the system does not identify which times are not included in the calculations.

NOTE:
BCMS can track agents based on their phone numbers, or based on login IDs. If BCMS/VuStats Login IDs is optioned, BCMS tracks login IDs.

```
 list bcms summary agent 4222-4224 4869 time 8:00-12:00

                           BCMS AGENT SUMMARY REPORT

Switch Name: Lab Model                           Date: 11:05 am MON MAY 15, 1995
       Time: 8:00-12:00

                     AVG    TOTAL   TOTAL   TOTAL         AVG   TOTAL    TOTAL
             ACD     TALK   AFTER   AVAIL   AUX/  EXTN    EXTN  TIME     HOLD
AGENT NAME   CALLS   TIME   CALL    TIME    OTHER CALLS   TIME  STAFFED  TIME

 s-jones        10   1:15    7:30   25:00   10:40     1   4:00    60:00    :20
 t-anderson     18   1:40   18:00    4:20     :00     2   3:20    60:00   1:00
 j-jacobsen     10   1:20    8:20   16:10     :00     0    :0     38:00    :10
-----------  ----- ------ ------- ------- ------- ----- ----- -------- ------
SUMMARY         38   1:28   33:50   45:30   10:40     3   3:33   158:00   1:30
```

Screen 3-13. BCMS Agent Summary Report — Hourly Summary

NOTE:
4222-4224 in the command line could be a login ID or an extension, depending on whether BCMS/VuStats Login IDs is administered.

```
 list bcms sum agent 4222-4223 4869 day 5/14

                          BCMS AGENT SUMMARY REPORT

Switch Name: Lab Model                          Date: 11:05 am MON MAY 15, 1995
        Day: 5/14

                     AVG    TOTAL   TOTAL   TOTAL         AVG   TOTAL    TOTAL
             ACD     TALK   AFTER   AVAIL   AUX/  EXTN    EXTN  TIME     HOLD
AGENT NAME   CALLS   TIME   CALL    TIME    OTHER CALLS   TIME  STAFFED  TIME

 s-jones        10   1:15    7:30   25:00   10:40     1  4:00    60:00    :20
 t-anderson     18   1:40   18:00    4:20     :00     2  3:20    60:00   1:00
 j-jacobsen     10   1:20    8:20   16:10     :00     0    :0    38:00    :10
-----------  ----- ------ ------- ------- ------- ----- ----- -------- ------
SUMMARY         38   1:28   33:50   45:30   10:40     3  3:33   158:00   1:30
```

Screen 3-14. BCMS Agent Summary Report — Daily Summary

NOTE:
4222-4224 in the command line could be a login ID or an extension, depending on whether BCMS/VuStats Login IDs is administered.

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number of the report, the title of the report, and the time and date the report was generated. If this is a time report and there are more than 11 time intervals, this report is displayed on multiple pages. A summary time is displayed on the last page of the report.

TIME/DAY — The time or day interval specified in the command line.

Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the earliest time interval (either hour or half-hour). If no start time is given, the most recent time interval is the default. A stop time requires an associated start time. If no stop time is given, only the start interval/day is used. If no start time or stop time is given, the most current interval/day is used. If you specify ‘‘day’’ in the command and do not include a start day or stop day, the report displays data for the current day accrued through the most recently completed interval (hour or half-hour).

AGENT NAME — The name of the agent. If no name is administered, the agent’s extension is displayed in the form “EXT 65432.’’

ACD CALLS — The number of ACD calls answered by this agent for all splits during the reporting interval. This value includes calls that flowed in from other splits and Direct Agent calls.

AVG TALK TIME — The average duration of ACD calls for all splits the agent was logged into. This value includes time spent talking but does not include the amount of time the agent was holding an ACD call or ring time at the agent's voice terminal. The calculation is:

$$AVG\ TALK\ TIME = \frac{Total\ ACD\ Talk\ Time}{Total\ Number\ of\ ACD\ Calls\ Answered}$$

TOTAL AFTER CALL — The total amount of time that the agent spent in call-related or non-call-related ACW work states for all splits during the reporting interval. This does not include time spent on direct incoming or outgoing calls while in ACW. If an agent entered ACW in one interval, but ended ACW in another interval, the appropriate amount of ACW time is credited to each of the intervals.

TOTAL AVAIL TIME — The sum of the time that the agent was available to receive ACD calls during the current interval. During this time, the agent:

- Was in Auto-In or Manual-In work modes for at least one split
- Was not in ACW in any split
- Was not on any call or placing any call
- Did not have ringing calls

TOTAL AUX/OTHER — The sum of the time that the agent has the AUX button pressed and is not doing anything else for any of the other splits (that is, the sum of the time that the agent is in AUX work mode for all splits). This value does not include time the agent spent on an EXTN call or in Manual-In, Auto-In, or ACW mode for another split. Note that if the agent was in Other for all logged-in splits that time is reflected here. For example, ringing calls can cause several seconds of AUX time to accrue.

For the agent report, any non-ACD call time is totaled in the AVG EXTN TIME column. Two points of contrast are:

1. The measurement TOTAL AUX/OTHER is time-interval based, rather than being call related. For example, assuming that the previously identified stipulations are met, then if the agent is in AUX from 9:55 to 10:05, five minutes is pegged in the 9:00 to 10:00 time interval and five minutes is pegged in the 10:00 to 11:00 time interval.
2. The measurement AVG EXTN TIME is call related. For example, if an agent is on a non-ACD call from 9:55 to 10:05, the call and ten minutes of EXTN time is pegged in the 10:00 to 11:00 time interval.

Because the agent report includes some call-related items, the sum of all items for a given hour cannot exactly equal 60 minutes.

EXTN CALLS — The total number of non-ACD incoming and outgoing calls for this agent during the reporting interval. Only those non-ACD calls that are originated and/or received while the agent is logged into at least one split are counted.

AVG EXTN TIME — The average amount of time that the agent spent on non-ACD calls while logged into at least one split during the reporting interval. This average does not include time when the agent was holding the EXTN call. The calculation is:

$$AVG\ EXTN\ TIME = \frac{Total\ Ext\ Time}{Total\ Number\ of\ Ext\ Calls}$$

TOTAL TIME STAFFED — The total time that the agent spent logged into at least one split during the reporting interval. Staff time is clocked for an agent who is in multiple splits as long as the agent is logged into any split. Concurrent times for each split are not totaled.

TOTAL HOLD TIME — The total time that the agent placed ACD calls on hold. This time is the ‘‘caller’s hold time’’ and is independent of the state of the agent. TOTAL HOLD TIME does not include the hold time for non-ACD calls.

SUMMARY — The total of each of the columns that do not contain averages. Columns that do contain averages are the total time divide by the number of calls.

BCMS Split Report

The BCMS Split Report provides traffic information for the specified split number. Depending on specifics from the command line, the information may be displayed as either a time interval or a daily summary. If neither `time` nor `day` is specified, `time` is the default. In this case, the report displays data accrued for the previous 24 time intervals (hour or half-hour), including data from the most recently completed time interval. To get information on the current time interval, you must use a `monitor bcms` command. Screen 3-15 shows the BCMS Split or Skill Summary Report — Hourly, and Screen 3-16 shows the BCMS Split or Skill Report — Daily.

NOTE:
When analyzing this report, keep the following things in mind:

- All averages are for completed calls only.

- A completed call may span more than one time interval. ACD calls that are in process (have not terminated) are counted in the time interval in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time interval, but terminates in the 11:00 to 12:00 time interval, the data for this call is counted in the 11:00 to 12:00 time interval.
- Asterisks within a field indicate that the maximum for that field has been exceeded.

```
 list bcms split 3 time 8:00-10:00
                               BCMS SPLIT REPORT

Switch Name: Lab Model                         Date: 11:05 am MON MAY 15, 1995
      Split: 03
 Split Name: services                           Acceptable Service Level: 17

                  AVG          AVG    AVG    TOTAL              TOTAL        % IN
            ACD   SPEED ABAND  ABAND  TALK   AFTER  FLOW FLOW  AUX/  AVG    SERV
TIME        CALLS ANS   CALLS  TIME   TIME   CALL   IN   OUT   OTHER STAFF  LEVL

 8:00- 9:00    32   :25     4    :32  5:15   16:00     3    5   3:30   4.0   80*
 9:00-10:0      8   :07     1    :03  3:20     :00     0    0   9:30   2.2   85
----------- ----- ----- -----  ----- ----- -------  ---- ---- ------ ----- ---
 SUMMARY       40   :21     5    :26  4:52     :26     3    5  13:00   3.1   81
```

Screen 3-15. BCMS Split or Skill Report — Hourly

* Acceptable service level changed.

“Split” is displayed as “Skill’’ when EAS is optioned.

```
 list bcms split 3 day 5/14/95

                               BCMS SPLIT REPORT

Switch Name: Lab Model                         Date: 11:05 am MON MAY 15, 1995
      Split: 03
 Split Name: services                           Acceptable Service Level: 17

                  AVG          AVG    AVG    TOTAL              TOTAL        % IN
            ACD   SPEED ABAND  ABAND  TALK   AFTER  FLOW FLOW  AUX/  AVG    SERV
DAY         CALLS ANS   CALLS  TIME   TIME   CALL   IN   OUT   OTHER STAFF  LEVL

 5/14/95       40   :21     5    :26  4:52   17:20     3    5  13:00   3.1   81
 --------    ----- -----  ----  ----- ----- ------- ---- ---- ------- ----- ---
SUMMARY        40   :21     5    :26  4:52   17:20     3    5  13:00   3.1   81
```

Screen 3-16. BCMS Split or Skill Report — Daily

"Split'' is displayed as "Skill'' when EAS is optioned.

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number of the report, the title of the report, and the time and date the report was generated. If this is a time report and there are more than 10 time intervals, this report is displayed on multiple pages. A daily summary report is displayed on the last page of the report.

SPLIT — The split number specified with the command line.

SPLIT NAME — Displays the name that is administered for this split number. If no name exists, BCMS displays the split extension (for example, EXT 65432).

ACCEPTABLE SERVICE LEVEL — The desired time to answer for a given hunt group. Timing for a call begins when the call enters the hunt group queue.

TIME/DAY — The time or day interval specified in the command line.

Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the earliest time interval (either hour or half-hour). If no start time is given, the oldest time interval is the default. A stop time requires an associated start time. If no stop time is given, the last completed time interval (hour or half-hour) is the default. If no start or stop time is given, the report displays data accrued for the previous 24 time intervals. If you specify ''day'' in the command and do not include a start day or stop day, the report displays data accrued for the previous six days and data accrued through the most recently completed interval (hour or half-hour).

ACD CALLS — The number of ACD calls completed for this split during the current interval. This number also includes calls that flowed in from other splits and Direct Agent calls.

AVG SPEED ANS — The average amount of time answered ACD calls (split and Direct Agent) spent in queue and ringing at an agent's station before being answered during the reporting interval. Calls that flowed in do not have queue time from the previous split included in this average. This calculation is:

$$\text{AVG SPEED ANS} = \frac{\text{Sum of Each Answered Call's Time In Queue} + \text{Time Ringing at the Agent's Extension}}{\text{Total Number of ACD Calls Answered}}$$

NOTE:
Keep the following things in mind:

- This value does not include time listening to a forced first announcement.
- A completed call may span more than one time period. ACD calls that are in process (have not terminated) are counted in the time period in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time period, but terminates in the 11:00 to 12:00 time period, the data for this call is counted in the 11:00 to 12:00 time period.

ABAND CALLS — The total number of ACD calls that have hung up while waiting to be answered during this time interval. This value includes those calls that have abandoned while in queue or while ringing. Calls that are not queued (because the queue is full, the caller receives a forced first announcement and abandons during the announcement, or no agents are staffed) are not counted as abandoned. Also, calls that abandon while on hold are not counted as abandoned.

AVG ABAND TIME — The average time before an ACD call abandons. This value does not include any time spent in another split's queue before flowing into this split. The calculation is:

$$AVG\ ABAND\ TIME = \frac{Total\ Abandon\ Time}{Total\ Number\ of\ Abandoned\ Calls}$$

NOTE:
This value does not include time listening to a forced first announcement or calls that ‘‘abandon’’ while listening to a forced first announcement.

AVG TALK TIME — The average amount of time agents are active on ACD calls (split and direct agent) for each split. This includes time spent talking. The calculation does not include ring time at an agent's voice terminal or time spent on hold. The calculation is:

$$AVG\ TALK\ TIME = \frac{Total\ ACD\ Talk\ Time}{Total\ Number\ of\ ACD\ Calls}$$

TOTAL AFTER CALL — The amount of time that the agents in this split spent in call-related or noncall-related ACW mode during the reporting interval. This value includes time spent on direct incoming or outgoing calls while in ACW. If an agent entered ACW in one interval, but left ACW in another interval, each interval is credited with ACW time.

FLOW IN — The total number of completed calls that this split received as a coverage point (intraflowed) from another BCMS-measured split, or are call forwarded (interflowed) to this split during the reporting interval. This total does not include calls that are interflowed from a remote switch by means of the Look Ahead Interflow feature. FLOW INs are recorded when a call ends.

FLOW OUT — The total number of calls queued to this split that were:

- Successfully sent to the split's coverage point after queuing for the specified ''don't answer'' interval. (This does not include calls that went to coverage based on any other criterion.)
- Forwarded-out via call forwarding
- Forwarded-out via a route to station extension vector step
- Answered via the Call Pickup feature
- Forwarded-out via Look Ahead Interflow
- First queued to this split and answered by the second or third split queued to
- Were redirected back to this split or its coverage path due to Redirect On No Answer timing.

FLOW OUTs are recorded when a call ends.

NOTE:
In a multiple split-queuing environment, inflows and outflows become a bit more complicated. Consider the following scenarios:

- If a multiply queued call is answered in a nonprimary split (that is, a second or third split), an outflow is recorded to the statistics for the first split, and an inflow and an answer are recorded to the statistics for the answering split. For example, suppose there are three splits numbered 1 through 3. A call comes in for split 1, but all agents are busy on this split. The call then goes into queue for splits 2 and 3. An agent on split 3 answers the call. In this example, an outflow is recorded to the statistics for split 1, and an inflow and an answer are recorded to the statistics for split 3. The statistics for split 2 are unaffected because the call was not answered in this split. This scenario is shown in the following table.

Table 3-13. Call Answered by Nonprimary Split

	Split Pegging		
	Split 1	Split 2	Split 3
BCMS	outflow	dequeued	inflow answer

If the call is answered in the primary split, no inflows or outflows are recorded to the statistics for any split. Splits 2 and 3 record the call as dequeued.

- If a call is queued on three splits (for example, splits 1, 2, and 3, with split 1 being the primary split), then encounters a `route-to` command that sends the call to another VDN, that queues to different splits (for example, splits 4 and 5), an outflow is recorded to the statistics for split 1. If the call is answered in split 4, an answer is recorded to the statistics for split 4. However, no inflow is recorded to the statistics for split 4. This scenario is shown in the following table.

Table 3-14. Call Answered by Primary Split After a Route to VDN

	Split Pegging				
	Split 1	Split 2	Split 3	Split 4	Split 5
BCMS	outflow	dequeued	dequeued	answer	dequeued

If the call is answered on split 5, an outflow is recorded for the statistics to split 4, and both an inflow and an answer are recorded to the statistics for split 5. This scenario is shown in the following table.

Table 3-15. Call Answered by Non-Primary Split After a Route to VDN

	Split Pegging				
	Split 1	Split 2	Split 3	Split 4	Split 5
BCMS	outflow	dequeued	dequeued	outflow	inflow answer

Similarly, if a multiply queued call routes to another split, an outflow is recorded to the statistics for the primary split, but no inflow is recorded to the statistics for the routed-to split.

TOTAL AUX/OTHER — The total time that logged-in agents in this split were unavailable to receive calls during the reporting interval. This value includes time spent on non-ACD calls while in AUX for this split. This value does not include the time agents spent on another split's calls or in ACW for another split.

NOTE:
A split totals AUX TIME whenever any agent logs into the split and:

- Receives a EXTN call while in AUX or AVAIL state
- Makes a EXTN call while in AUX or AVAIL state
- Hits his/her AUX button
- Other

Furthermore, the split report measurement AUX TIME is time-interval based, since it is not directly related to a call. For example, if an agent is in AUX for any of the previously identified reasons from 9:55 to 10:05, then five minutes is pegged in the 9:00 to 10:00 time interval and five minutes is pegged in the 10:00 to 11:00 time interval.

If you perform these calculations for each agent within a split and total them — The calculated number should generally be the same as displayed on the split report. However, because of differences in how the agent and split reports handle EXTN calls you may (occasionally) see different numbers between the two reports.

AVG STAFF — The average number of agents who were logged into this split (staffed) during the reporting interval.

$$AVG\ STAFF = \frac{Total\ Staff\ Time}{Time\ Interval}$$

% IN SERV LEVL — The percentage of calls answered within the administered service level.

$$\%\ IN\ SERV\ LEVL = \frac{Accepted * 100}{ACD\ calls + abandons + outflows + dequeued}$$

where

accepted is calls answered whose queue time was less than or equal to the administered service level for the split

dequeued is a call that encountered the split's queue, but that was NOT answered, abandoned, or outflowed. This occurs with multiple split queuing.

SUMMARY — For those columns that specify averages, the summary is an average for the entire reporting interval. For the ACD CALLS, ABAND CALLS, TOTAL AFTER CALL, FLOW IN, FLOW OUT, AUX TIME, and TOTAL HOLD TIME columns, the summary is the sum of individual time intervals or specified days.

BCMS Split Summary Report

NOTE:
This report replaces (and enhances) the BCMS System Report. Customers with upgrades from previous DEFINITY system releases running BCMS will see that their scheduled `list bcms system` command is changed automatically to the `list bcms summary split` command to get this new report.

The BCMS Split Summary Report provides traffic measurement information for a specified group of BCMS splits. Depending on specifics from the command line, the information may be displayed as either a time interval or daily summary. If neither ‘‘time’’ nor ‘‘day’’ is specified, ‘‘time’’ is the default. In this case, the report displays data accrued for the previous 24 time intervals (hour or half-hour), including data from the most recently completed time interval. To get information on the current time interval, you must use a `monitor bcms` command. Screen 3-17 shows the BCMS Split or Skill Summary Report — Hourly Summary, and Screen 3-18 shows the BCMS Split or Skill Summary Report — Daily Summary.

This report is similar to the Split Report except that this report provides one line of data for each split, and that includes all data for the specified times. A split does not appear on the report if there is no data for that split. If you specify more than one time period, and data exists for one or more, but not all, of the specified times, the system uses the available data to calculate and display the one-line summary; the system does not identify which agents are not included in the calculations.

Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the earliest time interval (either hour or half-hour). If no start time is given, the most recent time interval is the default. A stop time requires an associated start time. If no stop time is given, then, only the last interval of data will be used to calculate the one-line display for each split. If you specify ‘‘day’’ in the command and do not include a start day or stop day, the report displays data for the current day accrued through the most recently completed interval (hour or half-hour).

NOTE:
When analyzing this report, keep the following things in mind:

- All averages are for completed calls only.
- Asterisks indicate that the maximum for the associated field has been exceeded.

```
list bcms summary split 3 15 time 9:00-16:00

                             BCMS SPLIT SUMMARY REPORT

Switch Name: Lab Model                           Date: 11:05 am MON MAY 15, 1995
       Time: 9:00-16:00

                  AVG          AVG    AVG    TOTAL               TOTAL        % IN
            ACD   SPEED ABAND  ABAND  TALK   AFTER  FLOW FLOW    AUX/   AVG   SERV
SPLIT NAME  CALLS ANS   CALLS  TIME   TIME   CALL   IN   OUT     OTHER  STAFF LEVL

 Sales         32   :25     4    :32  5:15   16:00     3    5     3:30   4.0   75
 Service        8   :07     1    :03  3:20     :00     0    0     9:30   2.2   83*
--------    ----- -----  ----  ----- ----- ------- ---- ---- -------  ----- ---
SUMMARY        40   :21     5    :26  4:52   16:00     3    5    13:00   3.1   76
```

Screen 3-17. BCMS Split or Skill Summary Report — Hourly Summary

"SPLIT" is displayed as "SKILL" when EAS is optioned.

```
list bcms summary split 5 3 day

                             BCMS SPLIT SUMMARY REPORT

Switch Name: Lab Model                           Date: 11:05 am MON MAY 15, 1995
        Day: 5/15/95

                  AVG          AVG    AVG    TOTAL               TOTAL        % IN
            ACD   SPEED ABAND  ABAND  TALK   AFTER  FLOW FLOW    AUX/   AVG   SERV
SPLIT NAME  CALLS ANS   CALLS  TIME   TIME   CALL   IN   OUT     OTHER  STAFF LEVL

 Sales         32   :25     4    :32  5:15   16:00     3    5     3:30   4.0   75
 Service        8   :07     1    :03  3:20     :00     0    0     9:30   2.2   83*
--------    ----- -----  ----  ----- ----- ------- ---- ---- -------  ----- ---
SUMMARY        40   :21     5    :26  4:52   16:00     3    5    13:00   3.1   76
```

Screen 3-18. BCMS Split or Skill Summary Report — Daily Summary

"SPLIT" is displayed as "SKILL" when EAS is optioned.

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number of the report, the title of the report, and the time and date the report was generated. If this is a time report and there are more than 10 time intervals, this report is displayed on multiple pages. A daily summary report is displayed on the last page of the report.

TIME/DAY — The time or day interval specified in the command line.

Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the earliest time interval (either hour or half-hour). If no start time is given, the oldest time interval is the default. A stop time requires an associated start time. If no stop time is given, the last completed time interval (hour or half-hour) is the default. If no start or stop time is given, the report displays data accrued for the previous 24 time intervals. If you specify ‘‘day’’ in the command and do not include a start day or stop day, the report displays data accrued for the previous six days and data accrued through the most recently completed interval (hour or half-hour).

SPLIT NAME — Displays the name that is administered for this split number. If no name exists, the split extension (for example, EXT 65432) is displayed.

ACD CALLS — The number of ACD calls completed for this split during the current interval. This number also includes calls that flowed in from other splits and Direct Agent calls.

AVG SPEED ANS — The average amount of time ACD calls (split and Direct Agent) spent in queue and ringing at an agent’s station before being answered during the reporting interval. Calls that flowed in do not have queue time from the previous split included in this average. This calculation is:

$$AVG\ SPEED\ ANS\ =$$

$$\frac{Sum\ of\ Each\ Completed\ Call's\ Time\ In\ Queue\ +\ Time\ Ringing\ at\ the\ Agent's\ Extension}{Total\ Number\ of\ ACD\ Calls\ Answered}$$

NOTE:
Keep the following things in mind:

- This value does not include time listening to a forced first announcement.
- Asterisks indicate that the maximum for the associated field has been exceeded.

ABAND CALLS — The total number of ACD calls that have hung up while waiting to be answered during this time interval. This value includes those calls that have abandoned while in queue or while ringing. Calls that are not queued (because the queue is full, the caller receives a forced first announcement and abandons during the announcement, or no agents are staffed) are not counted as abandoned. Also, calls that abandon while on hold are not counted as abandoned.

AVG ABAND TIME — The average time before an ACD call abandons. This value does not include any time spent in another split's queue before flowing into this split. The calculation is:

$$AVG\ ABAND\ TIME = \frac{Total\ Abandon\ Time}{Total\ Number\ of\ Abandoned\ Calls}$$

NOTE:
This value does not include time listening to a forced first announcement or calls that ''abandon'' while listening to a forced first announcement.

AVG TALK TIME — The average duration of ACD calls (split and direct agent) for each split. This includes time spent talking. The calculation does not include ring time at an agent's voice terminal or time spent on hold. The calculation is:

$$AVG\ TALK\ TIME = \frac{Total\ ACD\ Talk\ Time}{Total\ Number\ of\ ACD\ Calls}$$

TOTAL AFTER CALL — The amount of time that the agents in this split spent in call-related or noncall-related ACW mode during the reporting interval. This value includes time spent on direct incoming or outgoing calls while in ACW. If an agent entered ACW in one interval, but left ACW in another interval, each interval is credited with ACW time.

FLOW IN — The total number of calls that this split received as a coverage point (intraflowed) from another BCMS-measured split, or are call forwarded (interflowed) to this split during the reporting interval. This total does not include calls that are interflowed from a remote switch by means of the Look Ahead Interflow feature. FLOW INs are recorded as they occur.

FLOW OUT — The total number of calls queued to this split that were:

- Successfully sent to the split's coverage point after queuing for the specified ''don't answer'' interval. (This does not include calls that went to coverage based on any other criterion.)
- Forwarded-out via call forwarding
- Answered via the Call Pickup feature
- Forwarded-out via Look Ahead Interflow
- Forwarded-out via a "route to" station extension vector step

- First queued to this split and answered by the second or third split queued to
- Were redirected back to this split or its coverage path due to Redirect On No Answer timing.

FLOW OUTs are recorded when a call ends.

NOTE:
In a vectoring environment, inflows and outflows become a bit more complicated. Consider the following scenarios:

- If a multiple queued call is answered in a nonprimary split (that is, a second or third split), an outflow is recorded to the statistics for the first split, and an inflow and an answer are recorded to the statistics for the answering split. For example, suppose there are three splits numbered 1 through 3. A call comes in for split 1, but all agents are busy on this split. The call then goes into queue for splits 2 and 3. An agent on split 3 answers the call. In this example, an outflow is recorded to the statistics for split 1, and an inflow and an answer are recorded to the statistics for split 3. The statistics for split 2 are unaffected because the call was not answered in this split. This scenario is shown in the following table.

Table 3-16. Call Answered by Nonprimary Split

	Split Pegging		
	Split 1	Split 2	Split 3
BCMS	outflow	dequeued	inflow answer

If the call is answered in the primary split, no inflows or outflows are recorded to the statistics for any split. Splits 2 and 3 record the call as not recorded.

- If a call is queued on three splits (for example, splits 1, 2, and 3, with split 1 being the primary split), then encounters a `route-to` command that sends the call to another VDN, that queues to different splits (for example, splits 4 and 5), an outflow is recorded to the statistics for split 1. If the call is answered in split 4, an answer is recorded to the statistics for split 4. However, no inflow is recorded to the statistics for split 4. This scenario is shown in the following table.

Table 3-17. Call Answered by Primary Split After a Route to VDN

	Split Pegging				
	Split 1	Split 2	Split 3	Split 4	Split 5
BCMS	outflow	dequeued	dequeued	answer	dequeued

If the call is answered on split 5, an outflow is recorded for the statistics to split 4, and both an inflow and an answer are recorded to the statistics for split 5. This scenario is shown in the following table.

Table 3-18. Call Answered by Non-Primary Split After a Route to VDN

	Split Pegging				
	Split 1	Split 2	Split 3	Split 4	Split 5
BCMS	outflow	dequeued	dequeued	outflow	inflow answered

Similarly, if a multiple queued call routes to another split, an outflow is recorded to the statistics for the primary split, but no inflow is recorded to the statistics for the routed-to split.

TOTAL AUX/OTHER — The total time that logged-in agents in this split were unavailable to receive calls during the reporting interval. This value includes time spent on non-ACD calls while in AUX for this split. This value does not include the time agents spent on another split's calls or in ACW for another split.

NOTE:
A split totals AUX/OTHER TIME whenever any agent logs into the split and:

- Receives a EXTN call while in AUX or AVAIL state
- Makes a EXTN call while in AUX or AVAIL state
- Hits his/her AUX button
- Other

Furthermore, the split report measurement AUX TIME is time-interval based, since it is not directly related to a call. For example, if an agent is in AUX for any of the previously identified reasons from 9:55 to 10:05, then five minutes is pegged in the 9:00 to 10:00 time interval and five minutes is pegged in the 10:00 to 11:00 time interval.

If you perform these calculations for each agent within a split and total them — the calculated number should generally be the same as displayed on the split report. However, because of differences in how the agent and split reports handle EXTN calls you may (occasionally) see different numbers between the two reports.

AVG STAFF — The average number of agents who were logged into this split (staffed) during the reporting interval.

$$AVG\ STAFF\ =\ \frac{Total\ Staff\ Time}{Time\ Interval}$$

% IN SERV LEVL — The percentage of calls answered within the administered service level.

$$\%\ IN\ SERV\ LEVL\ =\ \frac{Accepted*100}{ACD\ calls+\ abandons+\ outflows+\ dequeued}$$

where

accepted is calls answered whose queue time was less than or equal to the administered service level for the split

dequeued is a call that encountered the split's queue, but that was NOT answered, abandoned, or outflowed. This occurs with multiple split queuing.

SUMMARY — For those columns that specify averages, the summary is an average for the entire reporting interval. For the ACD CALLS, ABAND CALLS, TOTAL AFTER CALL, FLOW IN, FLOW OUT, AUX TIME, and TOTAL HOLD TIME columns, the summary is the sum of individual time intervals or specified days.

BCMS Trunk Group Report

The BCMS Trunk Group Report gives statistical information for all BCMS trunk groups. The BCMS Trunk Group Report may be used by the ACD administrator and/or manager to monitor use of the trunk group and to determine the optimal number of trunks for the trunk group. Depending on specifics from the command line, the information may be displayed as either a time interval or a daily summary. If neither ''time'' nor ''day'' is specified, ''time'' is the default. In this case, the report displays data accrued for the previous 24 time intervals (hour or half-hour), including data from the most recently completed time interval. The next two screens show the BCMS Trunk Group Time Interval Report and the BCMS Trunk Group Daily Report.

NOTE:
When analyzing this report, keep the following things in mind:

- All averages are for completed calls only.
- A completed call may span more than one time interval. ACD calls that are in process (have not terminated) are counted in the time interval in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time interval, but terminates in the 11:00 to 12:00 time interval, the data for this call is counted in the 11:00 to 12:00 time interval.
- Asterisks in a field indicate that the maximum for that field has been exceeded.
- A single asterisk at the end of a time or date field indicates that during the interval, trunk group administration occurred that changed the number of trunks.

```
 list bcms trunk 1 time 8:00 11:00

                              BCMS TRUNK GROUP REPORT
Switch Name: Lab Model                           Date: 12:59 pm THU APR 20, 1995
      Group: 1
 Group Name: TG 1                                         Number of Trunks: 11

            |         INCOMING            |          OUTGOING          |%ALL %TIME
TIME        |CALLS ABAND   TIME       CCS | CALLS  COMP   TIME     CCS|BUSY  MAINT

 8:00- 9:00*    23     2   2:15    31.02       1     1   1:36    .96    0      0
 9:00-10:00     35     2   1:48    35.74       4     4   1:42   4.08    0      0
10:00-11:00     24     1   1:40    22.93       0     0    :00    .00    0      0
-----------  -----  ---- ------ --------   -----  ---- ------ ------  ---    ---
SUMMARY         82     5   1:54    29.89       5     5   1:39   2.52    0      0
```

Screen 3-19. BCMS Trunk Group Time Interval Report

```
 list bcms trunk 1 day 4/17

                              BCMS TRUNK GROUP REPORT
Switch Name: Lab Model                                  Date: 12:59 pm THU APR 20, 1995
      Group: 1
 Group Name: TG 1                                                Number of Trunks: 11

            |         INCOMING           |          OUTGOING        |%ALL  %TIME
DAY         |CALLS ABAND   TIME      CCS | CALLS  COMP   TIME    CCS|BUSY  MAINT

 4/17/95*       82     5   1:54    29.89       5     5   1:39   2.52    0      0
-----------  -----  ---- ------ --------   -----  ---- ------ ------  ---    ---
 SUMMARY        82     5   1:54    29.89       5     5   1:39   2.52    0      0
```

Screen 3-20. BCMS Trunk Group Daily Report

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number of the report, the title of the report, and the time and date the report was generated. If this is a time interval report and there are more than 11 time intervals, this report is displayed on multiple pages. A daily summary report is displayed on one page.

Group — The trunk group number specified with the command line.

Group Name — The name that is administered for this trunk group. If no name is administered, then this field is displayed as blank.

Number of Trunks — The number of individual trunks in the trunk group at the end of the first interval being reported.

TIME/DAY — The time or day interval specified in the command line.

Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the top of the time interval (either hour or half-hour). If no start time is given, the report displays data accrued for the previous 24 time intervals. A stop time requires an associated start time. If no stop time is given, the last completed time interval (hour or half-hour) is the default. If no start time or stop time is given, the report displays data accrued for the previous 24 time intervals. If you specify ‘‘day’’ in the command and do not include a start day or stop day, the report displays data accrued for the previous six days and data accrued through the most recently completed interval (hour or half-hour).

If switch administration causes the number of trunks in a BCMS-measured trunk group to change during a day or a time interval, an asterisk appears in the ‘‘DAY/TIME’’ field.

INCOMING CALLS — The total number of incoming calls carried by this trunk group.

INCOMING ABAND — The number of incoming calls that queued to ACD splits, then abandoned (without being answered by a staffed agent within this split) during the reporting interval. Calls that cannot queue (for example, queue full, or calls that receive a busy signal from the Central Office because there aren’t any available trunks) are not included in the INCOMING ABAND number. Also included are calls directly to staffed ACD agents that are unanswered.

INCOMING TIME — The average holding time for incoming calls to this trunk group during the specified reporting interval. Holding time is defined as the length of time in minutes and seconds that a facility is used during a call. The calculation for incoming time is:

$$INCOMING\ TIME = \frac{Total\ Holding\ Time\ \text{for}\ all\ Incoming\ Calls}{Total\ Number\ of\ Incoming\ Calls}$$

INCOMING CCS — The total holding time (usage) for incoming calls to the trunk group during the specified reporting interval. The units are expressed in hundred call seconds (CCS). Refer to the Glossary for a description of the term CCS or hundred call seconds.

OUTGOING CALLS — The total number of outgoing calls for this trunk group during the specified reporting interval.

OUTGOING COMP — The total number of outgoing calls that were placed over this trunk group and answered during the specified reporting interval.

NOTE:
Completion is determined by either return of network answer supervision, or a call that lasts longer than the answer supervision time-out parameter; whichever occurs first.

OUTGOING TIME — The average holding time for outgoing calls during the specified reporting interval. The calculation is:

$$OUTGOING\ TIME = \frac{Total\ Holding\ Time\ \text{for}\ Outgoing\ Calls}{Total\ Number\ of\ Outgoing\ Calls}$$

OUTGOING CCS — The total holding time for outgoing calls from this trunk group. The units are expressed in CCS.

% ALL BUSY — The percentage of time that all the trunks in this trunk group were busy. This value includes trunks that are maintenance busy. The calculation is:

$$\%\ ALL\ BUSY = \frac{Total\ of\ all\ Busy\ Times}{Time\ Interval} \times (100)$$

where Busy Times is expressed in minutes and is the sum of all times when all trunks were simultaneously busy.

% TIME MAINT — The percentage of time that one or more trunks have been busied-out for maintenance purposes. The calculation is:

$$\%\ TIME\ MAINT = \frac{Total\ Maintenance\ Busy\ Time \times 100}{Time\ Interval \times Number\ of\ Trunks\ in\ Group}$$

where:

- Total Maintenance Busy Time is the sum of Maintenance Busy Time (in minutes) for all trunks (individually) in this trunk group during this interval
- Time Interval is expressed in minutes (for example, *30* if using a half-hour interval, *60* if using a one-hour interval, and *1440* if using a daily summary)

NOTE:
For reporting purposes, call data is stored during the time interval (hour or half-hour) that the trunk goes idle, not when the station releases. Also, changing the number of trunks in a trunk group can cause unexpected results for that interval.

BCMS Trunk Group Summary Report

The BCMS Trunk Group Summary Report provides information about BCMS-measured trunk groups. You can specify the trunk groups you want included in the report. The BCMS Trunk Group Report can be used by the ACD administrator and/or manager to monitor use of one or more trunk groups and to determine the optimal number of trunks for the trunk groups. Note that this applies only to trunk groups measured by BCMS.

This report is similar to the BCMS Trunk Group Report except that the information for a trunk appears on separate lines of the report, with totals of activity for all trunks in the trunk group for the specified time. You can specify up to 30 trunk groups. You can print the report for a certain time period specified in either hours or days (up to 7 days).

The report displays only the information that exists and does not identify absent data. If data does not exist for a specified trunk group, the trunk group does not appear on the report. Also, if information does not exist for a portion of the specified time period, the report displays all existing information but does not identify where there is no data. Screen 3-21 shows the BCMS Trunk Group Summary Report for an interval of hours, and Screen 3-22 shows the BCMS Trunk Group Summary Report for a daily interval.

NOTE:
When analyzing this report, keep the following things in mind:

- All averages are for completed calls only.
- Asterisks in a field indicate that the maximum for that field is exceeded.
- A single asterisk at the end of a time or date field indicates that during the interval, trunk group administration occurred that changed the number of trunks.

```
 list bcms trunk sum 23-25 time 8:00

                          BCMS TRUNK GROUP SUMMARY REPORT

Switch Name: Lab Model                              Date: 12:59 pm THU APR 20, 1995
       Time: 8:00-13:00

            |         INCOMING           |           OUTGOING         |%ALL %TIME
GROUP NAME  |CALLS ABAND    TIME      CCS | CALLS  COMP   TIME     CCS|BUSY  MAINT

IN-800         23     2   2:15    31.02       1     1   1:36   0.96    0      0
OUT-WATTS*     35     2   1:48    35.74       4     4   1:42   4.08    0      0
TIE-GROUP      24     1   1:40    22.93       0     0    :00   0.00    0      0
----------- -----  ---- ------ --------   ----- ----- ------ ------  ---    ---
SUMMARY        82     5   1:54    29.89       5     5   1:39   2.52    0      0
```

Screen 3-21. BCMS Trunk Group Summary Report — Hourly

```
 list bcms trunk sum 23 day 4/20/95

                           BCMS TRUNK GROUP SUMMARY REPORT

Switch Name: Lab Model                           Date: 12:59 pm THU APR 20, 1995
        Day: 5/17/95

            |         INCOMING            |           OUTGOING         |%ALL %TIME
GROUP NAME  |CALLS ABAND   TIME       CCS | CALLS  COMP   TIME    CCS|BUSY  MAINT

IN-800*         82     5   1:54    29.89       5      5   1:39   2.52    0      0
-----------  -----  ---- ------ --------   ----- ----- ------ ------   ---    ---
SUMMARY         82     5   1:54    29.89       5      5   1:39   2.52    0      0
```

Screen 3-22. BCMS Trunk Group Summary Report — Daily

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number of the report, the title of the report, and the time and date the report was generated.

TIME/DAY — The time or day interval specified in the command line. Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the top of the time interval (either hour or half-hour). If no start time is given, the report displays data accrued for the previous 24 time intervals. A stop time requires an associated start time. If no stop time is given, the last completed time interval (hour or half-hour) is the default. If no start time or stop time is given, the report displays data accrued for the previous 24 time intervals. If you specify ‘‘day’’ in the command and do not include a start day or stop day, the report displays data accrued for the previous six days and data accrued through the most recently completed interval (hour or half-hour).

If switch administration causes the number of trunks in a BCMS-measured trunk group to change during a day or a time interval, an asterisk appears in the ‘‘DAY/TIME’’ field.

GROUP NAME — The name that is administered for this trunk group. If no name is administered, this field is displayed as blank.

INCOMING CALLS — The total number of incoming calls carried by this trunk group.

INCOMING ABAND — The number of incoming calls that queued to ACD splits, then abandoned (without being answered by a staffed agent within this split) during the reporting interval. Calls that cannot queue (for example, queue full, or calls that receive a busy signal from the central office because there are no available trunks) are not included in the INCOMING ABAND number. Also included are calls directly to staffed ACD agents that are unanswered.

INCOMING TIME — The average holding time for incoming calls to this trunk group during the specified reporting interval. Holding time is defined as the length of time in minutes and seconds that a facility is used during a call. The calculation for incoming time is:

$$INCOMING\ TIME = \frac{Total\ Holding\ Time\ for\ all\ Incoming\ Calls}{Total\ Number\ of\ Incoming\ Calls}$$

INCOMING CCS — The total holding time (usage) for incoming calls to the trunk group during the specified reporting interval. The units are expressed in hundred call seconds (CCS). Refer to the Glossary for a description of the term CCS or hundred call seconds.

OUTGOING CALLS — The total number of outgoing calls for this trunk group during the specified reporting interval.

OUTGOING COMP — The total number of outgoing calls that were placed over this trunk group and answered during the specified reporting interval.

NOTE:
Completion is determined by either return of network answer supervision, or a call that lasts longer than the answer supervision time-out parameter; whichever occurs first.

OUTGOING TIME — The average holding time for outgoing calls during the specified reporting interval. The calculation is:

$$OUTGOING\ TIME = \frac{Total\ Holding\ Time\ for\ Outgoing\ Calls}{Total\ Number\ of\ Outgoing\ Calls}$$

OUTGOING CCS — The total holding time for outgoing calls from this trunk group. The units are expressed in CCS.

% ALL BUSY — The percentage of time that all the trunks in this trunk group were busy. This value includes trunks that are maintenance busy. The calculation is:

$$\%\ ALL\ BUSY = \frac{Total\ of\ all\ Busy\ Times}{Time\ Interval} \times (100)$$

where

Busy Times is expressed in minutes and is the sum of all times when all trunks were simultaneously busy.

% TIME MAINT — The percentage of time that one or more trunks have been busied-out for maintenance purposes. The calculation is:

$$\% \ TIME \ MAINT = \frac{Total \ Maintenance \ Busy \ Time \times 100}{Time \ Interval \times Number \ of \ Trunks \ in \ Group}$$

where

- Total Maintenance Busy Time is the sum of Maintenance Busy Time (in minutes) for all trunks (individually) in this trunk group during this interval
- Time Interval is expressed in minutes (for example, *30* if using a half-hour interval, *60* if using a one-hour interval, and *1440* if using a daily summary)

NOTE:
For reporting purposes, call data is stored during the time interval (hour or half-hour) that the trunk goes idle, not when the station releases. Also, changing the number of trunks in a trunk group can cause unexpected results for that interval.

BCMS VDN Report

The BCMS VDN Report provides statistical information for the specified VDN. Depending on specifics from the command line, the information may be displayed as either a time interval or a daily summary. If neither ‘‘time’’ nor ‘‘day’’ is specified, ‘‘time’’ is the default. In this case, the report displays data accrued for the previous 24 time intervals (hour or half-hour), including data from the most recently completed interval. The next two screens show the VDN Number Report — Hourly and the VDN Number Report — Daily.

NOTE:
When analyzing this report, keep the following things in mind:

- All averages are for completed calls only.
- A completed call may span more than one time period. ACD calls that are in process (have not terminated) are counted in the time period in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time period, but terminates in the 11:00 to 12:00 time period, the data for this call is counted in the 11:00 to 12:00 time period.
- Asterisks indicate that the maximum for the associated field has been exceeded.

```
list bcms vdn 12345 time 8:00 12:00

                        BCMS VECTOR DIRECTORY NUMBER REPORT

Switch Name: Lab Model                         Date: 11:05 am MON MAY 15, 1995
        VDN: 12345
   VDN Name: Ginsu Knives                       Acceptable Service Level:  17

                             AVG          AVG    AVG                CALLS  % IN
              CALLS   ACD    SPEED ABAND  ABAND  TALK/ CONN   FLOW BUSY/  SERV
TIME          OFFERED CALLS  ANSW  CALLS  TIME   HOLD  CALLS  OUT  DISC   LEVL

08:00-09:00        79    50    :39     5    :45   2:30     0     0    24    85*
-----------     ----- ----- -----  -----  -----  ----- -----  ----  ----   ---
SUMMARY            79    50    :39     5    :45   2:30     0     0    24    85
```

Screen 3-23. BCMS VDN Report — Hourly

```
list bcms vdn 12345 day 5/14

                        BCMS VECTOR DIRECTORY NUMBER REPORT

Switch Name: Lab Model                         Date: 11:05 am MON MAY 15, 1995
        VDN: 12345
   VDN Name: Ginsu Knives                       Acceptable Service Level:  17

                             AVG          AVG    AVG                CALLS  % IN
              CALLS   ACD    SPEED ABAND  ABAND  TALK/ CONN   FLOW BUSY/  SERV
DAY           OFFERED CALLS  ANSW  CALLS  TIME   HOLD  CALLS  OUT  DISC   LEVL

5/14/95            79    50    :39     5    :45   2:30     0     0    24    85*
-----------     ----- ----- -----  -----  -----  ----- -----  ----  ----   ---
SUMMARY            79    50    :39     5    :45   2:30     0     0    24    85
```

Screen 3-24. BCMS VDN Report — Daily

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number of the report, the title of the report, and the time and date the report was generated. If this is a time report and there are more than 10 time intervals, this report is displayed on multiple pages. A daily summary report is displayed on the last page of the report.

VDN — The VDN specified with the command line.

VDN NAME — The name that is administered for this VDN. If no name exists, the VDN extension (for example, EXT 64532) is displayed.

ACCEPTABLE SERVICE LEVEL — The desired time to answer the VDN. Timing for a call begins when the VDN is encountered.

TIME/DAY — The time or day interval specified in the command line.

Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the earliest time interval (either hour or half-hour). If no start time is given, the oldest time interval is the default. A stop time requires an associated start time. If no stop time is given, the last completed time interval (hour or half-hour) is the default. If no start or stop time is given, the report displays data accrued for the previous 24 time intervals. If you specify ‘‘day’’ in the command and do not include a start day or stop day, the report displays data accrued for the previous six days and data accrued through the most recently completed interval (hour or half-hour).

CALLS OFFERED — The total number of ended calls that accessed the VDN during the current interval. This calculation is:

$$CALLS\ OFFERED = NUM\ ANS + FLOW\ OUT + CALLS\ BUSY/DISC + NUM\ ABAND$$

ACD CALLS — The total number of calls to the VDN that ended in the specified interval and were answered by an agent in a BCMS-measured hunt group. ACD calls include calls that reached the split via the queue-to-main, check backup, route-to, messaging split, or adjunct routing commands.

AVG SPEED ANS — The average speed of answer for answered ACD and CONNect calls that have ended for this VDN during the current period. This includes time in vector processing, time in a split’s queue, and time ringing. This calculation is:

$$AVG\ SPEED\ ANS = \frac{Total\ Answer\ Time}{Total\ ACD\ Calls + Total\ Connect\ Calls}$$

NOTE:
A completed call can span more than one time period. ACD calls that are in process (have not terminated) are counted in the time period in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time period, but terminates in the 11:00 to 12:00 time period, the data for this call is counted in the 11:00 to 12:00 time period.

ABAND CALLS — The total number of calls that have abandoned from the VDN before being answered or outflowed to another position during the current interval. This value includes calls that abandoned while in vector processing or while ringing an agent. Calls that abandoned immediately after the agent answered are recorded as NUM ANS.

AVG ABAND TIME — The average time calls spent waiting in this VDN before being abandoned by the caller during the current interval. The calculation is:

$$AVG\ ABAND\ TIME = \frac{Total\ VDN\ Abandon\ Time}{Total\ Number\ of\ Abandoned\ VDN\ Calls}$$

AVG TALK/HOLD TIME — The average duration of calls (from answer to disconnect) for this VDN during the current interval. This includes time spent talking and on hold. The calculation does not include ring time at an agent's voice terminal. The calculation is:

$$AVG\ TALK\ TIME = \frac{Total\ VDN\ Talk/Hold\ Time}{NUM\ ANS}$$

CONN CALLS — The number of ended calls that were routed to a station, attendant, or announcement, and were answered there.

FLOW OUT — The total number of ended calls that were routed to another VDN or to a trunk, including successful lookahead attempts.

FLOW OUT does not include calls that encounter a `goto vector` command.

Once a call outflows, the system does not take further measurements on the call for this VDN. As a result, if an outflowed call later abandons, it is not recorded in NUM ABAND for this VDN.

CALLS BUSY/DISC — The total number of calls that were forced busy or forced disconnect during the current interval. This value does not include abandoned calls.

% IN SERV LEVL — The percentage of calls that were answered with the administered service level for this VDN. Calculate as the following:

$$\%\ IN\ SERV\ LEVL = \frac{accepted * 100}{calls\ offered}$$

where

accepted is the number of answered calls (*num ans*) whose answer time was less than or equal to the administered service level for the VDN. *num ans* here refers to the data item on the form of the same name.

calls offered the total number of completed calls that accessed the VDN during the current interval.

This field is blank if no calls have been recorded for this time interval. This field is also blank if no ''Acceptable Service Level'' is administered on the VDN form.

SUMMARY — For those columns that specify averages, the summary is also an average for the entire reporting interval. For the TOTAL ATTEMPTS, NUM ANS, NUM ABAND, FLOW OUT, and OTHER CALLS columns, the summary is the sum of individual time intervals or specified days. *enter command:* prompt.

BCMS VDN Summary Report

This report is similar to the VDN Report except that you can specify up to 30 extensions to include in the report. Also, this report provides one line of data for each VDN included in the report, and the one line includes all data for the specified times. If no data exists for a VDN, the VDN does not appear on the report.

```
 list bcms summary vdn 12345, 13443-13448 time 8:00-12:00

                     BCMS VECTOR DIRECTORY NUMBER SUMMARY REPORT

Switch Name: Lab Model                         Date: 11:05 am MON MAY 15, 1995
       Time: 8:00-12:00
                             AVG          AVG    AVG                CALLS  % IN
              CALLS   ACD    SPEED ABAND  ABAND  TALK/ CONN    FLOW BUSY/  SERV
VDN NAME      OFFERED CALLS  ANSW  CALLS  TIME   HOLD  CALLS   OUT  DISC   LEVL

EXT 13443           0     0    :00     0    :00    :00     0     0     0
EXT 13444           0     0    :00     0    :00    :00     0     0     0
EXT 13445           0     0    :00     0    :00    :00     0     0     0
EXT 13446           0     0    :00     0    :00    :00     0     0     0
EXT 13447           0     0    :00     0    :00    :00     0     0     0
EXT 13448           0     0    :00     0    :00    :00     0     0     0
Ginsu Knive        79    50    :39     5    :45   2:30     0     0    24     85*
-----------     ----- ----- -----   ----  -----  ----- -----  ----  ----     ---
SUMMARY            79    50    :39     5    :45   2:30     0     0    24     85
```

Screen 3-25. BCMS VDN Summary Report — Hourly Summary

```
 list bcms summary vdn 12345 day 5/14

                   BCMS VECTOR DIRECTORY NUMBER SUMMARY REPORT

Switch Name: Lab Model                          Date: 11:05 am MON MAY 15, 1995
        Day: 5/14/95
                              AVG          AVG    AVG                CALLS  % IN
              CALLS   ACD     SPEED ABAND  ABAND  TALK/ CONN   FLOW BUSY/  SERV
VDN NAME      OFFERED CALLS   ANSW  CALLS  TIME   HOLD  CALLS  OUT  DISC   LEVL

Ginsu Knives       79    50    :39     5    :45   2:30     0     0    24     85*
 -----------   -----   ----- -----  ----  -----  -----  -----  ----  ----   ---
SUMMARY            79    50    :39     5    :45   2:30     0     0    24     85
```

Screen 3-26. BCMS VDN Summary Report — Daily Summary

Report Headers, Abbreviations, and Their Definitions

This report presents header information at the top of each page. This information includes the command entered to generate the report, the page number of the report, the title of the report, and the time and date the report was generated. If this is a time report and there are more than 10 time intervals, this report is displayed on multiple pages. A daily summary report is displayed on the last page of the report.

TIME/DAY — The time or day interval specified in the command line.

Time is always expressed in 24-hour format. Start and stop times are optional. Reports always start at the earliest time interval (either hour or half-hour). If no start time is given, the oldest time interval is the default. A stop time requires an associated start time. If no stop time is given, the last completed time interval (hour or half-hour) is the default. If no start or stop time is given, the report displays data accrued for the previous 24 time intervals. If you specify ‘‘day’’ in the command and do not include a start day or stop day, the report displays data accrued for the previous six days and data accrued through the most recently completed interval (hour or half-hour).

VDN NAME — The name that is administered for this VDN. If no name exists, the VDN extension (for example, EXT 64532) is displayed.

CALLS OFFERED — The total number of completed calls that accessed the VDN during the current interval. This calculation is:

$$CALLS\ OFFERED = NUM\ ANS + FLOW\ OUT + OTHER\ CALLS + NUM\ ABAND$$

where OTHER CALLS are calls classified as forced busy or forced disconnect.

ACD CALLS — The total number of calls to the VDN that ended in the specified interval and were answered by an agent as a result of a queue to main or check backup split step.

AVG SPEED ANS — The average time that calls spend in a vector before being connected as an ACD call to an agent (for example, via a queue to the main split or check backup step) during the current interval. This includes queue time and time ringing at an agent’s station. This calculation is:

$$AVG\ TIME\ TO\ CONNECT = \frac{Total\ Time\ Calls\ spend\ in\ VDN\ before\ being\ answered}{NUM\ ANS}$$

NOTE:
A completed call may span more than one time period. ACD calls that are in process (have not terminated) are counted in the time period in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time period, but terminates in the 11:00 to 12:00 time period, the data for this call is counted in the 11:00 to 12:00 time period.

ABAND CALLS — The total number of calls that have abandoned from the VDN before being answered or outflowed to another position during the current interval. This value includes calls that abandoned while in vector processing or while ringing an agent. Calls that abandoned immediately after the agent answered are recorded as NUM ANS.

AVG ABAND TIME — The average time calls spent waiting in this VDN before being abandoned by the caller during the current interval. The calculation is:

$$AVG\ ABAND\ TIME = \frac{Total\ VDN\ Abandon\ Time}{Total\ Number\ of\ Abandoned\ VDN\ Calls}$$

AVG TALK/HOLD TIME — The average duration of calls (from answer to disconnect) for this VDN during the current interval. This includes time spent talking and on hold. The calculation does not include ring time at an agent's voice terminal. The calculation is:

$$AVG\ TALK\ TIME = \frac{Total\ VDN\ Talk/Hold\ Time}{NUM\ ANS}$$

CONN CALLS — The number of calls that were routed to a station, attendant, or announcement, and were answered there.

FLOW OUT — The total number of calls that were routed to another VDN or to a trunk.

FLOW OUT does not include calls that encounter a `goto vector` command or calls that forward to another extension (which are tracked as CONNected CALLS).

Once a call outflows, the system does not take further measurements on the call for this VDN. As a result, if an outflowed call later abandons, it is be recorded in NUM ABAND for this VDN.

CALLS BUSY/DISC — The total number of calls that were forced busy or forced disconnect during the current interval. This value does not include abandoned calls.

% IN SERV LEVL — The percentage of calls that were answered with the administered service level for this VDN. Calculate as the following:

$$\%\ IN\ SERV\ LEVL = \frac{accepted * 100}{calls\ offered}$$

where

accepted is the number of answered calls (*num ans*) whose answer time was less than or equal to the administered service level for the VDN. *num ans* here refers to the data item on the form of the same name.

calls offered the total number of completed calls that accessed the VDN during the current interval.

SUMMARY — For those columns that specify averages, the summary is also an average for the entire reporting interval. For the TOTAL ATTEMPTS, NUM ANS, NUM ABAND, FLOW OUT, and OTHER CALLS columns, the summary is the sum of individual time intervals or specified days.

Considerations

BCMS provides a set of internal switch measurement reports for telemarketing centers or customer service centers. These reports can help in managing ACD splits (hunt groups) without the need for an adjunct CMS.

The maximum number of measured agents for the BCMS feature is limited to 200 (Generic 3). An agent can be a member of up to three splits, but is treated as a single agent. An agent can be assigned to as many splits as you like, but the agent can only log into three splits at most.

The maximum number of CMS measured agents (both basic and adjunct) is restricted to 400. Agents in multiple splits are counted as one for each split that he or she is assigned.

The maximum number of internally measured trunk groups is limited to 32 (Generic 3).

The maximum number of internally measured splits is limited to 99 (Generic 3). If a split is assigned more than 200 agents in Generic 3, it cannot be measured internally.

A maximum of 25 time intervals are allocated for storing data. A time interval can be either a one-hour or a one-half hour interval.

A maximum of seven summary days is stored for each historical report.

The maximum number of internally measured trunk group members is limited to 400.

The addition of an EPN can affect the operation of the measurements only when the EPN is unavailable. Any resource that resides in the EPN cabinet is not available for use or for measurement data. If a remote Management Terminal connected to the EPN and the fiber link goes down, the Management Terminal session is dropped and the login prompt appears.

Interactions

The following features interact with the BCMS feature:

- Call Coverage

  Calls extended to a BCMS measured split as a coverage point are treated like new incoming calls to that split. These calls increment the FLOW IN field on the BCMS Split report, provided they were covered from the queue of another BCMS measured split. Calls successfully going to a coverage point from a BCMS measured split are included in the FLOW OUT field on the BCMS Split report. Again, those calls must have first been in queue for the split. Calls that cover due to the split queue being full do not cause the FLOW OUT field to be incremented

- Call Forwarding

  Calls forwarded to a BCMS measured split from an extension are treated like new incoming calls to that split. INFLOW and OUTFLOW counts are not affected.

  If a split's calls are forwarded, inflow and outflow apply. An agent's call forwarding does not forward ACD calls.

- Call Pickup

  Calls answered using the call pickup feature are treated as non-ACD calls (EXTN IN) for the agent picking up the call. ACD calls that are picked up for a BCMS measured agent are included in the FLOW OUT column on the BCMS Split report.

- Call Vectoring

  With Call Vectoring, calls can be queued to up to three splits. ACD call count are pegged to the answering split. Abandoned calls, outflows, and disconnects are credited to the first (primary) split. If a call that is queued for three splits is answered by the second or third split, BCMS records an outflow for the primary split, both an answer and inflow for the answering split, and nothing for the other split (the split that is neither the answering split nor the primary split).

- Conference/Transfer

  When an agent conferences or transfers an ACD call, the agent is credited with an ACD call and an ExtnOut call.

- EAS

  BCMS does not use LoginIDs. Tracking for agents is done strictly on physical extensions. Therefore if multiple agents use a single phone (different shifts) the data is combined for these agents and accessed via an extension number.

- Hunt Groups

  The BCMS measurements are not determined in the same way as hunt group measurements although some of the information is similar. Therefore, the two reports may represent the data differently.

- Move Agents From CMS

  If agents are moved from one split to another split by the CMS adjunct, measurements are stopped for the agent's "from" split and started for the agent's "to" split. Generic 3 denies agent move requests when agents are logged in (staffed). This denial is important since it eliminates measurement complications associated with move requests when the agent is on an ACD call. Move requests are also denied if the agent is being moved into an unmeasured split.

  If the adjunct CMS attempts to move an agent that is not being measured by BCMS into a split that is being measured by BCMS, and the move would exceed the maximum of 30 measured agents, the switch rejects the move. Otherwise, internal BCMS measurements are started for the agent. If the adjunct CMS moves an agent from a split that is measured by BCMS to a split that is not BCMS measured, internal measurements for the agent are stopped.

- Night Service

  When night service is activated for a split, new calls go to an alternate destination. The split in night service does not consider these calls to be OUTFLOW. The calls are treated as new incoming calls if the destination is a measured split (that is, they are not considered INFLOW).

- System Measurements

  DEFINITY Communications System Generic 3 can have BCMS reports, adjunct CMS reports, and switch traffic measurements simultaneously.

  The BCMS measurements are not determined in the same way as trunk group measurements although some of the information is similar. Therefore, the two reports may represent data differently.

Administration

The BCMS is administered by the System Administrator. The following items require administration:

- Communication Interfaces

  If the link to the adjunct CMS has been administered, CMS measurements must be busied-out (`busyout sp-link` command) in order to add/remove adjunct CMS and BCMS measured agents or trunks to the switch. When the "busyout" has been released, adjunct CMS checks for translation changes and, if they exist, the current database is updated and measurements are restarted.

  If the link has not been administered (BCMS measurements only), the busyout `sp-link` command is not required to change translation data.

- Hunt Group and Trunk Group

  The measured field on the Hunt Group and Trunk Group forms should be administered as one of the following:

- internal—Measured by BCMS only
- external—Measured by CMS adjunct only
- both—Measured by both BCMS and the adjunct CMS
- none—Not measured (default)

If BCMS has not been administered in customer options, neither “internal” not “both” is allowed. If the split or trunk group is measured by BCMS only, the `busyout sp-link` command is not required to make changes. Measurements can be turned off for a split while agents are logged in, but agents must be logged off to start measurements.

- System Administration

  The BCMS field must be set to ***y*** by an authorized AT&T employee.

- System-Parameters

  The following items require administration:

  - Measurement interval—Specifies what time interval is used for polling and reporting measurement data. The time can be specified by hour or half-hour intervals with “hour” as the default. There is a maximum of 25 time slots available for measurement intervals. If hourly is specified, an entire day of traffic information is available for history reports; otherwise, only half a day is available. This does not affect daily summaries as they always reflect traffic information for the entire day. The interval may be changed anytime, but does not go into effect until the top of the hour.
  - Printer information for the system printer—This includes the printer extension, EIA device bit rate, and lines per page.

Hardware and Software Requirements

No additional hardware is required to support the BCMS feature. However, a customer may decide to use an asynchronous system printer to obtain hard copies of BCMS history reports. The system printer can be interfaced to the switch through the EIA port on the processor board or through any of the alternate data interfaces such as PDMs connected to a digital port, or ADUs connected to a data line circuit port.

BCMS software is required.

Bridged Call Appearance— Multi-Appearance Voice Terminal

Feature Availability

Bridged Call Appearance—Multi-Appearance Voice Terminal is available with all Generic 3 releases. Bridging enhancements available only with G3V4 and later releases are identified throughout this feature description.

Description

The appearance of a voice terminal’s primary extension number at another voice terminal is called a bridged call appearance. To set up a bridged call appearance, the primary extension and the button number associated with it is administered on any two-lamp button on the bridging station voice terminal.

The Bridged Call Appearance feature is used by lifting the handset and pressing the Bridged Appearance button. The user is then bridged onto the other voice terminal’s primary extension number and can handle calls on that extension number. The bridged appearance can be used to originate calls from, and answer calls to, the other voice terminal’s primary extension number. The user can also bridge onto an existing call to or from the other voice terminal.

An incoming call rings the primary extension number’s voice terminal and all voice terminals that have a bridged call appearance of the voice terminal’s primary extension number. Each voice terminal is visually alerted for all bridged appearances on the voice terminal, but has the option of audible ringing.

NOTE:
See the "Ringing — Abbreviated and Delayed" feature for other bridged appearance alerting option.

A bridged call appearance can be assigned to any two-lamp button. It does not require the use of a regular call appearance. A bridged call appearance can be used just like a regular call appearance for most features. For example, the Conference, Transfer, Hold, Drop, and Priority Calling features can be used from a bridged appearance, just as they would be used from a regular call appearance.

G3V4 and later releases allow for the administration of a voice terminal with zero call appearances of its primary extension. In this way, a voice terminal can be administered to have only bridged appearances.

Extension Administrable Buttons and Lamps

With G3V4 and later releases, the Message lamp and certain feature buttons can be administered to apply to a specified extension rather than the extension of the terminal they reside on.

- The Message lamp can be administered to light when messages are waiting for the extension specified on the station form. In this way, the bridged user's terminal can be set up to indicate when messages are waiting for the primary extension.
- The Call Forwarding All Calls and Call Forwarding Busy/Don't Answer buttons can be administered to activate call forwarding for any extension that is on the voice terminal even if this extension is a bridged appearance. In addition, the lamp associated with the call forwarding button can be administered to track the call forwarding status of any extension. In this way, a bridged user cannot only activate or deactivate call forwarding for all primary and bridged appearances of the extension from the bridged appearance terminal, but the bridged appearance terminal will show the call forwarding status of the specified extension.
- The Send All Calls button can be administered to activate Send All Calls for any administered extension. In addition, the lamp associated with Send All Calls tracks the status of the administered extension. In this way, a bridged user can activate Send All Calls for the primary extension user.

Sample Applications

The Bridged Call Appearance feature allows calls to be handled from more than one voice terminal. Some practical uses of this capability are as follows:

- A secretary making or answering calls on an executive's primary extension

  These calls can be placed on hold for later retrieval by the executive, or the executive can simply bridge onto the call. In all cases, the executive handles the call as if he or she had placed or answered the call. It is never necessary to transfer the call to the executive.
- A secretary taking care of details for an executive who is already active on a call

  A secretary can bridge onto an active call and take down information such as an address or telephone number.
- Visitor telephones

  An executive may have another voice terminal in his or her office that is to be used by visitors. It may be desirable that the visitor be able to bridge onto a call which is active on the executive's primary extension number. A bridged call appearance makes this possible.

- Service environments

  It may be necessary that several people be able to handle calls to a particular extension number. For example, several users may be required to answer calls to a hot line number in addition to their normal functions. Each user may also be required to bridge onto existing hot line calls. A bridged call appearance provides this capability.

- A user frequently using voice terminals in different locations

  A user may not spend all of his or her time in the same place. For this type of user, it is convenient to have his or her extension number bridged at several different voice terminals.

- Executive suite environments

  With G3V4 and later releases, terminals can be administered with zero call appearances of their primary extension number. When this feature is combined with extension administrable buttons and lamps, it is possible to administer several nearly identical phones for one office suite. The user can then operate his or her voice terminal the same way regardless of which voice terminal is being used.

Security

G3V4 and later releases provide a system option that prohibits bridged terminals from bridging on to a call when the call has Data Privacy or Data Restriction enabled.

Considerations

A voice terminal's primary extension number can have an appearance on up to seven (15 for G3r) other voice terminals. The number of bridged call appearances allowed at each voice terminal is limited only by the number of two-lamp buttons available on the voice terminal.

Up to six parties can be off-hook and involved in a conversation on a bridged appearance of an extension.

It is recommended that a bridging voice terminal have a bridged call appearance corresponding to each call appearance of the primary extension number at the bridged voice terminal. For example, if a primary voice terminal has three call appearances, then a bridging voice terminal should have three bridged call appearances of that primary extension. This allows users to refer to the individual call appearances when talking about a specific call.

Bridged call appearances may result in the reduction of available feature buttons, thereby reducing a user's capabilities. A Call Coverage module or expansion module can be used to provide up to 20 bridged call appearances.

This leaves the other two-lamp buttons as call appearances, or with other features such as CAS.

If a call terminates at a voice terminal on an extension number other than the primary extension number (for example, Terminating Extension Group, UCD Group, Call Coverage Answer Group, or DDC Group extension number), a bridged call appearance is not maintained. Therefore, it is recommended that the primary terminal not be made a member of such a group (even though administration of this is not prohibited).

The Bridged Call Appearance feature should not be considered as a replacement for Call Coverage.

For G3V2 and later releases, a conference tone can be administered. If the conference tone is enabled, a conference tone is heard when two parties are bridged together on an active call with a third party.

Interactions

The following features interact with the Bridged Call Appearance— Multi-Appearance Voice Terminal feature.

- **Abbreviated Dialing**

  A user, accessing Abbreviated Dialing while on a bridged call appearance, accesses his or her own Abbreviated Dialing lists. The user does not access the Abbreviated Dialing lists of the primary extension associated with the bridged call appearance. This is true even when the voice terminal has no appearances of its primary extension.

- **Attendant Display and Voice Terminal Display**

  A call from the primary extension number or from a bridged call appearance of the primary extension number is displayed as a call from the primary extension number (that is, the call is displayed as coming from the primary extension number regardless of which appearance placed the call).

  The display at a principal will show the same information for a bridged call appearance as it would for a non-bridged call. For calls to the principal’s extension number, the display at a zero call appearance bridging station shows a call from the originator to the principal with no “redirection reason” character. As stations bridge onto the call, the display updates to show the number of parties in the conference.

- **Automatic Callback**

  Automatic Callback calls cannot originate from a bridged call appearance. However, when Automatic Callback has been activated from the principal user’s station, the Callback call rings (with priority call distinctive ringing signal) at all bridged appearances of the extension as well as at the principal user's station. Displays at all (principal and bridged users) stations show that it is a Callback call.

- **Automatic Call Distribution (ACD)**

  ACD directs calls only to the primary extension, not to bridged appearances of the primary extension. Furthermore, if a bridged appearance is active, the primary is considered active and no calls are directed to the primary from the ACD group (unless administered for Multiple Call Handling).

- **Intercom—Automatic and Intercom—Dial**

  Bridged appearances of a primary extension number are not rung for intercom calls. Furthermore, if a station has no primary call appearances it can never be rung for an intercom call. Therefore, if a secretary is screening all calls for the principal, and is indicating who is calling via intercom, the principal must have a call appearance on which to receive and send intercom calls.

- **Call Coverage**

  Coverage criteria for bridged call appearances is based entirely on the criteria of the primary extension associated with bridged appearance. That is, a call to the primary extension that requires call coverage treatment follows the coverage path of the primary extension and not the path of any of the bridged appearances. Bridged call appearances do not receive redirection notification.

  A user with bridged call appearances can activate or deactivate Send All Calls for a principal’s primary call appearance.

  It is recommended that the primary terminal not be a member of a call coverage group, because calls to the primary terminal as a member of the group are not bridged.

  With G3V4 and later releases, the system can be administered so that a call can appear at a terminal as both a bridged call and a redirected call. In this way, if the bridged user is the first coverage point the call will redirect to that terminal when the coverage criteria are met.

  With G3V3 and earlier releases, if the principal is an analog voice terminal with a bridged call appearance on a multi-appearance voice terminal, an incoming call to the analog voice terminal that goes to coverage will terminate at a primary call appearance on the bridging user’s voice terminal as a coverage call. the call is dropped from the analog voice terminal and the bridged call appearance on the bridging user’s voice terminal. In G3V4, if the bridging user is a zero primary call appearance voice terminal, the call cannot redirect to the bridging user since there are no primary call appearances. Therefore, the call will redirect to the next available coverage point.

- **Call Detail Recording (CDR)**

  If a bridging user originates and/or answers a call on a bridged appearance, the extension of the bridge is recorded as the calling/called terminal. A conference or transfer by a bridging user also appears as though it was performed by the terminal user.

When a call originated from a bridged call appearance on a terminal administered for zero primary call appearance is recorded by CDR, the extension number associated with the appearance is recorded as the calling party. A conference or transfer by a bridged call appearance on a zero primary call appearances terminal also appears as though it were performed by the extension associated with the appearance.

- **Call Forwarding All Calls**
  Call Forward Busy/Don’t Answer

  Call Forwarding can be activated or canceled for the primary extension number from any bridged call appearance of that number; when activated, calls to the primary extension number do not terminate at the bridged call appearances, but go to the designated forwarding destination. Bridged call appearances do not receive redirection notification of the call to the primary extension when it is forwarded unless Ringing — Abbreviated and Delayed is administered.

- **Call Park**

  When a call is parked from a bridged call appearance, it is parked on the primary extension number associated with the bridged appearance.

- **Call Pickup**

  If a voice terminal receives ringing on a bridged call appearance, the incoming call can be picked up by members of that voice terminal’s Call Pickup group. This causes all bridged call appearances to be dropped. Calls ringing at a primary terminal can be picked up by members of the terminal’s Call Pickup group. However, if the primary terminal and the bridging user’s primary terminal are not in the same Call Pickup group, then the bridging user cannot pick up calls to other members of the primary terminal’s Call Pickup group.

  Originating on a bridged appearance and dialing the Call Pickup FAC is interpreted as an attempt to pick up a call from the primary terminal’s Call Pickup group.

  A bridging user can use Call Pickup to pick up a call that is alerting at a bridged appearance, instead of selecting the bridged appearance button. This causes the call to terminate on the bridging user’s primary extension button, and the primary terminal and all bridged appearances of the call are dropped.

  If the bridging user has appearances of numerous terminals (for example, Sales, Service, Warehouse, and so on), and it is not desired that the calls be answered by anyone other than the terminal user or the bridging users, then the bridging user(s) should not be assigned to a pick up group.

  A terminal with zero primary call appearances can be assigned to a Call Pickup group.

- **Class of Restriction (COR)**

  The COR assigned to a voice terminal’s primary extension also applies to calls originated from a bridged call appearance.

- **Conference—Attendant**
  Conference—Terminal

Conferences can be set up on bridged appearances using the usual conference operations. Either a primary extension button or a bridged appearance button can be used to make the calls for adding to the conference. With G3V3 and earlier releases, if the bridging user has at least one available bridged appearance of the primary extension (other than the one he or she is currently on), the system automatically selects a bridged call appearance for the conference when the Conference/Transfer button is pressed. If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the Conference/Transfer button, must select a call appearance (or the users' own primary extension) or a bridged appearance of some other extension to be used for the conference, before dialing the number.

With G3V4 and later releases, the system can be administered to automatically select the first idle appearance if there is no idle appearance with an extension matching the extension that is conferencing the call.

NOTE:
When the user depresses the Conference button (the second time) to connect the parties together, the newly formed conference call appears on the primary or bridged appearance to which the user was connected at the time of that last Conference button depression. The other appearance is disassociated from the Conference call. Thus, if the original call was on a bridged appearance, and the conference is formed on an appearance of the bridged user's own primary extension, the bridged extension becomes disassociated from the conference call and the principal user of that bridged extension can no longer bridge onto the conference.

This disassociation of the conference from the bridged extension can be avoided by setting up the conference in the opposite order. To do so the user first uses the Hold button to hold the original call on the bridged appearance, then selects a call appearance and calls the party to be added, then depresses the Conference (or Transfer) button, then selects the held bridged appearance, then depresses the Conference button (again). When this procedure is used, the conference is formed on the bridged appearance so the primary user of the bridged extension can still bridge onto the conference call.

NOTE:
If the primary user and the bridged user are both on the call when one user transfers the call, the user performing the transfer becomes the controlling user for the participation of both users on the

conference. To disassociate the appearance from the call, the controlling user must be the latter of the two users to hang up from the call. If the controlling user hangs up first, the appearance goes on soft hold when the non-controlling party hangs up. In this case, one of two things must occur to disassociate the appearance from the call: (1) all other parties on the call hang up, or (2) the controlling user rejoins the call and hangs up again.

The display shows the number of other active parties in a call, including active bridged appearances.

- **Consult**

  Bridged call appearances of the primary extension do not ring on a consult call to the primary extension.

- **Coverage Answer Group (CAG)**

  Bridged call appearances of a primary extension do not ring when there is a CAG call to the primary. Bridged call appearances cannot bridge onto the call.

- **Data Privacy**
  Data Restriction

  When Data Privacy is activated or Data Restriction is assigned to a station involved in a bridged call and the primary terminal and/or bridging user attempts to bridge onto the call, Data Privacy and Data Restriction are automatically overridden.

  G3V4 and later releases provide a system option that prohibits bridged terminals from bridging on to a call when the call has Data Privacy or Data Restriction enabled.

- **Facility Busy Indication**

  With G3V3 and earlier releases, when the user presses the busy-ind button, a call is placed to the resource from an idle primary call appearance. With G3V4 and later releases, for zero primary call appearance voice terminals, the call is placed to the resource from the first available bridged call appearance.

- **Hold—Automatic**

  Any user (primary or bridged appearance) can place an active call on hold. If only one user is active on a call and places that call on hold, then the indicator lamp at both the principal’s appearance button and the bridged party’s appearance button shows that the call is on hold. If more than one user is bridged onto the active call, and one of the users activates Hold, the activator receives ‘‘hold’’ indication for the call and status lamp of all other bridged users remain ‘‘active.’’

- **Hunt Group (DDC or UCD)**

  Bridged call appearances cannot be used in conjunction with DDC or UCD hunt groups.

Although a bridged extension can be assigned to a hunt group, such assignment is not recommended because DDC/UCD calls do not terminate at any bridged appearances of that extension on other stations.

If a member of a UCD group is off-hook on a bridged appearance, and that user's primary extension number (which is in the UCD group) is idle, then a UCD call will not terminate on that user's primary extension number unless it is administered for Multiple Call Handling.

- **Intercom—Automatic**

  Automatic and/or intercom calls to the primary extension do not ring at the associated bridged appearances.

- **Internal Automatic Answer (IAA)**

  Calls terminating to a bridged appearance of an IAA-eligible station are not eligible for IAA.

- **Last Number Dialed**

  Activation of the Last Number Dialed feature causes the last number dialed from the voice terminal to be redialed, regardless of whether its own call appearance or a bridged call appearance is used.

- **Leave Word Calling (LWC)**

  A LWC message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension number assigned to the bridged call appearance.

  When a user calls a primary extension, and activates LWC, the message is left for the primary extension, even if the call was answered at a bridged call appearance.

  LWC messages left by the primary user can be canceled by a bridged appearance user (for example, a secretary can cancel a LWC message left by a boss).

- **Personal Central Office Line (PCOL)**

  If a user is active on his or her primary extension number on a PCOL call, bridged call appearances of that extension number cannot be used to bridge onto the call. The call can only be bridged onto if another voice terminal is a member of the same PCOL group and has a PCOL button.

- **Priority Calling**

  The primary terminal user or the bridging user can make a priority call. If a priority call is made to an idle terminal, the primary terminal and all bridging users are alerted by priority alerting.

- **Privacy-Manual Exclusion**

  Exclusion prevents any other user from bridging onto the call. Activation of exclusion by any user (primary or bridged appearance) prior to placing a call, prevents any other user from bridging onto the call. Activation of exclusion by any user active on a call, while the primary user and/or any other bridging users are active on the call, drops all other users from the call (including the primary user), leaving only the activator and the calling/called party on the call.

- **Redirection Notification**

  Redirection Notification is not provided at stations with a bridged appearance of a primary extension number unless Ringing — Abbreviated and Delayed is administered to give notification.

- **Ringback Queuing**

  Ringback Queuing is not provided on calls originated from a bridged call appearance. However, after the principal user of the bridged extension has activated ringback queueing, the resulting Callback call alerts at bridged appearances as well as at the principal user's station. The call can be answered from the primary user's station or from any bridged appearance.

- **Ringer Cutoff**

  When activated at a multi-appearance station, Ringer Cutoff prevents any nonpriority (or non-intercom) incoming call from ringing at that station. This is independent of whether the call is to the station's primary extension or to any of the bridged appearances' extension(s).

- **Service Observing**

  The terminal user or bridging user can bridge onto a Service Observed call at any time. If the terminal is being Service Observed and an incoming call is answered by the bridging user, the call is not observed unless or until the terminal user bridges onto the call. Conversely, if the bridging user is being Service Observed and an incoming call is answered by the terminal user, the call is not observed unless or until the bridging user bridges onto the call.

  If the bridging user activates Service Observing, using a bridged appearance, Service Observing is activated for the bridging user.

- **Terminating Extension Group (TEG)**

  TEG calls to the primary extension do not ring at the associated bridged appearances. TEG calls cannot be answered or bridged onto from a bridged appearance of the TEG Member's primary extension. The primary terminal should not be assigned to a TEG.

- **Transfer**

  With G3V3 and earlier releases, if the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the Conference/Transfer button, must select a call appearance to be used for the transfer, before dialing the number.

If the bridging user has at least one available bridged appearance of the primary extension (other than the one he or she is currently on), the system automatically selects a bridged call appearance for the transfer when the Conference/Transfer button is pressed.

With G3V4 and later releases, the system can be administered to automatically select the first idle appearance if there is no idle appearance with an extension matching the extension that is transferring the call.

NOTE:
If the primary user and the bridged user are both on the call when one user transfers the call, the user performing the transfer becomes the controlling user for the participation of both users on the conference. The controlling user is immediately dropped from the call. When the non-controlling user hangs up, the appearance goes on soft hold. In this case, one of two things must occur to disassociate the appearance from the call: (1) all other parties on the call hang up, or (2) the controlling user rejoins the call and hangs up again.

- **Voice Message Retrieval**

  A voice message to the primary extension can be retrieved on a bridged appearance by the bridged appearance user. If a security code is required to retrieve the message, the bridging user must use the security code of the primary terminal.

- **Voice Paging**

  The use of Voice Paging automatically invokes exclusion; therefore, interactions for this feature is the same as for Privacy-Manual Exclusion.

Administration

The Bridged Call Appearance — Multi-Appearance Voice Terminal feature is administered on a per-voice terminal basis by the System Manager. See “Bridged Call Appearance — Multi-Appearance Voice Terminal” in *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for instructions.

Hardware and Software Requirements

No additional hardware or software is required. A Call Coverage module or expansion module can be used to provide up to 20 bridged call appearances.

Bridged Call Appearance— Single-Line Voice Terminal

Feature Availability

Bridged Call Appearance—Single-Line Voice Terminal is available with all Generic 3 releases. Other bridging enhancements available only with G3V4 and later releases are identified throughout this feature description.

Description

Allows a multi-appearance terminal to have an appearance of a primary terminal's extension number. The appearance of the primary terminal's extension number at a multi-appearance terminal is called a bridged call appearance.

G3V4 and later releases also allow an analog voice terminal to have a single bridged appearance of a multi-appearance voice terminal primary call appearance.

The bridged call appearance can be used to originate, answer, or bridge onto an existing call to or from the primary terminal user's extension number.

The multi-appearance terminal user can use the bridged call appearance by lifting the handset and pressing the bridged appearance button or by pressing the bridged appearance button and lifting the handset. The user is then bridged onto the primary terminal's extension number and can handle calls on that extension number.

The bridged single-line voice terminal user bridges onto a call by going off-hook.

When the primary terminal is a single-line voice terminal, the user can also bridge onto an existing call originated, answered, or bridged onto by the associated multi-appearance terminal(s) by just going off-hook.

An incoming call alerts at the primary terminal, and at all voice terminals that have a bridged call appearance of the primary terminal's extension number. Each multi-appearance bridging voice terminal has visual alerting with the option of audible ringing for the bridged appearance of the primary terminal.

When the primary terminal user answers the call, the audible ringing stops at the primary terminal and at all the bridging user's terminals, and the status lamps at all the multi-appearance bridged appearance buttons light steadily. The call can then be bridged onto by any of the bridging users.

When a bridging user answers the call, the audible ringing stops at the primary terminal and at all of the bridging user's terminals, and the status lamps at all of the multi-appearance bridged appearance buttons light steadily. The call can then be bridged onto by any of the bridging users or by the primary terminal station user. After ringing ceases at the single-appearance terminal (primary or bridged) the terminal user has no indication of the call's existence. In this case, if the terminal user did not hear the ringing, the user would not know of the existence of the call and could inadvertently pick up on an active call.

NOTE:
See the "Ringing — Abbreviated and Delayed" feature for other bridged appearance alerting option.

A bridged call appearance can be assigned to any two-lamp button. It does not require the use of a regular call appearance.

A bridged call appearance can only be used to originate and/or answer calls on the primary terminal's extension number, or to bridge onto an active call. The bridging user cannot access a Call Waiting call or a call on hard or soft hold by the primary terminal user.

Because of the aforementioned restrictions, certain limitations are placed on the use of the Conference, Transfer, and Hold features for both the primary terminal user and the bridging users. When more than one user is active on a call (that is, a primary terminal user and one or more bridging users; or two or more bridging users; or any configuration that has more than one bridging party on an established call), attempts to use the Conference, Transfer, or Hold features are denied.

When Call Waiting and/or Priority Call Waiting is assigned to the primary terminal, these features apply only when the primary terminal user is off-hook on a call. They are not active when the multi-appearance terminal user is off-hook on a call on the bridged appearance of the primary terminal.

When a single-line primary or bridged appearance terminal user is off-hook on an active call, normal single-line Conference, Transfer, and Hold procedures apply.

When a multi-appearance bridging or primary appearance user is off-hook on an active call on the primary terminal's extension number, then normal multi-appearance terminal Conference, Transfer, and Hold procedures apply.

Sample Applications

The bridging of a primary terminal satisfies certain conditions that require handling a call from locations other than that of the primary terminal. Some of these situations are as follows:

- A secretary placing calls for an executive(s)

  These calls can be placed on hold for later retrieval by the executive, or the executive can simply bridge onto the call. In all cases, the executive handles the call as if he or she had placed or answered the call. It is never necessary to transfer the call to the executive.

- A secretary taking care of details for an executive, such as a call to the finance department, traffic department, and so on (any call that requires automatic identification of the executive's extension number).

  A secretary can bridge onto an active call and take down information such as an address or telephone number.

- Visitor telephones

  An executive may have another voice terminal in his or her office which is to be used by visitors. It may be desirable that the visitor be able to bridge onto a call which is active on the executive's primary extension number. G3V4 and later releases also allow for the administration of a single-line, analog terminal as a visitor phone with a bridged appearance of a primary extension from the executive's phone.

- Service environments

  It may be necessary that several people be able to handle calls to a particular extension number. For example, several users may be required to answer calls to a hot line number in addition to their normal functions. Each user may also be required to bridge onto existing hot line calls.

- A user frequently using voice terminals in different locations

  A user may not always spend time in the same place. For this type user, it is convenient to their extension number bridged at several different voice terminals.

Considerations

The primary terminal's extension number can have an appearance on up to seven ''bridging'' user's terminals for G3vs/G3s, and G3i, and 15 for G3r. (Bridging means other terminals bridged onto the call.) A bridging user cannot have more than one bridged appearance for a particular primary terminal. However, a multi-appearance bridging user can have appearances of more than one analog terminal on their terminal (that is, a multi-appearance bridging user, by use of different buttons, can bridge onto several different primary terminals).

The number of bridged appearances allowed on a multi-appearance bridging user's terminal is limited only by the number of two-lamp buttons available on the terminal.

If the primary single-line terminal is correctly administered, but not in service, calls can still be placed, by the bridging users, and received on the bridged appearances of the terminal. The primary terminal can be out of service for several reasons, such as an unplugged terminal, a non-existent terminal system technician busyout command, and so on.

If more than one user goes off-hook on a bridged appearance at the same time, only the user that was the first to go off-hook can dial.

If a bridging user is not active on a call, and bridges onto the appearance of an active call, then the user is bridged onto the active call. If a multi-appearance bridging user is active on a call, and bridges onto the appearance of an active call, then the previously selected call is dropped and the user is bridged onto the active call.

The Privacy-Manual Exclusion feature can be activated by the bridging user only, while active on a call, to prevent accidental bridging of an active call.

If a call terminates at a voice terminal on an extension number other than the primary extension number (for example, Terminating Extension Group, UCD Group, Call Coverage Answer Group, or DDC Group extension number), a bridged call appearance is not maintained. Therefore, it is recommended that the primary terminal not be made a member of such a group (even though administration of this is not prohibited).

The Bridged Call Appearance feature should not be considered as a replacement for Call Coverage or any other similar features.

If two parties are bridged together on an active call with a third party, and if the conference tone feature is enabled, the conference tone is heard.

Interactions

NOTE:
In the following description of interactions, the term *''TERMINAL-BASED''* means that the feature applies to that user whether a call appearance or a bridged appearance is being used to activate/deactivate the feature. The term *"EXTENSION-BASED"* means that activation/deactivation of the feature from a bridged appearance is seen by the system as having been made by the primary (single-line) terminal.

- **Abbreviated Dialing**

  A user, accessing Abbreviated Dialing while on a bridged call appearance, accesses his or her own Abbreviated Dialing lists. The user does not access the Abbreviated Dialing lists of the primary extension associated with the bridged call appearance.

- **Attendant Display and Voice Terminal Display**

  A call from the primary extension number or from a bridged call appearance of the primary extension number is displayed as a call from the primary extension number (that is, the call is displayed as coming from the primary extension number regardless of which appearance placed the call).

- ***Authorization (COR, FRL)***

  The COR (including restrictions and FRL) of the primary terminal are always used when authorization checking is required, even when the call is originated from a bridged appearance button by a bridging user.

- **Automatic Callback**

  Automatic Callback calls cannot originate from a bridged call appearance. However, when Automatic Callback has been activated from the principal user's terminal, the Callback call rings (with priority call distinctive ringing signal) at all bridged appearances of the extension as well as at the principal user's terminal. Displays at all (principal and bridged users) stations show that it is a Callback call.

- **Automatic Call Distribution (ACD)**

  Bridged appearances cannot be ACD members (although administrable), but can be used in multiple call situations.

- **Call Coverage**

  When an analog station is administered as a bridged call appearance, the terminal user cannot invoke Send All Calls for the extension of their voice terminal. The user does not have a Send All Calls button, and the call appearance is associated with another extension. When the user dials a Feature Access Code (FAC) Send All Calls is activated for the extension associated with the call appearance.

- **Call Forwarding All Calls**
  Call Forward Busy/Don't Answer

  Call Forwarding can be activated or canceled for the primary terminal number from the primary terminal or from any bridged call appearance of that number; then, when activated, calls to the primary extension number do not terminate at the principal or the bridged call appearances, but go to the designated forwarding destination. Bridged call appearances do not receive redirection notification of the call to the primary extension when it is forwarded unless Ringing-Abbreviated and Delayed is administered.

  When an analog station is administered as a bridged call appearance, the terminal user cannot invoke call forwarding for the extension of their voice terminal. The user does not have a call forwarding button, and the call appearance is associated with another extension. When the user dials a Feature Access Code (FAC) call forwarding is activated for the extension associated with the call appearance.

- **Call Park**

  Call Park is *EXTENSION-BASED*; when a call is parked from a bridged call appearance, it is parked on the primary terminal extension number.

- **Call Pickup**

  Calls to the primary terminal, alerting at bridged appearances of the primary terminal, can be picked up by member's of the bridging user's Call Pickup group; this causes all bridged appearances of the call to be dropped.

  Calls ringing at a primary terminal can be picked up by members of the primary terminal's Call Pickup group. However, if the primary terminal and the bridging user's terminal are not in the same Call Pickup group, then the bridging user cannot pick up calls to other members of the primary terminal's Call Pickup group.

  Originating on a bridged appearance and dialing the Call Pickup FAC is interpreted as an attempt to pick up a call from the primary terminal's Call Pickup group.

  A bridging user can use Call Pickup to pick up a call that is alerting at a bridged appearance, instead of selecting the bridged appearance button. This causes the call at the primary terminal and all bridged appearances of the call to be dropped.

  If the bridging user has appearances of numerous single-line (primary) terminals (for example, Sales, Service, Warehouse, and so on), and it is not desired that the calls be answered by anyone other than the primary terminal user or the bridging users, then the bridging user(s) should not be assigned to a pick up group.

- **Call Waiting Termination**

  Call Waiting Termination applies only to an active call on the primary terminal which has no one else bridged on. If one or more bridging users are active on a call, Call Waiting calls are denied whether or not the primary user is also off-hook on the call. A bridging user cannot bridge onto a call with the primary user if there is also a call waiting.

- **Call Detail Recording (CDR)**

  If a bridging user originates and/or answers a call on a bridged appearance, the primary terminal is recorded as the calling/called terminal. A conference or transfer by a bridging user also appears as though it was performed by the primary terminal user.

- **Conference—Attendant**
  Conference—Terminal

  A bridged call cannot be conferenced if more than one user is active on that call. This is because the bridging user has no access to the call after the primary terminal user places the call on soft hold, and the primary terminal user has no access to the bridging user's call appearance used for conference/transfer attempts.

  If a bridging user is active on a bridged call and the primary analog terminal user attempts a conference, the attempt is ignored. The same is true if an analog bridging user attempts a conference when the primary terminal user and another bridging user is active on a call.

When the primary terminal user is active on a call, and no other bridging user is active on the call, then that call can be placed on hold by the primary terminal user utilizing normal single-line conference procedures. Any attempt by a bridging user to bridge onto the call during a successful conference attempt is denied.

A bridging user, alone on a bridged call, can conference the call utilizing the normal multi-appearance terminal conference procedures. Any attempt by the analog primary terminal user to bridge onto the call during a successful conference attempt is ignored; any attempt by other bridging users is denied (standard denial response is returned to the bridged appearance).

If a conference is not allowed because of the preceding limitations, the user can accomplish a transfer by asking an internal non-bridged party in the connection to create the conference, or asking the remaining bridging users and/or primary user to disconnect so that the conference can be completed. At completion of the conference, the parties that left the call can reenter the call if control of the conference remains with the primary terminal. If control of the conference does not remain with the primary terminal, the bridging user must conference the primary terminal and the bridging user back into the call as required.

If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the Conference/Transfer button, must select a call appearance to be used for the conference, before dialing the number.

- **Coverage Answer Group**

  It is recommended that the primary (analog) terminal not be a member of a call coverage answer group, because calls to the primary terminal as a member of the group *are not bridged*.

  If the primary terminal is made a member of a coverage group, coverage criteria is based entirely on the criteria of the primary terminal. This means that a call to the primary terminal that requires call coverage treatment follows the path of the primary terminal and not the path of any of the terminals with bridged appearances of the primary terminals. In this case, it would be desirable to have the bridging user in the coverage path of the primary terminal. Then, when a call to the primary terminal requires coverage treatment, it follows the coverage path to the bridging user's terminal, call appearances of the call are dropped, and the call terminates at the bridging user's terminal as a coverage call.

- **Data Privacy**
  Data Restriction

  When Data Privacy is activated or Data Restriction is assigned to a station involved in a bridged call and the primary terminal and/or bridging user attempts to bridge onto the call, Data Privacy and Data Restriction are automatically overridden (or deactivated in the case of Data Privacy).

- **Priority Calling**

  Call Priority Calling apply only to an active call on the primary terminal which has no one else bridged on. If one or more bridging users are active on a call, Priority Calling calls are denied whether or not the primary user is also off-hook on the call. A bridging user cannot bridge onto a call with the primary user if there is also a call waiting.

- **Privacy-Manual Exclusion**

  Privacy-Manual Exclusion can only be activated by a bridging user (a button is required for Privacy-Manual Exclusion). Activation of Privacy-Manual Exclusion prohibits any further bridging onto the call. If a bridging user activates Privacy-Manual Exclusion while the primary terminal and/or other bridging users are active on the call, the primary terminal and all bridging users except the activator are dropped from the call.

- **Hold—Automatic**

  A call cannot be put on hold if more than one user is active on that call.

  The primary terminal user, when no other bridges are active on the call, can put the call on hold, using normal single-line hold procedures. If the primary terminal user successfully soft holds the call, the status lamp at all of the bridged appearances shows the hold indication; and then the call can be put on hard hold by dialing the hard hold FAC. The hard held call is no longer accessible to the bridging users until it is taken off hold by the primary terminal user. After the call is put on hard hold, any new call to the primary terminal is tracked by the bridged appearances.

  A bridging user can place an active call on hold (if the primary terminal or any other bridges are not active on the call) by using normal multi-appearance hold procedures. Any attempt to enter the held call returns it to the status of an active call that can then be accessed using bridging procedures.

  If hold is not allowed because of the preceding reasons, the user can just go on-hook and then reenter the call as required, because the call remains accessible as long as the primary terminal or any bridging user is active on it.

- **Hot Line Service**

  If a single-line voice terminal is administered for Hot Line Service, bridged appearances of that voice terminal’s extension also places a hot line call automatically when a user goes off-hook on that bridged appearance.

- **Hunt Group (DDC or UCD)**

  Bridged call appearances cannot be used in conjunction with DDC or UCD hunt groups.

  Although a bridged extension can be assigned to a hunt group, such assignment is not recommended because DDC/UCD calls do not terminate at any bridged appearances of that extension on other stations.

- **Internal Automatic Answer (IAA)**

  Calls terminating to a bridged appearance of an IAA-eligible station are not eligible for IAA.

- **Last Number Dialed**

  Activation of the Last Number Dialed feature causes the last number dialed from the activating voice terminal to be redialed, regardless of the extension number used (primary or bridged call appearance).

- **Leave Word Calling (LWC)**

  A LWC message left by a user on a bridged call appearance leaves a message for the called party to call the primary terminal extension number (that is, the feature is *EXTENSION-BASED*).

  When a user calls a primary terminal, and activates LWC, the message is left for the primary terminal, even if the call was answered at a bridged call appearance.

  LWC messages left by the primary terminal user can be canceled by a bridged appearance user (for example, the secretary can cancel a LWC message left by the boss).

- **Personal Central Office Line (PCOL)**

  A single-line primary terminal cannot be a member of a PCOL group.

- **Preference**

  Ringing Line Preference selects an alerting bridged appearance; Idle Line Preference does not.

- **Priority Calling**

  The primary terminal user or the bridging user can make a priority call. If a priority call is made to an idle primary terminal, the primary terminal and all bridging users are alerted by priority alerting.

  For information on termination of a priority call to an active primary terminal, see Call Waiting Termination/Priority Calling.

- **Ringer Cutoff**

  Ringer Cutoff requires a button; therefore, it cannot be activated by the primary terminal user. Bridging user activation of Ringer Cutoff has no impact on the primary terminal or the other bridging users. However, it prevents the bridged user’s station from ringing on a non-priority call whether the call is to the user’s primary extension or a bridged appearance. This a *TERMINAL-BASED* feature.

- **Ringback Queuing**

  Ringback Queuing is not provided on calls originated from a bridged call appearance. Ringback Queuing is automatically invoked for a primary terminal (primary terminal), and the resulting Callback call alerts at bridged appearances as well as at the principal user's station. That call can be answered from the primary user's station or from any bridged appearance.

- **Service Observing**

  The primary terminal user or bridging user can bridge onto a Service Observed call at any time. If the primary terminal is being Service Observed and an incoming call is answered by the bridging user, the call is not observed unless or until the primary terminal user bridges onto the call. Conversely, if the bridging user is being Service Observed and an incoming call is answered by the primary terminal user, the call is not observed unless or until the bridging user bridges onto the call.

  If the bridging user activates Service Observing, utilizing a bridged appearance, Service Observing is activated for the bridging user (that is, *TERMINAL-BASED* feature).

- **Terminating Extension Group (TEG)**

  TEG calls cannot be answered or bridged onto from a bridged appearance of the TEG member's primary extension.

- **Transfer**

  A call cannot be transferred by an analog station if more than one user is active on that call.

  The primary terminal user, when no other bridges are active on the call, can transfer the call using normal single-line transfer procedures. Any attempt by a bridging user to bridge onto this call during a successful transfer attempt is denied (a standard denial response is returned to the bridged appearance).

  An analog bridging user, alone on a bridged call, can transfer the call, using normal transfer procedures. Any attempt by the primary terminal user to bridge onto this call during a successful transfer attempt is ignored; and any attempt to bridge on by a bridging user is denied.

  If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the Conference/Transfer button, must select a call appearance to be used for the transfer, before dialing the number.

- **Videophone 2500**

  A user may not use an analog bridge to a Videophone 2500 principal that is on a video call.

- **Voice Message Retrieval**

  A voice message to the primary terminal can be retrieved on a bridged appearance by the bridging user. If a security code is required to retrieve the message, the bridging user must use the security code of the primary terminal.

- **Voice Paging**

  The use of Voice Paging automatically invokes Privacy Manual-Exclusion; therefore, interactions for this feature are the same as for Privacy-Manual Exclusion.

Administration

The Bridged Call Appearance — Single-Line Voice Terminal feature is administered on a per voice terminal basis by the system manager. A primary single-line terminal must be translated before any bridged appearances can be translated to point to it. Also, the bridged appearances of the primary terminal must be removed before the primary terminal can be removed.

See “Bridged Call Appearance — Single-Appearance Voice Terminal” in the *DEFINITY Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for instructions.

Hardware and Software Requirements

No additional hardware or software is required. A Call Coverage module can be used to provide up to 20 bridged call appearances.

Busy Verification of Terminals and Trunks

Feature Availability

Busy Verification of Terminals and Trunks is available with all Generic 3 releases.

Description

Allows attendants and specified multiappearance voice terminal users to make test calls to trunks, voice terminals, and hunt DDC and UCD groups. These test calls check the status of an apparently busy resource.

Busy verification of voice terminal extensions, hunt group extensions, and trunks can be done by either multiappearance voice terminal users or attendants or station user. Feature activation is via a Busy Verify button.

An attendant, station user or multiappearance voice terminal user can activate Busy Verification of Terminals and Trunks by pressing the Busy Verify button. The attendant or station user then dials an extension number if a voice terminal or hunt group is to be verified. If a trunk is to be verified, the attendant dials a trunk access code, followed by a two-digit number (leading 0s may be required) to specify which member of the trunk group is to be verified.

After an attendant or multiappearance voice terminal user has activated Busy Verification of Terminals and Trunks, the system checks the validity of the entered extension number or trunk access code and member number. If the entered number is not a voice terminal extension number, a DDC/ UCD group extension number, an ACD split number, or a trunk access code with a valid member number, the verification attempt is denied.

NOTE:
For G3vs/G3s and G3i, the member number is a two-digit number; for G3r, the member number is a three-digit number.

If an attendant activates Busy Verification of Terminals and Trunks for a valid voice terminal extension number, the system initiates a priority call to that extension. One of the following then occurs:

- Voice terminal is idle.

  Priority ringing is heard at the voice terminal and the voice terminal is successfully verified. The call proceeds as a normal attendant-originated call.

- Voice terminal is active on a call.

The system first searches for an idle call appearance on the voice terminal. If one is found, that call appearance is rung. If an idle call appearance cannot be found, or if the voice terminal is a single-line voice terminal, the attendant bridges onto the active call. All parties on the active call receive a warning tone (two-second burst of 440 Hz tone) to let them know that the attendant is bridging onto the call. A half-second burst of warning tone is repeated every 15 seconds, as long as the attendant is bridged onto the call.

NOTE:
If a country requires a different tone than 440 Hz, the attendant should use the intrusion feature rather than busy verification to make these test calls.

- Voice terminal is out of service.

  Busy verification is denied and the attendant receives reorder tone.

If an attendant activates Busy Verification of Terminals and Trunks for a valid ACD split, UCD group, or DDC group, the system initiates a priority call to that group. One of the following then occurs:

- At least one group member is available for incoming calls.

  The call rings the available group member and is treated as a priority call from an attendant to the group.

- All group members have activated the Make Busy function.

  Busy verification is denied and the attendant receives reorder tone.

- Not all group members have activated Make Busy, but no group members are available for incoming calls.

  The call is not queued, even if a queue is available. Busy verification is denied.

If an attendant or a multiappearance voice terminal user activates Busy Verification of Terminals and Trunks for a valid trunk, the system checks the status of that trunk. One of the following then occurs:

- The trunk is idle.

  If the trunk is an outgoing trunk, the originator of the busy verification receives a dial tone and can make a call on that trunk to verify that it is in working order. If the trunk is an incoming trunk, the originator of the busy verification receives a confirmation tone as an indication that the trunk is available for use.

- The trunk is busy with an active call.

The originator of the busy verification is bridged onto the active call. All parties on the active call receive a warning tone (two-second burst of 440 Hz tone) to let them know that the originator of the busy verification is bridging onto the call. A half-second burst of warning tone repeats every 15 seconds, as long as the busy verification originator remains on the call.

- The trunk is out of service.

  The busy verification is denied. The attendant receives reorder tone.

If busy verification is denied for any other reason, intercept tone or reorder tone is returned to the user.

Considerations

Busy Verification of Terminals and Trunks provides attendants with an easy method of checking the condition of certain extensions and trunks. An attendant or multifunction voice terminal can distinguish between a voice terminal that is truly busy and one that only appears busy because of some trouble condition. Attendants or multifunction voice terminal users can also use the feature to quickly identify faulty trunks. As a result, better communications service is provided and faulty trunks can be corrected more quickly.

A busy verification can be performed on the following:

- Voice terminal extensions
- UCD and DDC hunt group extensions
- Members of the following types of trunk groups:
  - DID
  - CO
  - FX
  - WATS
  - APLT
  - Tie
  - Remote Access
  - RLT

The bridging capability associated with Busy Verification of Terminals and Trunks is not provided on verification attempts to UCD and DDC groups or RLTs.

Outgoing test calls cannot be made on DID trunks.

Busy verification may be activated for a phantom extension that is ‘‘administered without hardware.’’ In this case, an ‘‘Out of Service’’ indication is provided.

Interactions

The following features interact with the Busy Verification of Terminals and Trunks feature.

- Automatic Callback

  Once the called party in an Automatic Callback call hangs up, neither extension number can be busy verified until both the calling and called parties are connected or the callback attempt is canceled (by the activating party or by time-out of the callback interval).

- Call Coverage

  Since the busy verification call to an extension number is originated as a priority call, the call does not go to coverage.

- Call Forwarding

  A busy verification made to an extension with call forwarding activated, does not busy verify the forwarded-to extension. Only the called extension is busy verified.

- Call Waiting Termination

  A busy verification cannot be made to an extension which is waiting to be answered at another extension.

- Conference — Attendant and Terminal

  If a conference call involves six parties, busy verification on any extension number in the conference is denied. If the number of parties in the conference is five or less, a busy verification can be performed on any of the associated extension numbers. Busy Verification of a trunk that is busy on a 6-party call is also denied.

- Data Privacy

  Busy verification is denied if it results in a bridging attempt on a voice terminal which has activated Data Privacy.

- Data Restriction

  If Data Restriction is active on a call, and a busy verification bridging attempt is made on that call, the busy verification is denied.

- Hold

  A busy verification of a multiappearance voice terminal is denied if all call appearances have calls on hold.

- Individual Attendant Access

  An attendant cannot make a busy verification of another individual attendant console or of the attendant group.

- Loudspeaker Paging Access

  If the voice terminal or trunk to be verified is connected to paging equipment, the verification attempt is denied.

- Voice Terminal Origination Restriction

  A voice terminal that is origination restricted can be assigned a Busy Verify button. However, the button cannot be used.

- Voice Terminal Termination Restriction

  Voice terminals that are termination restricted cannot be busy verified.

- Transfer

  Once the originator of the busy verification has bridged onto a call, any attempt to transfer the call is denied until the Busy Verification originator drops from the call.

Administration

Busy Verification of Terminals and Trunks is administered on a per-voice terminal or per-console basis by the System Manager. The only administration required is the assignment of a Busy Verify button to the desired attendant consoles and multiappearance voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Call Coverage

Feature Availability

Call Coverage is available with all Generic 3 releases. However, for G3vs/G3s ABP, Linked Call Coverage Paths are only available with the Voice Mail Application Software Option package, which is available when you purchase an AT&T voice processing adjunct (AUDIX, AUDIX Voice Power, AUDIX Voice Power Lodging, DEFINITY AUDIX). Linked Call Coverage Paths are always included with all other releases.

Description

Provides automatic redirection of certain calls to alternate answering positions in a Call Coverage path.

Call Coverage Path

A Call Coverage path is a list of one, two, or three alternate answering positions (covering users) that are accessed, in sequence, when the called individual or group (principal) is not available to answer the call. Any of the following can be assigned a call coverage path and are thus eligible to have their calls redirected to coverage.

- Voice terminal
- UCD group
- DDC group
- TEG
- PCOL group
- ACD split

The System Manager establishes the coverage paths and sets the redirection criteria at the time the system is implemented. These paths and criteria can be changed at later dates. If a coverage path is not assigned to a particular facility, calls are not redirected from that facility, unless another feature such as Call Forwarding All Calls is assigned. A coverage path can include any of the following:

- Voice terminal
- Attendant group
- UCD group
- DDC group
- ACD split

- Coverage Answer group, which is a group of up to eight voice terminals specifically established to answer redirected calls. All group members are rung simultaneously. Any group member can answer the call.
- AUDIX

Multiple Coverage Paths

A principal can be assigned multiple coverage paths. Each extension is assigned a coverage path. That coverage path, in turn, can be linked to up to three other coverage paths. This makes a total of four coverage paths that can be assigned to each extension and so on. If a call does not meet any of the redirection criteria in the first coverage path, the call then goes to the next coverage path.

Redirection Criteria

The redirection criteria determine the conditions under which a call redirects from the principal (called) extension number to the first position in the coverage path. The criteria and conditions that apply are as follows:

- **Active**

  Redirects call-to-call coverage immediately when the principal is active on at least one call appearance. For a voice terminal with only one appearance or a single-line extension, the Busy criterion (discussed below) should be assigned instead of the Active criterion.

- **Busy**

  Redirects calls to coverage when all available call appearances at the principal extension are in use. For multiappearance voice terminals, one call appearance is reserved for outgoing calls or incoming priority calls (discussed later). The remaining assigned call appearances are available for other incoming calls. An incoming call (other than a priority call) redirects to coverage only when all of these unreserved call appearances are in use. If at least one unreserved call appearance is idle at the principal extension, the call remains at that idle appearance.

  A Terminating Extension Group (TEG) is considered busy if any voice terminal in the group is active on a call.

  For a UCD or DDC group, each voice terminal in the group must be active on at least one call appearance in order for the call to be redirected to coverage. If any voice terminal in the group is idle (not active on any call appearance) the call directs to that voice terminal. If no voice terminal is available, the call can queue if queuing is provided. If queuing is not provided or if the queue is full, the call routes to coverage. Queued calls remain in queue for a time interval equal to the Don't Answer Interval (discussed next).

- **Don’t Answer**

  Redirects calls to coverage if unanswered during the assigned Don’t Answer Interval (1 to 99 ringing cycles). A call rings for the assigned Don’t Answer Interval and then redirect to coverage.

- **Cover All Calls**

  Redirects all incoming calls to coverage. This criterion has precedence over any other criterion previously assigned.

- **Send All Calls/Go To Cover**

  Allows users to activate Send All Calls or Go To Cover as an overriding coverage criteria. This redirection criteria must be assigned before a user can activate the Send All Calls or Go To Cover features (discussed later).

- **No Coverage**

  Occurs when none of the above criteria have been assigned. Calls are only redirected to coverage when the principal has activated Send All Calls or the caller has activated Go to Cover. Both of these overriding criteria are discussed later.

Redirection criteria can be assigned in combinations; Active/Don’t Answer and Busy/Don’t Answer can be useful. Other combinations are not possible or do not provide any useful function. For example, Active/Busy does not accomplish anything. A busy voice terminal is always active.

Redirection criteria is assigned separately for internal and external calls. Thus, Busy/Don’t Answer can be assigned for internal calls and Active can be assigned for external calls. Similarly, Busy/Don’t Answer could apply for external calls and No Coverage could apply for internal calls. In the latter case, internal calls remain directed to the called terminal or group.

Certain overriding criteria are possible. These criteria, of course, are checked before the redirection criteria are checked. The overriding criteria are:

- **Go to Cover**

  Allows users, when making a call to another internal extension, to send the call directly to coverage. This is optionally assigned to a button on a voice terminal and is activated by the internal calling party. Use of Go to Cover is discussed later.

- **Send All Calls**

  Allows principals to temporarily direct all incoming calls to coverage regardless of the assigned redirection criteria. For example, if the redirection criteria are administered so that no calls redirect, all incoming calls terminate at the principal’s voice terminal unless Send All Calls is activated. Also, activating Send All Calls allows covering users to temporarily remove their voice terminals from the coverage path.

Send All Calls is activated by pressing the Send All Calls button or by dialing the Send All Calls access code. The option is deactivated by pressing the button a second time or by dialing the deactivate code.

If a user is not assigned a coverage path with Send All Calls or Cover All Calls redirection criteria, that user cannot activate Send All Calls. An activation attempt under this condition is denied for both button and dial access.

Send All Calls is similar to Cover All Calls, discussed previously. However, Cover All Calls is set by the System Manager and would be used for screening the principal's call. The principal may or may not be rung on an incoming call, depending on how this function is assigned. Send All Calls is controlled by the principal and is normally used when the principal is away temporarily.

TEG calls are not affected by the activation of Send All Calls.

If a user has activated Send All Calls and only has one coverage point, and receives a call from that coverage point, the call rings silently at the user's voice terminal, because the coverage point is already on the call.

- **Send Term**

  This is the same function as Send All Calls, except Send Term is for a TEG. Since a TEG cannot be in a coverage path, this function only applies to a directly called TEG.

- **Call Forwarding All Calls**

  Call Forwarding provides a temporary override of the redirection criteria. The call attempts to complete to the forwarded-to extension number before redirecting to coverage. If the principal's redirection criteria are met at the forwarded-to extension, then the call is redirected to the principal's coverage path.

All calls extended by the attendant are treated as external.

Call Coverage provides redirection of calls from the called principal or group to alternate answering positions when certain criteria are met. Yet the call is intended for the called principal or group. Certain provisions allow calls to direct to and/or be answered by the principal even though the redirection or overriding criteria are met. These provisions are:

- If no answering positions are available in the Coverage Path, the call rings the called voice terminal, if possible; otherwise, the calling party receives busy tone. This applies even if the Cover All Calls redirection criterion or the Send All Calls overriding criterion is active.
- Similarly, calls directed to a UCD or DDC group are queued, if queuing is available, when no group members are available to answer the call. The call remains in queue for a time interval equivalent to the Don't Answer Interval before routing according to the Coverage Path. If no points on the

path are available, the call remains in queue. The worst case is when group queuing and the coverage points are both unavailable. In this case, the caller receives busy tone.

- If the redirection criterion is Active or Cover All Calls, a called principal can receive a redirection notification signal (a short burst of ringing) when the call routes to coverage. (Redirection Notification is optional on a per-terminal basis.) Note that in the Active, Cover All Calls, and Don't Answer cases, the principal could answer the call. Busy means no call appearances are available to answer the call. Redirected calls maintain an appearance on the called voice terminal, if possible. The call appearance status lamp flashes to indicate an incoming call before the call redirects. When the call does redirect, the status lamp lights steadily. The user can answer the call by pressing the call appearance button. If the call has already been answered, the principal is bridged onto the call. This provision is called Temporary Bridged Appearance.
- Priority Calling, Dial Intercom, and Automatic Intercom Calls always route directly to the principal's voice terminal until the calling party activates Go to Cover. These calls take precedence over the redirection criteria and can seize the call appearance normally reserved for outgoing calls, if no other call appearances are available.

An internal calling party is informed that a call is redirecting to coverage by a single, short burst of ringing, called a Call Coverage tone. This tone is followed by an optional period of silence, called a Caller Response Interval. This interval allows the calling party time to decide what to do: hang up or activate Leave Word Calling, Automatic Callback, or Go to Cover. Activating Go to Cover cancels the remaining interval.

Covering User Options

For specific Call Coverage needs, the following options are available to voice terminal users:

- **Consult**

  Allows the covering user, by first pressing the or Transfer button and then the Consult button, to call the principal (called party) for private consultation. These two actions place the calling party on hold and establish a connection between the principal and the covering user. If the principal wishes, the covering user can complete the conference and add the calling party to the conversation. Similarly, the call can be transferred to the principal. Consult calls use the Temporary Bridged Appearance maintained on the call, if there is one. If not, the Consult call seizes any idle call appearance. If there is no idle call appearance, the Consult call is denied.

- **Coverage Callback**

  Allows a covering user, by pressing the Cover Callback button, to leave a message for the principal to call the calling party. The Coverage Callback feature uses Implied Principal Addressing to infer both extension numbers so the covering user does not have to dial either the principal's or the calling party's number. The calling party must be an internal caller. The principal receives no indication that the covering user handled the call.

Alternatively, if the covering user presses the LWC button, a ''call me'' message is left for the principal. The principal calls the covering user to get the message. This method is used when an external call is received or when an internal caller wants to leave a message but is not available for a return call.

- **Coverage Answer Group**

  A Coverage Answer Group can have up to eight members. When a call is redirected to a Coverage Answer Group, all voice terminals in the group ring simultaneously. Anyone in the group can answer the call. A Coverage Answer Group member already handling a group call is rung when another call is redirected to that Coverage Answer Group. If a Coverage Answer Group member is also a member of another Coverage Answer Group, he or she can also receive calls for the other group. A second call directed to a Coverage Answer Group lights a Coverage Incoming Call Identification (ICI) lamp.

- **Coverage ICI**

  A Coverage ICI button can be assigned to multiappearance voice terminal users without a display in a Coverage Answer Group.

  The Coverage ICI status lamp simply identifies a call incoming to that Coverage Answer Group. If a Coverage Answer Group is assigned to more than one Call Coverage path, the path number cannot be identified. Likewise, if a given path is assigned to more than one principal, the individual principals cannot be identified. To provide unique path and principal identification, the System Manager must establish a unique path for each principal and a unique Coverage Answer Group to be included in the path. A second coverage call takes control of the Coverage ICI lamp and does not return control to the previous call when the second call is released.

What Happens When a Call Goes to Coverage

When a call meets the redirection criteria of the principal, the call attempts to route to one of up to three points in the coverage path, beginning with point one. If no coverage points are available, the call may revert to the called principal or group. If any point in the path is available, the call either rings the individual voice terminal or member of a group specified for that point or queues on the group. Once a call is ringing or queued at any point in a coverage path, the call never reverts to the called principal or group, or to the previous point. A call remains at a coverage point for a time equal to the Don't Answer Interval for Subsequent Redirection (1 to 99 ringing cycles). At the end of this time, the call attempts to route to any remaining points in the coverage path. If no other point is available to accept the call, the call remains queued or continue ringing the current coverage point.

VDN in a Call Coverage Path

Allowing VDN extensions to be the last point in a Call Coverage path provides the flexibility of Call Vectoring for access to a covering point. This provides more flexibility when using Call Vectoring and the "Call Coverage" features which can then be used by AUDIX and Message Server Coverage.

The vector assigned to the VDN in the coverage path can be programmed to queue a redirected call to a messaging split for call answer operation and to allow the caller to leave a message for the called principal. Vector controlled splits are used in coverage paths. The same VDN can also be used to retrieve messages. The vector program may also be varied by time of day or split status to provide different types of coverage.

When a redirected call covers to a VDN, the caller’s temporary bridged appearance is removed at the time vector processing starts. VDN override does not apply to calls that are redirected by the Call Coverage feature.

When covered or direct calls are connected to AUDIX or messaging split via call vectoring, the original reason for redirection and called principal must be passed to the adjunct over the Switch Communication Interface (SCI) link. Also the VDN number must be entered in the ‘‘Group Extension” field instead of the split extension number.

Use of a VDN as a coverage point functions with the Centralized Messaging feature. That is, the Distributed Communications System (DCS) message sent to the remote switch with AUDIX includes the original reason for redirection and called principal.

An administration change is required to allow an extension that is assigned as a VDN to be entered as the last point in the coverage path. See *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 585-230-520, and *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for more information.

Typical Call Coverage Arrangements

Call Coverage is an extremely flexible feature and allows various combinations of coverage points. To illustrate the usefulness of Call Coverage, three typical coverage arrangements are given below as an example.

- **Executive Coverage**

  Provides a principal with call redirection to covering users having a close working relationship with the principal. Because of the status of the principal, personalized answering should be provided. Also, the principal may or may not choose to answer his or her own calls.

A typical example of this form of coverage is when a principal's calls are redirected to a secretary. The secretary would be informed of the principal's daily schedule and other pertinent facts such as the importance of certain calls. The secretary could provide personalized answering by answering calls with the principal's name.

If the secretary is unavailable to answer the coverage call for the principal, the call redirects to a backup answering position. Personalized answering should also be provided at the backup position.

- **Middle Manager Coverage**

  Provides a group of principals with call redirection to one or more covering users (such as a secretary). The secretary should have some knowledge of the principal's daily schedule. A backup answering position should be provided in case the secretary is unavailable.

- **General User Coverage**

  Provides less-personal coverage for a broader spectrum of users. Covering users typically consist of a group or pooled answering arrangement. With this type of arrangement, coverage calls may be distributed among the members of the answering group.

As an example of how to provide a particular cover arrangement, the following provisions for the Executive Coverage arrangement are given.

- Determine if the secretary and backup position have a call display capability.
  - If so, Coverage Answer Groups are not required.
  - If not, establish a unique Coverage Answer Group for each one without a display. Specify only the applicable extension number. The Coverage Answer Group contains only one member. Establish two groups, if required. Note that if the secretary and/or the backup answering position are in a Coverage Answer Group, each receives only one redirected call for the executive at any given time. Calls do ring a Coverage Answer Group member already busy on a call to the group. For frequently called executives, it is desirable that the secretary and possibly the backup answering position have a digital display capability.
- Establish a unique Call Coverage Path for the executive.
  - If the secretary screens the calls, specify Cover All Calls as the redirection criteria.
  - If the executive answers calls, specify Active, Busy, Don't Answer, Active/Don't Answer, or Busy/Don't Answer as desired.
  - Specify the secretary and the backup position [or the Coverage Answer Group(s) containing the secretary's and backup position's extension numbers] as the coverage points in the path.

- Optionally, specify a Send All Calls button on the executive's voice terminal. If someone else answers the executive's calls, the button is not needed.
- Specify a Send All Calls button and a Consult button on the secretary's voice terminal. Specify a Coverage ICI button if the secretary does not have a call display capability. Send All Calls is needed if the secretary is unavailable for a period of time. Consult is needed to enable private consultation with the executive during an established call. Coverage ICI is needed to identify the call as a call to the executive rather than a personal call to the secretary.
- Specify a Consult button and a Coverage ICI button on the backup position's voice terminal for the same reasons these buttons were specified for the secretary.

Considerations

Call Coverage provides the means to redirect calls to alternate answering positions. The feature is versatile enough to permit suitable alternate answering arrangements for virtually every level of employee. Special functions, such as Send All Calls and Consult, accommodate the day-to-day variations that occur in an employee's work schedule. Call Coverage was designed on the premise that incoming calls are intended for the called party, but suitable alternatives must be available if the called party cannot, or does not wish to, answer his or her own calls.

The system allows for hundreds of Coverage Answer groups (See Appendix A, "System Parameters") with up to eight voice terminals in each group.

Incoming tie trunk calls can be administered as either internal or external and are redirected to Call Coverage accordingly.

Interactions

The following features interact with the Call Coverage feature.

- **Administration Without Hardware**

  Stations administered without hardware translation interact with call coverage feature as if all their call appearances are busy. A disassociated station can have call coverage active for the following call coverage groups:

  - coverage answer groups
  - hunt groups
  - intercom groups
  - pickup groups
  - Terminating Extension Groups (TEGs)

- **Agent Call Handling**

  Cover All Calls should not be assigned to agents with the Automatic Answer option. Any call (ACD or non-ACD), to an extension that has Automatic Answer enabled and has its coverage redirection criteria administered as Cover All Calls, does not go to coverage but to the called extension. Cover All Calls redirection criteria has no effect on an incoming call when a user is in the Auto-In mode.

- **Attendant Display and Voice Terminal Display**

  These features provide call identification for the covering user.

- **Automatic Callback and Ringback Queuing**

  Callback calls do not redirect to coverage. The caller can activate Automatic Callback when ringing, redirection notification signal, or busy signal is heard.

- **Automatic Intercom, Dial Intercom, and Priority Calling**

  Calls using these features are not redirected to coverage unless the caller presses the Go to Cover button.

- **Bridged Call Appearance**

  Coverage criteria for bridged call appearances is based entirely on the criteria of the primary extension associated with the bridged call appearance.

  If a voice terminal user has activated Send All Calls on its primary extension, incoming calls still ring bridged call appearances of that extension as long as a Temporary Bridged Appearance of the call is maintained at the primary extension.

- **Call Forwarding**

  Call Forwarding provides a temporary override of the redirection criteria. Normally, calls forward instead of redirecting to coverage. When a forwarding extension number's redirection criteria are met at the designated (forwarded-to) extension number, the call redirects to the forwarding extension's coverage path. However, when the forwarded call goes to coverage, a Temporary Bridged Appearance remains at the forwarded-to voice terminal until the call is answered and released.

  If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding is active at a voice terminal, incoming Priority Calling calls forward to the designated extension number.

  The Redirection Notification Signal applies to both Call Coverage and Call Forwarding.

  If an extension has both Send All Calls and Call Forwarding activated, calls to that extension that can immediately be redirected to coverage are redirected. However, other calls, such as priority calls, are forward to the designated extension.

Activation of Send All Calls at the forwarded-to extension does not affect calls forwarded to that extension.

- **Call Pickup**

  Any call redirected to a covering user who is a member of a Call Pickup group can be answered by other members of the Call Pickup Group.

- **Centralized Attendant Service (CAS)**

  If an incoming CAS call is directed to a hunt group, the call is not redirected to the hunt group's coverage path.

- **Class of Restriction (COR) and Controlled Restrictions**

  Users who may normally be restricted from receiving calls can still receive calls directed to them via Call Coverage.

- **Direct Department Calling (DDC), Uniform Call Distribution (UCD), and Automatic Call Distribution (ACD)**

  If a user has an Auxiliary Work button, and activates or deactivates Send All Calls, the Auxiliary Work function associated with DDC or UCD is activated or deactivated simultaneously.

  If a user has no Auxiliary Work button, activating or deactivating Send All Calls still makes the user available or unavailable for DDC, UCD, and ACD calls, but Auxiliary Work is not activated or deactivated. The Auxiliary Work activate or deactivate code and the DDC, UCD, and ACD extension must be dialed to activate the Auxiliary Work function.

  Activating or deactivating the Auxiliary Work function does not activate or deactivate Send All Calls.

- **Hold**

  If a covering user puts a call on hold, and the principal picks up on the call, the coverage appearance may or may not be dropped, depending on administration.

- **Internal Automatic Answer (IAA)**

  If an internal call is redirected to another voice terminal by a Call Coverage redirection criteria, then that call is eligible for IAA at that voice terminal.

  IAA does not apply to calls to the original called extension when:

  — The called voice terminal has ‘‘Do Not Disturb’’ activated

  — The called voice terminal has activated Send All Calls

  — The called voice terminal has selected Cover All Calls

  — The calling voice terminal has selected Go To Cover before placing the call

  Calls directed to a Coverage Answering Group are not eligible for IAA.

- **Leave Word Caling (LWC)**

  Call Coverage can be used with or without Leave Word Calling (LWC). However, the two features complement each other. When a covering user activates LWC during a coverage call, a message is left for the principal to call the covering user. When a covering user activates Coverage Callback during a coverage call, a message is left for the principal to call the internal caller.

- **Night Service — Night Station Service**

  A call routed to the Direct Inward Dialed (DID) Listed Directory Number (LDN) night extension via Night Station Service does not go to coverage, even if the coverage criteria of the DID LDN night extension is met.

  Calls routed to the attendant via Call Coverage or Call Forwarding do not route to the DID LDN night extension.

- **Temporary Bridged Appearance**

  Calls redirected to coverage maintain an appearance on the called voice terminal if a call appearance is available to handle the call. The called party can bridge onto the call at any time. The system can be administered to allow a temporary bridged appearance of the call to either remain at or be removed from the covering voice terminal after the principal bridges onto the call.

  Consult calls use the Temporary Bridged Appearance maintained on the call. At the conclusion of a consult call, the bridged appearance is no longer maintained. If the principal chooses not to talk with the calling party, the principal cannot bridge onto the call later.

  If a call has or has had a Temporary Bridged Appearance, is conferenced or transferred, and redirects to coverage again, a Temporary Bridged Appearance is not maintained at the conferenced-to or transferred-to extension.

- **Tenant Partitioning**

  The caller and called party must be able to access coverage point. The caller is considered to be the “covering” user and the called party is considered to be the “covered” user. Both parties must be able to access the coverage point.

- **Transfer**

  The Transfer feature interacts with Call Coverage as listed in Table 3-19.

Table 3-19. Transfer and Coverage Interactions

Source	Transfer Initiator	Destination	Coverage Type
External	Local Station	Local Station	External
	Local Station	Remote Station	External
	Remote Station	Local Station	Internal
	Remote Station	Remote Station	Internal
	Attendant	Local Station	External
	Attendant	Remote Station	External
Internal	Local Station	Local Station	Internal
	Local Station	Remote Station	Internal
	Remote Station	Local Station	Internal
	Remote Station	Remote Station	Internal
	Attendant	Local Station	External
	Attendant	Remote Station	External

NOTE:
Transferred DID calls follow the external coverage criteria since they are external calls (trunks).

Administration

Call Coverage is administered by the System Manager. The following items require administration:

- Coverage Paths

  The same coverage path can be used for as many voice terminal users as desired.

- Cover Answer Groups
- Don't Answer Interval and Coverage Subsequent Redirection No Answer Interval

  The Don't Answer Interval specifies the number of ringing cycles heard at the principal's terminal before the call is redirected to the first coverage point. This interval is recommended to be two rings, but can be administered from 1 to 99 rings. All principals with the same coverage path are assigned the same Don't Answer Interval.

  The Don't Answer Interval for Subsequent Redirection specifies the number of rings at a covering terminal before the call attempts to redirect to the next coverage point. This interval is recommended to be two rings, but can be administered from 1 to 99 rings. This interval is administered as a system parameter.

- Caller Response Interval

  This interval can be from 0 to 10 seconds. If 0 is administered, the Caller Response Interval does not apply.

- Redirection Notification Signal

  This signal is administered on a per-terminal basis. If administered, the signal also applies to forwarded calls. With Call Coverage, the signal indicates to the caller that the call is being redirected to coverage because of the Active or Cover All Calls redirection criteria.

- Feature Access Codes for Activation and Deactivation of Send All Calls
- Whether incoming tie trunk calls are treated as internal or external calls
- Whether or not a temporary bridged appearance is maintained by the covering user after the principal bridges onto the call. (“Keep Held SBA at Coverage Point” field on ‘Feature-Related System Parameters Screen’ form.)
- Buttons on Multi-Appearance Voice Terminals, as desired:
  - — Consult
  - — Coverage Callback
  - — Go to Cover
  - — Coverage ICI
  - — Send All Calls

Hardware and Software Requirements

No additional hardware or software is required.

Call Detail Recording (CDR)

Feature Availability

This feature is available with all Generic 3 releases.

NOTE:
‘‘Call Detail Recording (CDR)’’ was previously called ‘‘Station Message Detail Recording (SMDR).’’

Description

Records detailed call information on all incoming and outgoing calls on specified trunk groups and extensions administered for intraswitch recording and sends this information to a CDR output device. The CDR output device provides a detailed printout that can be used by the System Manager to compute call costs, allocate charges, analyze calling patterns, detect unauthorized calls, and keep track of unnecessary calls. For additional information on CDR, see the *Call Detail Acquisition & Processing Reference* manual, 555-006-202.

Call detail information is provided on trunk groups, loudspeaker paging, and code calling access administered for CDR. CDR provides detailed call information for the following types of calls:

NOTE:
Some call accounting systems do not support all the call information offered by CDR. See your sales representative for details.

- Outgoing Calls — Calls originated by a system voice terminal user or attendant going out on a trunk group.
- Incoming Calls — Calls incoming on a trunk group and terminating at a system voice terminal or attendant console.
- Tandem Calls — Calls incoming on a trunk group and outgoing on another trunk group.
- Ineffective Call Attempt — Calls originated by a system voice terminal user blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy. This includes the unavailable incoming or outgoing trunks due to trunk usage allocation for ISDN Call-By-Call Service Selection trunks and incoming calls rejected by the switch due to NSF mismatch.
- Calls made using the Loudspeaker Paging Access and Code Calling Access features.
- Calls involving an auxiliary trunk.
- Temporary Signaling Connections (TSC) involving a trunk.

- Calls returned to a vector directory number (VDN) via the VDN Return Destination feature associated with Call Vectoring and "Call Coverage" features.
- Internal, direct calls that are originated by an extension optioned for intraswitch CDR, or that have a dialed number which is optioned for intraswitch CDR (the Intra-switch option on the ‘‘Change System Parameters CDR’’ form must also be turned on).

NOTE:
If an extension optioned for intraswitch CDR is neither the originator of the call nor the dialed number of the call, no CDR record is output even though the extension might be a party on the call (via Call Pickup, Call Forwarding, etc.).

You have the option of turning off CDR generation for incoming calls, specific trunk group(s), intra-switch calls, Non-Call Associated Temporary Signaling Connection (NCA-TSCs), Call Associated Temporary Signaling Connection (CA-TSCs), and ineffective call attempts with the **change sys cdr** command.

Intra-Switch CDR

Intra-Switch CDR is an administrable option that allows CDR records to be generated for some internal calls. An Intra-Switch call is one that originates and terminates on the same switch. The maximum number of extensions you can administer for Intra-Switch CDR varies from switch to switch.

If your system can record more than 100 stations, the system only displays two pages of extensions (112 per page) at one time. When you enter the **add intra-switch-cdr** command to add extensions, the system automatically begins after the last extensions. For example, if you add 575 extensions, the first field on the first of the two pages is field 576. If you enter the **change intra-switch-cdr** command, the system display begins with the first extension administered. If you enter the **change** command with an extension number, the system begins the display with that extension.

Splitting of CDR Records

Where long-distance calls are expensive and difficult to set up, the call transfer feature is commonly used among different parties (that is, call splitting) to optimize the use of the connections. It is, therefore, important to provide accurate cost allocation data for each leg of the call.

For outgoing calls only, an administrable option is provided in G3i-Global, G3V2, and later releases that creates a separate call record for each leg of calls that are transferred, attendant handled, or conferenced.

CDR Privacy

To ensure the privacy of calls, an administrable option is provided that allows up to seven digits of the ‘‘Dialed Number’’ to be blanked from right to left from the CDR record. This blanks the least significant digits. Certain countries have requirements that specify a certain number of digits must be blanked from every call.

NOTE:
When an adjunct originated call is made on behalf of a hunt group and the CDR system parameter "record hunt group or member extension" option is set to group, then CDR privacy does not apply. If this field is set to member, privacy does apply.

NOTE:
Certain report processors do not support this option.

VDN Return Destination

VDN Return Destination is an administrable option that allows users to re-route calls to an administrable VDN when all parties on the call drop, except for the originator. When the call re-routes to a return destination VDN, it goes back into the vector processing specified by the return destination VDN. The originator can then be re-routed to other destinations without having to hang up and redial or enter an identification number. Refer to the Interactions section for details of how CDR interacts with VDN Return Destination.

Resource Limitation Call Record Handling Options

G3rV1, G3V2, and later releases provide G2-like capabilities for handling of calls when CDR resource limits are exceeded. If new calls come in when the CDR link is down and the buffer is filled, G3r provides the following call record handling options:

- Calls overwrite records (default) - warning
- Calls are blocked with a reorder tone - reorder
- Calls are rerouted to an attendant as non-CDR calls but are not recorded - attendant

  NOTE:
  Rerouting calls to an attendant allows the attendant to manually track calls. This option is only available in MIPS, G3r.

For G3i, G3vsV1/G3sV1, G3iV1, and G3i-Global, records are always overwritten.

Set Time and Date

The system clock must be set for daylight savings time when the time changes. Changing the time and date ensures that CDR records have the correct time and date for the records being kept. The time and date can be changed using the Management Terminal.

If the time is changed while calls are in progress, the actual call durations for these calls are not reflected in the CDR record. A ‘‘9999’’ outputs in the “CDR Duration” field.

CDR Data Formats

This part covers two types of formats sent to the CDR output device, date record and call detail formats.

Date Record Format

Several formats are available for date records: one for CDRUs, one for the printer, and one for the TELESEER. CDR unit. The records sent to the TELESEER CDR and printer contain the date only while the records sent to the CDRU contain time. See Table 3-20, Table 3-21, and Table 3-22.

NOTE:
The date format for G3V2 and later releases can either be in month/day or day/month format, as selected on the “System Parameters CDR’’ form. For G3vsV1/G3sV1, G3iV1, and G3rV1, the month/day format is used. G3i-Global uses the day/month format.

Table 3-20. Date Record Format to LSU, LSU-Expand, Unformatted, and Customized

ASCII Character Position	Data Field Description
01-02	Hour (Leading Zero Added if Needed)
03	Colon (:)
04-05	Minute (Leading Zero Added if Needed)
06	Blank
07-08	Month (Leading Zero Added if Needed)
09	Slash (/)
10-11	Day (Leading Zero Added if Needed)
12	Carriage Return
13	Line Feed
14-16	Null

For G3V2 and later releases there is an option on the ‘‘System Parameters CDR’’ form that is used to administer the CDR date record format. It can be in day/month or month/day format.

Table 3-21. Date Record Format for Printer and Expanded

ASCII Character Position	Data Field Description
01-02	Month (Leading Zero Added if Needed)
03	Space
04-05	Day (Leading Zero Added if Needed)
06	Carriage Return
07	Line Feed
08-10	Null

Table 3-22. Date Record Format for TELESEER 59 Character, Int-Proc, Int-Direct, and Int-ISDN

ASCII Character Position	Data Field Description
01-02	Month (Leading Zero Added if Needed)
03-04	Day
05	Carriage Return
06	Line Feed
07-09	Null

NOTE:
The date/time may also be reversed for international standards.

Call Detail Record Format

The call detail record format provides detailed information concerning an incoming call, an outgoing call, or an intraswitch call. Call detail records are generated during call processing and are sent to the CDR output device in ASCII.

G3vsV1/G3sV1, G3iV1, and G3i-Global support a variety of standard fixed record formats. G3rV1, G3V2, and later releases support a variety of standard fixed record formats and a variable (customized) format.

Customized Format (G3rV1, G3V2, and later releases)

The Variable Record Format (VRF) capability, first introduced in System 85 V4, provides a flexible means to incorporate new data elements in the call record. The VRF method allows a record to be defined in terms of its content (from a set of available data elements) and the position of the data elements in the record. This method can be used to construct the 24-word standard formats (described in the next section), and custom formats. Even though the DEFINITY switches output the record in ASCII character representation, the terminology ‘‘word’’ is retained here to be compatible with the G2 defined records.

G3rV1, G3V2, and later releases have the same VRF capability as G2 in that it provides the flexibility to define the data presentation of the call record, along with its contents and layouts. That is, the record content, the spacing between the data items and the carriage return, line feed, and nulls can be defined. This does not require enforcing predefined data presentations on customer devices, but allows new devices to handle AT&T’s call detail records. This allows support of any data presentation, currently provided in G2, that is different from G1, thus allowing support of embedded base devices that handle the data presentation.

Standard Record Formats

Table 3-23 lists the standard record formats and indicates which switch/version supports each format, followed by a description of the formats. Specific field definitions are provided in the next section, "Call Detail Record Fields".

Table 3-23. Standard Fixed CDR Formats

CDR Record Fixed Formats Supported Format	Name	G3vs/ G3sV1/G1	G3iV1	G3rV1	G3i-Global	G3V2 and later releases
Older Formats for Early Vintage Switches and Adjuncts:						
18-word	TELESEER, non-ISDN	X	X	X	X	X
18-word	TELESEER, ISDN	X	X	X	X	X
18-word	59-Character, non-ISDN	X	X	X	X	X
18-word	Printer, non-ISDN	X	X	X	X	X
18-word	Printer, ISDN	X	X	X	X	X
Standard Domestic Formats:						
18-word	LSU, non-ISDN	X	X	X	X	X
18-word	LSU, ISDN	X	X	X	X	X
18-word	LSU-Expand					X
24-word	Expanded	X	X	X	X	X
24-word	Unformatted	X	X	X	X	X
Standard International Formats:						
Int'l	Processing Device				X	X
Int'l	Direct Output Device				X	X
Int'l 24-word	ISDN Expanded				X	X
Enhanced Formats:						
Enhanced	ISDN LSU					X
Enhanced	ISDN TELESEER					X
Enhanced	ISDN Printer					X
Enhanced	Expanded					X
Enhanced	Unformatted					X

NOTE:
The TELESEER and 94A LSU adjuncts are no longer available. So, although DEFINITY switches support the formats originally designed for the TELESEER and 94A LSU adjuncts, DEFINITY switches do not support using these adjuncts.

The following pages present a table for each CDR format presented in Table 3-23.

Table 3-24. CDR Data Format — TELESEER

ASCII Character Position	Data Field Description
01-03	Space
04	Time Hour- (tens)
05	Time Hour- (units)
06	Time Minute (tens)
07	Time Minute (units)
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Condition Code
13-15	Access Code Dialed[1]
16-18	Access Code Used[1]
19-33	Dialed Number[1]
34-38	Calling Number[1]
39-53	Account Code[1]
54	FRL
55	IXC
56-58	Incoming Circuit ID[2](tens, units, hundreds)
59-61	Outgoing Circuit ID[2](tens, units, hundreds)
62	Feature Flag
63-69	Authorization Code
70-76	Space
77	Carriage Return
78	Line Feed
79-81	Null

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

NOTE:
TELESEER CDR Units are no longer available.

Table 3-25. CDR Data Format — ISDN TELESEER

ASCII Character Position	Data Field Description
01-03	Space
04	Time Hour- (tens)
05	Time Hour- (units)
06	Time Minute (tens)
07	Time Minute (units)
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Condition Code
13-15	IXC[1]
16-18	Access Code Used[1]
19-33	Dialed Number[1]
34-38	Calling Number[1]
39-53	Account Code[1]
54	INS (units digit)
55	FRL
56-58	Incoming Circuit ID[2](tens, units, hundreds)
59-61	Outgoing Circuit ID[2](tens, units, hundreds)
62	Feature Flag
63-69	Authorization Code
70-71	INS (hundreds, tens)[1]
72-76	Space
77	Line Feed
78-80	Null

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-26. CDR Data Format — Enhanced TELESEER

ASCII Character Position	Data Field Description
1-3	Space
4-5	Time of Day-hours
6-7	Time of Day-minutes
8	Duration-hours
9-10	Duration-minutes
11	Duration-tenths of minutes
12	Condition Code
13-16	IXC Code[1]
17-19	Trunk Access Code Used[1]
20-34	Dialed Number[1]
35-39	Calling Number[1]
40-54	Account Code[1]
55	ISDN NSV (3rd digit)
56	FRL
57-59	Incoming Circuit ID[2] (tens, units, hundreds)
60-62	Outgoing Circuit ID[2] (tens, units, hundreds)
63	Feature Flags
64-70	Authorization Code
71-72	ISDN NSV (1st and 2nd digits)
73-76	Space
77	Carriage return
78	Line feed
79	3 null characters (indicates end of record)

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-27. CDR Data Format — 59 Character

ASCII Character Position	Data Field Description
01	Time-Hour (tens)
02	Time-Hour (units)
03	Time-Minute (tens)
04	Time-Minute (units)
05	Duration-Hour
06	Duration-Minute (tens)
07	Duration-Minute (units)
08	Duration-Minute (tenths)
09	Condition Code
10-12	Access Code Dialed[1]
13-15	Access Code Used[1]
16-30	Dialed Number[1]
31-35	Calling Number[1]
36-50	Account Code[1]
51	FRL
52	IXC
53-55	Incoming Circuit ID[2] (tens, units, hundreds)
56-58	Outgoing Circuit ID[2] (tens, units, hundreds)
59	Carriage Return
60	Line Feed
61-63	Null

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-28. CDR Data Format — Printer

ASCII Character Position	Data Field Description
01	Time Hour- (tens)
02	Time Hour- (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration Hour
07	Duration Minute (tens)
08	Duration Minute (units)
09	Duration Minute (tenths)
10	Space
11	Condition Code
12	Space
13-15	Access Code Dialed[1]
16	Space
17-19	Access Code Used[1]
20	Space
21-35	Dialed Number[1]
36	Space
37-41	Calling Number[1]
42	Space
43-57	Account Code[1]
58	Space
59-65	Authorization Code
66-69	Space
70	FRL
71	Space
72	IXC
73	Space

Continued on next page

Table 3-28. CDR Data Format — Printer — *Continued*

ASCII Character Position	Data Field Description
74-76	Incoming Circuit ID[2](tens, units, hundreds)
77	Space
78-80	Outgoing Circuit ID[2](tens, units, hundreds)
81	Space
82	Feature Flag
83	Carriage Return
84	Line Feed

Continued on next page

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-29. CDR Data Format — ISDN Printer

ASCII Character Position	Data Field Description
01	Time Hour- (tens)
02	Time Hour- (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration Hour
07	Duration Minute (tens)
08	Duration Minute (units)
09	Duration Minute (tenths)
10	Space
11	Condition Code
12	Space
13-15	IXC[1]

Table 3-29. CDR Data Format — ISDN Printer — *Continued*

ASCII Character Position	Data Field Description
16	Space
17-19	Access Code Used[1]
20	Space
21-35	Dialed Number[1]
36	Space
37-41	Calling Number[1]
42	Space
43-57	Account Code1
58	Space
59-65	Authorization Code[1]
66	Space
67-68	INS[1] (hundreds, tens)
69	Space
70	INS (units digit)
71	Space
72	FRL
73	Space
74-76	Incoming Circuit ID[2] (tens, units, hundreds)
77	Space
78-80	Outgoing Circuit ID[2] (tens, units, hundreds)
81	Space
82	Feature Flag
83	Carriage Return
84	Line Feed

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-30. CDR Data Format — Enhanced Printer

ASCII Character Position	Data Field Description
1-2	Time of Day-hours
3-4	Time of Day-minutes
5	Space
6	Duration-hours
7-8	Duration-minutes
9	Duration-tenths of minutes
10	Space
11	Condition Code
12	Space
13-16	IXC Code[1]
17	Space
18-21	Trunk Access Code Used[1]
22	Space
23-37	Dialed Number[1]
38	Space
39-43	Calling Number[1]
44	Space
45-59	Account Code[1]
60	Space
61-67	Authorization Code[1]
68	Space
69-71	ISDN NSV (1st and 2nd digits)
72	Space
73	FRL
74	Space
75-77	Incoming Circuit ID[2] (tens, units, hundreds)
78	Space
79-81	Outgoing Circuit ID[2] (tens, units, hundreds)
82	Space

Table 3-30. CDR Data Format — Enhanced Printer

ASCII Character Position	Data Field Description
83	Feature Flags
84	Carriage Return
85	Line Feed

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-31. CDR Data Format — LSU-Expand

ASCII Character Position	Data Field Description
01	Time Hours (tens)
02	Time Hours (units)
03	Time Minutes (tens)
04	Time Minutes (units)
05	Space
06	Duration-Hours (units)
07	Duration-Minutes (tens)
08	Duration-Minutes (units)
09	Duration-Minutes (tenths)
10	Space
11	Condition Code
12	Space
13-15	Access Code Dialed[1]
16-18	Access Code Used[1]
19	Space
20-34	Dialed Number[1]
35	Space
36-39	Calling Number[1]

Table 3-31. CDR Data Format — LSU-Expand

ASCII Character Position	Data Field Description
40	Space
41-45	Account Code[1]
46	Space
47-53	Authorization Code[1]
54	Space
55-56	Time in Queue[1]
57	Space
58	FRL
59	Space
60	Calling Number 1st digit
61	Space
62-63	Incoming Circuit ID[2] (tens, units)
64	Space
65	Feature Flag
66	Space
67-68	Outgoing Circuit ID[2] (tens, units)
69	Space
70	Outgoing Circuit ID[2](hundreds)
71	Space
72	Incoming Circuit ID[2] (hundreds)
73	IXC
74	Carriage Return
75	Line Feed
76-78	Null

1. Data is right justified and padded with blanks (spaces)
2. Data is right justified and padded with zeros.

Table 3-32. CDR Data Format — LSU

ASCII Character Position	Data Field Description
01	Duration-Hours (units)
02	Duration-Hours (tens)
03	Duration-Minutes (units)
04	Duration-Minutes (tenths)
05	Condition Code
06-08	Access Code Dialed[1]
09-11	Access Code Used[1]
12-26	Dialed Number[1]
27-30	Calling Number[1], [2] (second through fifth digits for five-digit dialing plan)
31-35	Account Code (First five Digits)[1]
36-42	Authorization Code or sixth through twelfth Digits of Account Code1
43-44	Time in Queue or thirteenth and fourteenth digits of account code[1]
45	FRL or fifteenth Digit of Account Code[1]
46	Calling Number (first digit of a five-digit calling number)
47-48	Incoming Circuit ID (tens, units)
49	Feature Flag
50-52	Outgoing Circuit ID[3] (tens, units, hundreds)
53	Incoming Circuit ID (hundreds)
54	IXC
55	Carriage Return
56	Line Feed
57-59	Null

1. Data is right justified and padded with blanks (spaces).
2. For a 4-digit dialing plan, this field records first four digits of calling number.
3. Data is right justified and padded with zeros.

Table 3-33. CDR Data Format — ISDN LSU

ASCII Character Position	Data Field Description
01	Duration-Hours
02	Duration-Minutes (tens)
03	Duration-Minutes (units)
04	Duration-Minutes (tenths)
05	Condition Code
06-08	IXC[1]
09-11	Access Code Used[1]
12-26	Dialed Number[1]
27-30	Calling Number[1] (second through fifth digits for five-digit dialing plan)
31-35	Account Code (first five digits)[1]
36-42	Authorization Code or sixth through twelfth Digits of Account Code[1]
43-44	INS or thirteenth and fourteenth Digits of Account Code[1]
45	INS (third Digit), FRL, or fifteenth Digit of Account Code[1]
46	Calling Number (first digit of five-digit calling number)
47-48	Incoming Circuit ID (tens, units)
49	Feature Flag
50-52	Outgoing Circuit ID[2] (tens, units, hundreds)
53	Incoming Circuit ID[2] (hundreds)
54	FRL
55	Carriage Return
56	Line Feed
57-59	Null

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-34. CDR Data Format — Enhanced LSU

ASCII Character Position	Data Field Description
1	Duration-hours
2-3	Duration-minutes
4	Duration-tenths of minutes
5	Condition Code
6-9	IXC Code
10-12	Trunk Access Code Used
13-27	Dialed Number
28-31	Calling Number[1]
32-35	Account Code (1st 4 digits)
36-42	Authorization Code or 6th-12th digits of Account Code
43-45	ISDN NSV
46	1st digit of a 5-digit Calling Number
47-48	Incoming Circuit ID (tens, units)
49	Feature Flags
50-52	Outgoing Circuit ID[2] (tens, units, hundreds)
53	Incoming Circuit ID[2] — 100ths digit
54	FRL
55	Carriage Return
56	Line Feed
57	3 Null Characters (indicates end of record)

1. Contains the 1st 4 digits of the Calling Number for a 4-digit dialing plan or the 2nd-5th digits for a 5-digit plan.
2. Data is right justified and padded with zeros.

Table 3-35. CDR Data Format — Expanded

ASCII Character Position	Data Field Description
01	Time Hours (tens)
02	Time Hours (units)
03	Time Minutes (tens)
04	Time Minutes (units)
05	Space
06	Duration-Hours (units)
07	Duration-Minutes (tens)
08	Duration-Minutes (units)
09	Duration-Minutes (tenths)
10	Space
11	Condition Code
12	Space
13-16	Access Code Dialed[1]
17	Space
18-21	Access Code Used[1]
22	Space
23-37	Dialed Number[1]
38	Space
39-48	Calling Number[1]
49	Space
50-64	Account Code[1]
65	Space
66-72	Authorization Code[1]
73	Space
74-75	Time in Queue[1]
76	Space
77	FRL
78	Space

Continued on next page

Table 3-35. CDR Data Format — Expanded — *Continued*

ASCII Character Position	Data Field Description
79-81	Incoming Circuit ID[1] (hundreds, tens, units)
82	Space
83-85	Outgoing Circuit ID[1] (hundreds, tens, units)
86	Space
87	Feature Flag
88	Space
89-90	Attendant Console[1]
91	Space
92-95	Incoming Trunk Group Access Code[1]
96	Space
97-98	Node Number
99	Space
100-102	INS[1]
103	Space
104-106	IXC[1]
107	Space
108	BCC
109	Space
110	MA-UUI
111	Space
112	Resource Flag
113	Space
114-117	Packet Count
118	Space
119	TSC Flag
120	Space
121	Reserved
122	Space

Continued on next page

Table 3-35. CDR Data Format — Expanded — *Continued*

ASCII Character Position	Data Field Description
123	Reserved
124	Space
125	Reserved
126	Space
127	Reserved
128	Space
129	Reserved
130	Space
131	Carriage Return
132	Line Feed
133-135	Null

Continued on next page

1. Data is right justified and padded with blanks (spaces).

Table 3-36. CDR Data Format — Enhanced Expanded

ASCII Character Position	Data Field Description
1-2	Time of Day-hours
3-4	Time of Day-minutes
5	Space
6	Duration-hours
7-8	Duration-minutes
9	Duration-tenths of minutes
10	Space
11	Condition Code
12	Space
13-16	Trunk Access Code Dialed[1]
17	Space
18-21	Trunk Access Code Used[1]
22	Space
23-37	Dialed Number[1]
38	Space
39-48	Calling Number[1]
49	Space
50-64	Account Code[1]
65	Space
66-72	Authorization Code[1]
73	Space
74-75	Time in Queue
76	Space
77	FRL
78	Space
79-81	Incoming Circuit ID[2]
82	Space
83-85	Outgoing Circuit ID[2]

Table 3-36. CDR Data Format — Enhanced Expanded

ASCII Character Position	Data Field Description
86	Space
87	Feature Flags
88	Space
89-90	Attendant Console Number
91	Space
92-95	Incoming Trunk DAC
96	Space
97-98	Node Number
99	Space
100-102	ISDN NSV
103	Space
104-107	IXC Code
108	Space
109	ISDN BCC
110	Space
111	ISDN MA-UUI
112	Space
113	Resource Flag
114	Space
115-118	Packet Count
119	Space
120	TSC Flag
121	Space
122-123	Bandwidth
124	Space
125-130	ISDN CC (1-6 digits)
131-135	ISDN CC (7-11 digits)/PPM Count (1-5)
136-146	Reserved for future use

Table 3-36. CDR Data Format — Enhanced Expanded

ASCII Character Position	Data Field Description
147	Carriage Return
148	Line Feed
149	3 Null Characters (indicates end of record)

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-37. CDR Data Format — Unformatted

ASCII Character Position	Data Field Description
01	Time Hour (tens)
02	Time Hour (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Duration-Hours
06	Duration-Minutes (tens)
07	Duration-Minutes (units)
08	Duration-Minutes (tenths)
09	Condition Code
10-13	Access Code Dialed[1]
14-17	Access Code Used[1]
18-32	Dialed Number[1]
33-42	Calling Number[1]
43-57	Account Code[1]
58-64	Authorization Code[1]
65-66	Space
67	FRL

Continued on next page

Table 3-37. CDR Data Format — Unformatted — *Continued*

ASCII Character Position	Data Field Description
68-70	Incoming Circuit ID[2] (hundreds, tens, units)
71-73	Outgoing Circuit ID[2] (hundreds, tens, units)
74	Feature Flag
75-76	Attendant Console[1]
77-80	Incoming Trunk Group Access Code[1]
81-82	Node Number
83-85	INS[1]
86-88	IXC[1]
89	BCC
90	MA-UUI
91	Resource Flag
92-95	Packet Count
96	TSC Flag
97-100	Reserved
101	Carriage Return
102	Line Feed
103-105	Null

1. Data is right justified and padded with blanks (spaces).
2. Data is right justified and padded with zeros.

Table 3-38. CDR Data Format — Enhanced Unformatted

ASCII Character Position	Data Field Description
1-2	Time of Day-hours
3-4	Time of Day-minutes
5	Duration-hours

Table 3-38. CDR Data Format — Enhanced Unformatted

ASCII Character Position	Data Field Description
6-7	Duration-minutes
8	Duration-tenths of minutes
9	Condition Code
10-13	Trunk Access Code Dialed
14-17	Trunk Access Code Used
18-32	Dialed Number
33-42	Calling Number
43-57	Account Code
58-64	Authorization Code
65-66	Time in Queue[1]
67	FRL
68-70	Incoming Circuit ID
71-73	Outgoing Circuit ID
74	Feature Flags
75-76	Attendant Console Number
77-80	Incoming Trunk DAC
81-82	Node Number
83-87	ISDN NSV
88-89	IXC Code
90	ISDN BCC
91	ISDN MA-UUI
92	Resource Flag
93-96	Packet Count
97	TSC Flag
98-99	Bandwidth
100-105	ISDN CC (1-6 digits)
106-110	ISDN CC (7-11 digits)/PPM Count (1-5)
111-114	reserved for future use

Table 3-38. CDR Data Format — Enhanced Unformatted

ASCII Character Position	Data Field Description
115	Carriage Return
116	Line Feed
117	3 Null Characters (indicates end of record)

1. The "Time in Queue" field is filled with 2 spaces in this record.

Table 3-39. CDR Data Format — Int Process

ASCII Character Position	Data Field Description
01	Format Code (first)
02	Format Code (second)
03	Time Hour - (tens)
04	Time Hour - (units)
05	Time Minute - (tens)
06	Time Minute - (units)
07	Duration Hour
08	Duration Minute - (tens)
09	Duration Minute - (units)
10	Duration Minute (tenths)
11	Space
12	Condition Code
13	Space
14-16	Access Code dialed
17-19	Access Code used
20	Space
21-38	Dialed Number (1st to 18th digit)[1,2]
39-43	Calling Number (1st - 5th digit)[2]
44	Space

Continued on next page

Table 3-39. CDR Data Format — Int Process

ASCII Character Position	Data Field Description
45-59	Account Code (1st - 15th digit)[2]
60	Space
61	IXC
62	FRL
63-65	Space
66-67	Incoming Circuit ID (1st - 2nd digit)[3]
68-70	Space
71-72	Outgoing Circuit ID (1st - 2nd digit)[3]
73	Space
74-78	PPM Count (1st - 5th digit)[2]
79	Carriage Return
80	Line Feed
81-83	Null

1. 21-23 are blank.
2. Data is right justified and padded with blanks (spaces).
3. Data is right justified and padded with zeros

Table 3-40. CDR Data Format — Int-Direct

ASCII Character Position	Data Field Description
01	Date of Month - (tens)
02	Date of Month - (units)
03	Month - (tens)
04	Month - (units)
05	Year - (tens)
06	Year - (units)
07	Space
08	Time Hour - (tens)
09	Time Hour - (units)
10	Time Minute - (tens)
11	Time Minute - (units)
12	Space
13	Duration Hour
14	Duration Minute - (tens)
15	Duration Minute - (units)
16	Duration Minute - (tenths)
17	Space
18	Condition Code
19	Space
20-22	Access Code dialed[1]
23-25	Access Code used[1]
26	Space
27-44	Dialed Number used[1,2]
45	Space
46-50	Calling Number[1]
51	Space
52-66	Account Code[1]
67	Space

Table 3-40. CDR Data Format — Int-Direct

ASCII Character Position	Data Field Description
68-72	PPM Count[1]
73	Space
74-75	Incoming Circuit ID[3]
76	Space
77-78	Outgoing Circuit ID[3]
79	Carriage Return
80	Line Feed

1. Data is right justified and padded with blanks (spaces).
2. 27-29 are blank
3. Data is right justified and padded with zeros

Table 3-41. CDR Data Format — Int-ISDN

ASCII Character Position	Data Field Description
01	Time - Hours (tens)
02	Time - Hours (units)
03	Time - Minutes (tens)
04	Time - Minutes (units)
05	Space
06	Duration - Hours (units)
07	Duration - Minutes (tens)
08	Duration - Minutes (units)
09	Duration - Minutes (tenths)
10	Space
11	Condition Code

Continued on next page

Table 3-41. CDR Data Format — Int-ISDN

ASCII Character Position	Data Field Description
12	Space
13-16	Access Code dialed[1]
17	Space
18-21	Access Code used[1]
22	Space
23-37	Dialed Number[1]
38	Space
39-48	Calling Number[1]
49	Space
50-64	Account Code[1]
65	Space
66-72	Authorization Code[1]
73	Space
74	Line Feed
75	Space
76	FRL
77	Space
78	Incoming Cir ID (hundreds)
79	Incoming Cir ID (tens)
80	Incoming Cir ID (units)
81	Space
82	Outgoing Cir ID (hundreds)
83	Outgoing Cir ID (tens)
84	Outgoing Cir ID (units)
85	Space
86	Feature Flag
87	Space
88	Attendant Console (1st digit)[1]

Continued on next page

Table 3-41. CDR Data Format — Int-ISDN

ASCII Character Position	Data Field Description
89	Attendant Console (2nd digit)
90	Space
91	Inc Trk Acc Code (1st digit)[1]
92	Inc Trk Acc Code (2nd digit)
93	Inc Trk Acc Code (3rd digit)
94	Inc Trk Acc Code (4th digit)
95	Space
96	Node Number (1st digit)[1]
97	Node Number (2nd digit)
98	Space
99	INS (1st digit)[1]
100	INS (2nd digit)
101	INS (3rd digit)
102	Space
103	IXC (1st digit)[1]
104	IXC (2nd digit)
105	IXC (3rd digit)
106	IXC (4th digit)
107	Space
108	BCC
109	Space
110	MA-UUI
111	Space
112	Resource Flag
113	Space
114	Reserved (1st digit)[1]
115	Reserved (2nd digit)
116	Reserved (3rd digit)

Continued on next page

Table 3-41. CDR Data Format — Int-ISDN

ASCII Character Position	Data Field Description
117	Reserved (4th digit)
118	Reserved (5th digit)
119	Reserved (6th digit)
120	Reserved (7th digit)/PPM (1st digit)
121	Reserved (8th digit)/PPM (2nd digit)
122	Reserved (9th digit)/PPM (3rd digit)
123	Reserved (10th digit)/PPM (4th digit)
124	Reserved (11th digit)/PPM (5th digit)
125-131	Space
132	Carriage return
133	Line Feed
134-136	Null

Continued on next page

1. Data is right justified and padded with blanks (spaces).

Call Detail Record Fields

The following list describes the CDR data collected for each call and the number of digits in each field. All information is right adjusted in the respective field, unless otherwise indicated.

— The CDR output for intraswitch CDR records contain only the time, duration, condition code, dialed number, calling number, the character "W'' for a Wideband call, the bandwidth values 1 through 31 for Bandwidth calls (only for Enhanced Unformatted, Enhanced Expanded, and customized formats), and optionally who disconnected first information (refer to the "FRL" field description) fields. The length of the field can be customized. For customized or customized formats, Vector Directory Number (VDN) is available for intraswitch calls.

— The field name for Customized (Variable) Record Formats for G3rV1, G3V2, and later releases are indicated for each field description.

- **Access Code Dialed** (up to three or four digits [24-word formats])

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘code-dial’’; the default length = 4.

  This field is used only for outgoing calls. This field can be the ARS access code, AAR access code, or the access code of a specific trunk group. This field does not exist in the ISDN 18-word record formats. In G3r, this field is also used to record the X.25 Feature Access Code of an outgoing X.25 addressed call. The field is labeled ‘‘access code dialed’’ in the record layouts.

  Intra-switch CDR will not output this field.

- **Access Code Used** (up to three or four digits [24-word formats])

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘code-used.’’

  This field is used only for outgoing calls and when the trunk group used is different than the access code dialed. It is not used when TAC is dialed. For example, most customers use FAC for ARS. This field contains the access code of the actual trunk group that the call was routed over. When the dialed and used access code are the same, this field will be blank (unless one of the ISDN record formats is used with ISDN, Enhanced, TELESEER, LSU, or Printer Only). In this case, the field always shows the access code of the used trunk group, even if it is the same as the dialed access code.

- **Account Code** (up to 15 digits)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘acct-code.’’

  This field is optional but can contain a number to associate call information with projects or account numbers. Account Codes must be prefixed with an access code which is either a fixed digit or a series of digits. The access code is administrable on the ‘Feature Access Code’ form. On outgoing calls, the account code access code must be dialed before the trunk access code, AAR access code, or ARS code. Information in this field is right adjusted. These account codes allow the System Manager to associate calling information with projects or account numbers. The access code is not recorded because it is not part of the CDR account code.

  Four digit IXCs will use one digit of the 15-digit account code on the Enhanced LSU format.

Account code dialing can be optional or mandatory (forced). Forced account code entry is set on a per-COR basis or per system. If the trunk group used for a call requires an account code and one is not dialed, the call is denied. Forced account code entry can also be assigned for all toll calls. Service calls, directory assistance calls, and WATS calls are excluded.

If the ISDN LSU 18-word format is used, a maximum of 12 account code digits may be output to the record. If the account code is longer than 12 digits, the least significant digits are dropped. If the Enhanced LSU format is used, only 11 digits can be used for account code output.

Intra-switch CDR will not output this field.

- **Attendant Console** (two digits) (24-Word Record Only)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ''attd-console.''

This field contains the attendant console number of the attendant that handled the call in a record that is marked as being attendant handled.

Intra-switch CDR will not output this field.

- **Authorization Code** (seven digits)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ''auth-code.''

This field contains the four- to seven-digit authorization code used to make the call. For non-ISDN and ISDN LSU formats, the authorization code is fewer than six digits in length. It is five for Enhanced LSU. On the 59-character record, the authorization code is never recorded.

Intra-switch CDR will not output this field.

- **Bearer Capability Class (BCC)** (one digit) (24-Word Record Only)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ''bcc.''

This field contains the BCC for ISDN calls, identifying the type of an ISDN call. It will distinguish between voice and different types of data. The BCC is a single digit. Either of the following digits may appear in this field.

- — 0 = Voice Grade Data and Voice
- — 1 = Mode 1 (56 kbps synchronous data)
- — 2 = Mode 2 (less than 19.2 kbps synchronous or asynchronous data)

— 3 = Mode 3 (64 kbps data for LAPD protocol)

— 4 = Mode 0 (64 kbps data clear)

— w = Wideband

Intra-switch CDR only outputs a value in this field for Wideband calls only and, in this case, the value is ‘‘w.’’

- **Bandwidth** (two digits)(24-word record and customized record for G3V2 and later releases only)

NOTE:
The field name for Customized CDR Records for G3rV1, G3v2, and later releases is “bandwidth.”

— Used to capture the bandwidth of the wideband calls to support H0, H11, H12, and N x 64 kbps data rates, where 2 <= N <= 31. For CDR record formats (Enhanced Expanded, Enhanced Unformatted and Variable) expressed as the number of DSOs of 64Kbps channels comprising a call and is stored in the two digit bandwidth field.

- **Calling Number** (up to five and 10 digits in 24-word format)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘calling-num.’’

For outgoing or intraswitch calls, this field contains the extension number of the originating voice terminal user. For incoming and tandem calls, this field contains the TAC of the trunk group used for the call (in the standard 18-word formats, this field is only four digits — the fifth digit, if used, is recorded in a separate field). (The fifth digit is the first digit of a 5-digit dialing plan). In formats where the field is less than 7-digits, this also shows the TAC of the incoming call, but in the 24-word format, the “Calling Number” field is 10 digits and contains the CPN/BN information on incoming ISDN calls. If the CPPN is not output, the field is blank for both Unformatted and Expanded. The “Calling Number” field contains the local extension of the NCA-TSC endpoint when the CDR record is for an outgoing (or originating) NCA-TSC. This field is blank for other NCA-TSC CDR records (that is, terminating, tandem, or unsuccessful). Information in this field is right adjusted.

- **Calling Number in TAC**

NOTE:
The field name for G3rV1 and G3V2 Customized CDR Records is ‘‘clg-num-in-tac.’’

If the Calling Number is related to SID/ANI, this field contains SID/ANI information; otherwise, this field contains TAC information. If an outgoing call type, this field will contain the calling extension. This field is specified on the CDR customized record.

- **Carriage Return** (one character)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘return.’’

  The ASCII carriage return character followed by a line feed is used to terminate CDR records.

- **Condition Code** (one character)

  These codes, reflect special events relating to the call. Condition codes for the 59-character CDR record are different from the codes in Table 3-43 but can be mapped to these codes as shown in Table 3-42:

Table 3-42. Condition Code Mapping

CONDITION CODE MAPPING FOR 59-CHARACTER RECORD	
59-Character Condition Code	**Code from Table 3-43**
A	1
D	4
E	6
G	7
H	8
I	9
L	C
N	E

Table 3-43 shows the actual condition codes.

Table 3-43. Condition Codes

Condition Codes	Description
0	Identifies an intraswitch call (a call that originates and terminates on the switch).
1	Identifies an attendant-handled call or an attendant-assisted call (except conference calls).
4	Identifies an extremely long call (10 hours or more) or an extremely high message count TSC (9999 messages or more). On a call exceeding 10 hours, a call record with this condition code and a duration entry of 9 hours, 59 minutes, and 1 to 9 tenths of a minute is produced after the first period. A similar call record with this condition code is produced after each succeeding 10-hour period. When the call does terminate, a final call record with a different condition code identifying the call type is produced.
6	Identifies calls that are not recorded because of resource exhaustion. A record with this condition code is only generated for calls that are routed to the attendant or calls that require CDR to overwrite records. This record includes the time and duration of the outage. Note: This condition code applies to G3r V1 and V2 only.
7	Identifies calls served by the AAR or ARS Selection feature.
8	Identifies calls which have been served on a delayed basis via the Ringback Queuing feature.
9	Identifies an incoming or tandem call.
A	Identifies an outgoing call.
B	Identifies an adjunct-placed outgoing call.
C	Identifies a conference call. For trunk CDR, a separate call record with this condition code is produced for each incoming or outgoing trunk serving the conference connection. The only voice terminal recorded for a conference call is the conference call originator. For intraswitch CDR, if the originator is optioned for intra-switch, each time the originator dials a non-trunk party a separate call record is produced with this condition code. If the originator is not optioned for intraswitch CDR, a separate record with this condition code is produced for each intraswitch party dialed.

Table 3-43. Condition Codes — *Continued*

Condition Codes	Description
E	An ineffective call attempt due to facilities not being available, such as all trunks are busy and either no queuing exists or the queue is full on an outgoing call, or the called voice terminal is busy or unassigned for an incoming call attempt. This also means an ISDN Call By Call Service Selection call was unsuccessful because of an administered trunk usage allocation plan. Incoming trunk calls to a busy terminal do *not* generate a CDR record.
F	Identifies an ineffective call attempt because of either insufficient calling privileges of the originator (assigned per FRL), ISDN calls rejected by the switch due to an NSF mismatch, or an authorization mismatch which prevents the completion of a data call.
G	Indicates a call terminating to a ringing trunk.
H	Indicates that a ringing call has been abandoned.
I	Indicates a call terminated to a busy trunk.

NOTE:
When more than one condition applies to a call, the overriding code is shown in Table 3-44.

When two condition codes apply on the same call, one will override the other. The matrix below, Table 3-44, defines the overrides. To illustrate how to use this matrix, assume that condition codes 7 and A apply to the same call. The matrix contains 12 horizontal rows (0, 1, 4, 6, 7, 8, 9, A, B, C, E, and F) and 12 vertical columns (0, 1, 4, 6, 7, 8, 9, A, B, C, E, and F). To find the condition code that overrides, look at the point of intersection between row 7 and column A. In this case, condition code 7 overrides. This can also be found by looking at the point where row A and column 7 intersect.

Table 3-44. Condition Code Override Matrix

	CONDITION CODE											
	0	**1**	**4**	**6**	**7**	**8**	**9**	**A**	**B**	**C**	**E**	**F**
0	NA	0	4	6	0	NA	NA	NA	B	C	NA	NA
1	0	NA	4	6	1	NA	9	1	B	C	E	NA
4	4	4	NA	6	4	4	4	4	4	4	NA	NA
6	6	6	6	NA	6	6	6	6	6	6	6	6
7	0	1	4	6	NA	7	9	7	B	C	E	F
8	NA	NA	4	6	7	NA	NA	8	B	C	E	NA
9	NA	9	4	6	9	NA	NA	NA	NA	C	E	F
A	NA	1	4	6	7	8	NA	NA	B	C	E	F
B	B	B	4	6	B	B	NA	B	NA	B	E	F
C	C	C	4	6	C	C	C	C	B	NA	NA	NA
E	NA	E	NA	6	E	E	E	E	E	NA	NA	NA
F	NA	NA	NA	6	F	NA	F	F	F	NA	NA	NA

- **Dialed Number** (up to 15 digits)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ''dialed-num.''

  This field contains the number dialed. If it is an outgoing call, the field contains the number dialed by a system user. If it is an incoming call, the field contains the extension that was dialed (or implied as in DNIS.) If more than 15 digits are dialed, the least significant digits are truncated. Intra-switch CDR will output this number.

  The # sign (''E" with standard 18-word formats) may be printed in this field in the following cases for both ARS and TAC calls. To eliminate the # from the CDR record, refer to "Remove # from CDR Record" on page 3-378.

  - When the user dials a feature access code that starts with a #
  - When the user dials # at the end of digit dialing (for example, for WATS and IDDD calls)
  - If an outgoing call experiences an interdigit timeout interaction on the ARS Analysis table

- If CDR Privacy is enabled for the calling number (this feature is available on a per station basis and is administered on the 'Station' form) and this is an outgoing call (not an incoming or intraswitch call). In this case, the trailing digits of the dialed number is blanked in the CDR for the call. If more than 15 digits are dialed, the dialed number will first be truncated to 15 digits, then the appropriate number of digits is blanked. The number of blanked digits is administered system-wide on the 'System Parameters CDR' form.

NOTE:
When a Trunk Access Code (TAC) is successfully dialed for a Look Ahead Interflow (LAI), the CDR record at the sending end has in the "Dialed Number" field of the outgoing CDR a # or E character depending on the output format. The Unformatted and LSU formats output an E and all other formats output a #. For example: A successful LAI to <TAC> 1001 where 1001 is the remote VDN extension will yield `1001E` or `1001#` in the "Dialed Number" field of the CDR record.

The "#" or "E" are used by the vector processing software for end of dialing digit.

For an outgoing (or originating) NCA-TSC or tandem NCA-TSC, this field contains the dialed digits used to establish a route to a far-end PBX. It contains the extension of the local extension used as the NCA-TSC endpoint when it is for a terminating NCA-TSC. For an unsuccessful NCA-TSC, this field is blank.

- **Duration** (four digits)
  Calls are rounded down in 6-second increments. Therefore, a call of 5-second duration will be indicated as 0 duration. A call indicated as 9999-second duration is a special sequence that indicates calls in progress during a time change made in the switch.

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is "duration."

All calls are timed. The timing is recorded in hours (0 through 9), minutes (00 through 59), and to the nearest tenth of a minute (0 through 9).

A 'G3V4 Customized CDR' form option allows the call duration to be reported in hours/minutes/seconds with no truncation to tenths of minutes.

- **FRL** (one digit)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is "frl."

FRLs, numbered zero through seven, are associated with the AAR and ARS features and define calling privileges. The information contained in this field is as follows:

— If the call is an outgoing call and an authorization code is not used to make the call, this field contains the originating voice terminal user's FRL.

— If the call is an outgoing call and an authorization code is used to make the call, this field contains the FRL associated with the dialed authorization code.

— If the call is an incoming or tandem call, this field contains the FRL assigned to the incoming trunk group.

— If the call is an incoming tandem tie trunk call, this field contains either the FRL assigned to the tandem tie trunk or the Traveling Class Mark (TCM) sent with the tandem tie trunk call, depending on which was used to complete the call. On ISDN calls, this field always contains the TCM, if it was received.

— The CDR System Parameters can be administered to have ''Disconnect Information in Place of FRL.'' For trunk CDR, the following call disconnect data is printed in this field in place of the FRL data:

Data	Meaning (for calls)
0	Don't know who dropped first
1	We dropped first
2	The CO dropped first
3	Maintenance got the trunk

For intraswitch CDR, if ''Disconnect Information in Place of FRL'' has been administered, the following call disconnect data is printed in this field in place of the FRL data (which is not output for intraswitch CDR):

Data	Meaning (for calls)
0	Indeterminate
1	calling number dropped first
2	dialed number dropped first

Indeterminate refers to all conference and transfer calls or any other call where it could not be determined who dropped first.

- **Feature Flag** (one digit)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘feat-flag’’.

  Intra-switch CDR will not output this field.

  The digit in this field indicates whether or not the switch has received answer supervision from the network and whether the call was a voice or data call (Data Call CDR):

  — A 4 in this field indicates a voice call with network answer supervision.

  — A 0 in this field indicates a voice call without network answer supervision.

  — A 5 in this field indicates a data call with network answer supervision.

  — A 1 in this field indicates a data call without network answer supervision.

  Answer Supervision is indicated for non interworked ISDN calls, E&M trunks (digital or analog), Ground Start trunks with battery reversal, calls placed by adjuncts over any of these trunks (for example, OCM), and calls that received data modem answer tone.

  The answer supervision flag is interpreted as follows:

  — For ISDN trunks, if the “Answer Supervision” field contains a 0, the call interworked with non-ISDN trunks and the duration was calculated but does not have the degree of accuracy of a strictly ISDN call. Thus, this field shows whether the call was interworked or went through a strictly ISDN network.

  — If the “Answer Supervision” field contains a 4 or 5, the call went over a strictly ISDN network and the duration marked is accurate.

  — For non-ISDN CO, FX, and WATS trunks that have the “Answer Supervision” field marked with a 4 or 5, and are receiving answer supervision from the network, the duration is accurate.

  — For Tie, Tandem and Access trunks that have the “Answer Supervision” field marked with a 4 or 5, the duration can only be assumed to be accurate if the PBX in question is the ‘‘network egress’’ PBX in a private network or a stand alone PBX. The duration is also accurate if all of the trunks the call goes over provide answer supervision.

  When the call duration is not accurate (a 0 appears in the “Answer Supervision” field), the calls have often been timed via an administered timeout on a switch or from an earlier point in the call than when they actually got answered, because the switch could not determine when the call was answered.

Calls are flagged as data calls if they use a conversion resource (such as a modem pool) and/or originate or terminate on a data module.

With G3V4 and later releases, the feat-flag bit on the CDR record can be administered to reflect whether an outgoing ISDN call was reported as interworked by the network. Prior to G3V4 interworked calls were treated as if no network answer was received. With the G3V4 option active, the feat-flag indicates that the call received network answer, but that the call was interworked in the network. In addition, the call duration starts at the point of receiving the network answer, not when the Answer Supervision Timer expired.

- **Format Code** (two digits)

  This field contains two values: 00 is no PPM; 03 denotes a PPM count in the digits record.

- **Incoming TAC** (four digits) (24-Word Records Only)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘in-trk-code.’’

  This field contains the access code of the incoming trunk group.

  Intra-switch CDR will not output this field.

- **INS** (three digits)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘ins.’’

This field specifies the INS requested for a call. This field applies only to ISDN calls. Each Network Specific Facility is translated into an INS according to Table 3-45.

Table 3-45. Network Specific Facility to INS Mapping

Network Specific Facility	INS Value
Network Operator	324
Presubscribed Common Carrier Operator	325
Software Defined Network (SDN)	352
MEGACOM 800	353
MEGACOM	354
INWATS	355
Maximum Banded WATS	356
AT&T Long Distance Service	358
ACCUNET Digital Service	357
OUTWATS Band 0	33
OUTWATS Band 1	34
. . .	. . .
OUTWATS Band 255	288
International 800	359
Multiquest	367

Intra-switch CDR will not output this field.

- IXC Code (one digit hexadecimal representation) (three or four digits with an ISDN format)

  — Non-ISDN Formats

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘ixc-code.’’

IXC codes, numbered one through 15 (1 through F hexadecimal), are associated with the AAR and ARS features and depict the carrier used on the call. This information is sent to the CDR output device in ASCII code as a hexadecimal representation (for example, ASCII “F” equals ‘‘15’’).

An IXC access number is used to access a specific common carrier for a call. In the US, this number is of the form 10XXX, 950 — 1XXX, or NXX — XXXX, where N is any digit 2 through 9 and X is any digit 0 through 9. The IXC access numbers applicable at a given location are associated with an IXC code on the ‘IXC’ form. When ARS is used, and a routing pattern inserts one of the administered IXC codes, the associated IXC code is recorded. If no IXC access number is used, a 0 is recorded. In this case, either an IXC carrier is not used on the call or the carrier is selected at the CO. A one-character index (1-15 and through F hexadecimal) corresponding to the administered IXC code’s index in the first page of the ‘IXC’ form is generated when that matching IXC code is used on an ARS call. If none of the administered IXC codes is used, a zero is recorded.

IXC codes on the first page of the ‘IXC’ form can be:

- Any number matching the second page of the ‘IXC’ form
- A 7-digit number of the form 950-XXXX, where “X’’ is any digit 0-9 or...
- Any 8 to 11-digit number

— ISDN Formats

With an ISDN record format, this field is a three or four-digit (for Enhanced) field that identifies the actual IXC used on an ISDN call. This information is determined from the routing pattern administration. On AAR and ARS calls, the three-digit IXC value is administered in the routing pattern for all ISDN calls. If a user dials an IXC code with a 10XXX format as administered on the ‘IXC Codes’ form, the CDR device will put only the last three digits (four for Enhanced) in the CDR record. If a user dials a seven-digit IXC code, this field will contain a zero.

Intra-switch CDR will not output this field.

- **Incoming Circuit Identification** (three digits)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘in-crt-id.’’

This field contains the member number of a trunk within a trunk group used for an incoming call. For outgoing calls, this field is blank. Tandem calls contain both incoming and outgoing circuit id-numbers.

Intra-switch CDR will not output this field.

- **Incoming Ring Interval Duration** (G3V4 and later releases)

  With G3V4 and later releases each trunk group can be administered so that CDR will record the ring time to answer or abandon for incoming calls originated by the trunk group. In addition, CDR will record if the incoming destination is busy. This record is separate from the normal call duration record printed for an answered call.

  When an incoming call originated by a trunk group with this option set is terminated to an internal destination, the call is tracked from the time ringing feedback is given to the originator. If the call is answered, a CDR record is printed with the condition code “G” and the duration reflects the time between the start of ringing and the answer of the call. If the call is abandoned before being answered, a CDR record is printed with the condition code “H” and the duration reflects the time between the start of ringing and the time the call was abandoned. If the destination is busy, a CDR record is printed with the condition code “I” and a duration of 0.

- **Line Feed** (one character)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘line-feed.’’

  The ASCII line feed character followed by a carriage return is used to terminate CDR records.

- **MA-UUI** (one digit)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, later releases is ‘‘ma-uui.’’

  MA-UUI is shown in the field which keeps track of the number of ISDN messages containing user data sent on an outgoing call. Data in this field can range from 0 to nine and is found only on 24-word records.

  Intra-switch CDR will not output this field.

- **Node Number** (two-digits) (24-Word Records Only)

  NOTE:
  The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ‘‘node-num.’’

  This field identifies the DCS node number of a switch within a DCS arrangement. The number output is the same as the node number on the ‘Dial Plan’ form (the local PBX id).

  Intra-switch CDR will not output this field.

- **Null** (one character)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is "null."

The NULL is used to terminate and divide CDR Records (usually in triplets) when needed by the receiving adjunct.

- **Outgoing Circuit Identification** (three digits)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is "out-crt-id."

For outgoing calls, this field contains the member number of the trunk within a trunk group used. This field is blank for incoming calls. Tandem calls include both incoming and outgoing circuit id numbers. For outgoing and tandem NCA-TSCs, this field contains the signaling group used to carry the NCA-TSC.

Intra-switch CDR will not output this field.

- **Packet Count** (four digits)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is "tsc_ct."

For ISDN TSCs, this field contains the number of ISDN-PRI USER INFOrmation messages sent, received, or (for tandem TSCs) passing through the switch.

Intra-switch CDR will not output this field.

- **Periodic Pulse Metering**

NOTE:
These enhancements are available with G3i-Global, G3V2, and later releases.

The CDR output interface has been modified to include three new CDR record formats: Int-Direct, Int-Process, and Int-ISDN. A new International 24-word ISDN Expanded Record has been added. The formats are identical to the TELESEER CDR Unit and printer formats used in International System 75 (IR1V2) and provide Periodic Pulse Metering (PPM) pulse counts in the output record. (PPM is available with G3i-Global, G3V2, and later releases.)

NOTE:
The field name for Customized CDR Records is "ppm."

The new CDR output formats are selected using System Parameters. PPM uses pulses transmitted over the trunk line from the serving CO at periodic intervals during the course of the outgoing call to determine call charges.

NOTE:
PPM is only provided by the central office in some countries; in particular, not in the United States.

Each pulse has an intrinsic value; at the end of the call, the sum of the pulses represents the charges for the call. The more expensive the call, the faster the sending rate of the pulses becomes.

The TN465B Loop Start CO Trunk circuit pack is capable of detecting

16 kHz PPM pulses. The pulses can occur at a maximum of once every 1.4 seconds during the call or, if the CO accumulates the total PPM count, they can occur up to three to five seconds after the call is dropped. In order to ensure that delayed PPM data is received, the Outgoing Glare Guard Timer on the trunk group should be adjusted accordingly. The PPM count may be output to `Int-Direct`, **`Int_Process`**, and `Int_ISDN`.

PPM pulse detection is applicable for outgoing calls on CO, DIOD, The "PPM" field on the 'Trunk Group' form should be set to ''y'' (yes) to enable the PPM detection and reporting functions.

- **Resource Flag** (one digit) (24-Word Records Only)

NOTE:
The field name for Customized CDR Records for G3rV1, G3V2, and later releases is ''res_flag.''

This is a four bit field that exists in the 24-word and customized record for G3xV2 only. Two bits of the field are used presently. The rest of the bits are reserved for future use. This field also applies to intraswitch calls. The new bits are defined as follows:

— The ''Circuit or Packet Switched Call'' bit is used to identify whether the call was circuit switched or packet switched. A ''0'' value in this bit indicates that the call used circuit switching. A ''1'' value in this bit indicates that the call used packet switching.

— The ''Conversion Devices Used'' bit indicates whether or not the call used a conversion device. A ''0'' value in this bit indicates that the call did not use a conversion device. A ''1'' value in this bit indicates that call used a conversion device.

— G3V1 has a 4-bit "Resource Flag" field of which only 1 bit is used to indicate whether or not a conversion device was used on the call

(0 means no conversion device was used, and 1 means a conversion device was used.

— For both releases the bits are interpreted as follows:

Intra-switch CDR will not output this field.

- **Space** (one to forty characters)

  NOTE:
  The field name for Customized CDR for G3rV1, G3V2, and later releases is ‘‘space.’’

  The ASCII blank character is used to separate other CDR fields or to blank fill unused record locations. This field defaults to one character length in customized records but can be incremented in size up to forty. The ASCII carriage return character followed by a line feed is used to terminate customized records.

- **TSC Flag** (one digit)

  NOTE:
  The field name for G3rV1 and G3V2 Customized CDR Records is ‘‘tsc_flag.’’

  This field distinguishes CDR records pertaining to TSCs. When not equal to zero, this field will indicate the status of the TSC. Table 3-46 presents the TSC Flag encoding.

  Intra-switch CDR will not output this field.

Table 3-46. Encoding for CDR TSC Flag

Encoding	Meaning
0	Circuit-switched call without TSC requests.
1	Reserved for future use.
2	Reserved for future use.
3	Reserved for future use.
4	Call Associated TSC requested, accepted in response to SETUP, no congestion control (applicable to originating node). Call Associated TSC received and accepted via SETUP, no congestion control (applicable to terminating node).
5	Call Associated TSC received and accepted via SETUP, congestion control (applicable to terminating node).
6	Call Associated TSC requested, accepted after SETUP, no congestion control (applicable to originating node). Call Associated TSC received and accepted after SETUP, no congestion control (applicable to terminating node).
7	Call Associated TSC received and accepted after SETUP, congestion control (applicable to terminating node).
8	Call Associated TSC requested, rejected (rejection came from outside the local switch).

Table 3-46. Encoding for CDR TSC Flag — *Continued*

Encoding	Meaning
9	Call Associated TSC requested, rejected (rejection came from the local switch, that is, lack of resource).
A	Non Call Associated TSC received, accepted, no congestion control (applicable to terminating node). Non Call Associated TSC received, accepted, no congestion control (applicable to terminating node).
B	Non Call Associated TSC requested, accepted, congestion control (applicable to originating node). Non Call Associated TSC received, accepted, congestion control (applicable to terminating node).
C	Non Call Associated TSC requested, rejected (rejection came from outside the local switch).
D	Non Call Associated TSC requested, rejected (rejection came from the local switch, that is, lack of resource).
E	Reserved for future use.
F	Reserved for future use.

- **Vector Directory Number** (five digits)

  NOTE:
  The field name for G3rV1 and G3V2 Customized CDR Records is ‘‘vdn.’’

  This field is only available on the customized record format. The CDR record will output the VDN extension number.

CDR Output Devices

The following adjuncts are available:

- CAS Plus
- Cost Allocator
- CDRP
- CDRU-NCR
- CDRU-S
- CDRU-SE

A system can have two output formats.

A standard 232C interface is provided by the system’s processor circuit pack. This allows for direct connection of the CDR output device to the system. If this port is not used, additional interface equipment is required as described in the Hardware and Software Requirements part of this feature description. The RS232 interface is only available with G3s/vs/i. The G3r must use additional equipment. The system can support two CDR output devices. One of these devices can use the direct EIA-232C connection; the other device will require the additional interface equipment.

When a system has two CDR output formats, one format is administered as the primary CDR output format; the other format is administered as the secondary CDR output format. The secondary output format is typically used for a local storage format (CDRU) to provide CDR data to NCOSS for assessing network performance or helping to find network problems.

The primary and secondary ports work independently. Each port will work even if the link to the other port is down. If a link is down for more than a minute, some data may be lost. However, the most recent 300 (G3i), or 1,900 (G3r) records are stored for the primary port even when a loss of records occurs. When the link comes back up, these records are output on a first-in, first-out basis.

The following information applies to the port used for the Secondary CDR output device:

- Data going to the secondary port should be the same as that going to the primary port. However, CDR records sent to the secondary port can only be in the 18-Word CDRU format, Unformatted, INT-Direct and INT-Process.

  NOTE:
  If the CDR buffer is full, G3r provides a call record handling option to select which of the following occurs:

  — Calls are blocked with a reorder tone

  — Calls overwrites records

  — Calls are routed to an attendant as non-CDR calls

  G3i and G3vs/G3s overwrite the CDR records.

- If the system experiences problems in sending records to the primary CDR Output Device, the system discontinues sending records to the secondary port for two minutes. The secondary port should be run at the highest possible speed in order to prevent loss of information on the port.

- If more than 300 (G3i/G3vsV1/G3sV1), 300 (G3iV2), or 1,900 (G3r) have not been sent to the primary CDR port, the secondary port is busied out for two minutes. This makes system resources available to send data to the primary CDR port before the data is lost. The system will continue to busy out the secondary port for two-minute intervals until less than 200 records (1800 for G3r) remain to be sent to the primary port.

  The system can store up to 300 (G3i), or 1,900 (G3r) CDR records which are sent to the output devices.

A 1,200 bits/second rate may only be used over the cable for the TELESEER CDR unit, 94A LSU, or printer, if the system line size is less than 1,000. If the system line size is greater than 1,000, a rate of at least 2,400 must be used with the 18-word CDR records. If the system line size is greater than 600, a rate of at least 2,400 must be used with the 24-word CDR records.

The time stamp on calls recorded by CDR is applied at the end of the call.

Remove # from CDR Records

One method of removing the # sign from CDR records is to eliminate the ARS/AAR analysis patterns requiring the inter-digit timeout to determine the pattern. For example, to remove the # from local directory assistance calls (411), delete the ARS analysis for the single digit "4" in the string. Until the inter-digit timeout occurs, the software does not know which pattern to match. By eliminating entry 1 (in this case), the inter-digit timeout is no longer required for 411.

Considerations

Capacities

The system can store up to 300 (G3i), or 1,900 (G3r) CDR records for the primary port, which (when the link is down) are sent to the output devices when the link comes back up.

Account Code Recording

With forced entry or optional entry of the Account Code feature (FEAC), a CDR record is generated for a particular account number. The CDR access code (for example, **6**) and the account number is dialed before the ARS, AAR, or TAC and called number are dialed.

If the attendant is extending a call to a voice terminal, an account code can be dialed before the extension number is dialed.

Voice terminal users cannot dial an account code when transferring a call to another voice terminal, unless they have console permissions. However, a voice terminal user transferring a call to a trunk can dial an account code before dialing the ARS or TAC.

CDR Device Baud Rate Limits

If the system line size is greater than 1,000, the CDR device must support a baud rate of at least 2,400 bps when using 18-word CDR records. If the system line size is greater than 600, the CDR device must support a baud rate of at least 2,400 bps when using 24-word CDR records.

Interactions

The following interaction discussions assume CDR is activated.

- **Abbreviated Dialing**

  When Abbreviated Dialing or a Facility Busy Indication button is used to make or complete a call, all digits outpulsed (up to a maximum of 15) will appear on the CDR record.

- **Answer Detection**

  G3 provides Answer Detection using the TN744 Call Classifier circuit pack. This feature is assigned as an option per system and requires sufficient (8-port) TN744 circuit packs. The circuit packs must be engineered to handle the expected outgoing calls from end of outpulsing until either answer is detected by the TN744 or actual Answer Supervision is received from the trunk. The time the answer is detected by the TN744 is used as the start of a call to record call duration in the CDR record.

- **Attendant Console**

  If an attendant-assisted call involves an outgoing trunk, the primary extension of the voice terminal user requesting attendant service is recorded as the calling number, even if the attendant dialed the outside number. Condition Code 1 indicates the call was assisted by the attendant.

  If the attendant allows through dialing, the primary extension of the voice terminal user who dialed the number is recorded as the calling party. Condition Code 1 indicates that a trunk access code was extended by the attendant. Condition Code 7 indicates that a feature access code was extended by the attendant.

  On attendant-assisted calls that require an account code, the account code must be entered before the trunk access code.

  If the attendant is redirecting an incoming call to a voice terminal, the attendant may dial an account code before dialing the extension number.

  It is not possible to option the attendant for intraswitch calls. -Intraswitch records are produced for an intraswitch optioned extension calling the attendant or for a call from the attendant to an intraswitch optioned extension. In the case of an attendant-assisted call involving an intraswitch extension, the calling number recorded is the extension of the party who called the attendant, and the dialed number recorded is the extension that the attendant extended the call to. The record will have a Condition Code 0.

- **AUDIX**

  For remote AUDIX over DCS, if station A on node 1 forwards its calls to AUDIX on node 2, the CDR record is produced on each switch. The record from node 1 contains A as the dialed number. The record from node 2 contains AUDIX as the dialed number. If the calling number is on a different switch within the DCS network, or the call comes in over ISDN, the actual calling number is recorded in the “Calling Number” field, and the TAC of the trunk bringing the call into the local switch is recorded in the “Incoming Trunk Access Code” field of 24-word records. If the forwarded call is an incoming call, then, as in all cases (other than vectoring) in which an incoming call is forwarded, transferred, or conferenced using an outgoing trunk, two separate CDR records are produced, one for incoming and one for outgoing trunk usage. The outgoing trunk usage record lists AUDIX as the Calling Number.

- **Authorization Codes**

  Authorization codes are recorded on CDR records if account codes do not exceed five digits for non-ISDN and ISDN LSU formats and four digits for Enhanced LSU formats. On the 59-character CDR International Processing and International Direct records, the authorization code is never recorded.

- **AAR and ARS**

  CDR records the following information for Automatic Route Selection (ARS):

  - — Fact that an ARS call was made
  - — Calling extension number
  - — FRL of the calling extension
  - — Called number
  - — Type of trunk group used for the ARS call
  - — Time of call completion
  - — Call duration (how long the parties talked)
  - — IXC code, if any

  If CDR is suppressed for the trunk group actually used on an ARS call, an CDR record is not generated; otherwise, Condition Code 7 applies. The ARS access code is recorded in the “Access Code Dialed” field and the trunk access code for the trunk group actually used is recorded in the “Access Code Used” field.

  If an AAR call is placed to a busy trunk group and CDR is suppressed for that trunk group, the user hears reorder tone and the CDR output shows an ineffective call attempt.

  If an ARS call is an attendant assisted call (a voice terminal user calls the attendant, the attendant dials the ARS access code, and then releases the call), the CDR record shows the call with a Condition Code of 7 (ARS call) instead of a Condition Code of 1 (attendant assisted call). This occurs because CDR is not notified until after the trunk is seized and, in this case, the trunk is not seized until the voice terminal user dials the number.

- **Automatic Callback**

  When the Automatic Callback feature is used for an intraswitch call, no CDR record will be generated for the first call attempt or the ringback. However, if the caller or extension being called is optioned for intraswitch CDR, a record of the actual call will be output provided the call is answered and completed.

- **ACA**

  ACA calls will generate intraswitch CDR if the terminating extension is monitored by CDR. The originating extension for ACA calls cannot be administered for intraswitch monitoring.

- **Automatic Call Distribution (ACD)**

  Either the hunt group extension number or individual hunt group member extension number (depending on administration) is recorded as the called number.

- **Automatic Wakeup**

  No CDR intraswitch records are generated for wakeup calls.

- **Bridged Call Appearance**

  CDR does not record any information on the party who bridges onto a call. Instead, the number that was called appears in the “Dialed Number” field of the CDR record. The duration of the call is recorded when the last party drops off the call. This also applies for intraswitch calls.

  If the calling user originates a call using a bridged appearance, CDR records the calling number of the bridged appearance extension and not the extension number of the original, calling station.

- **Call-By-Call Service Selection**

  When a successful call is made on a Call-By-Call Service Selection trunk, the network specific facility used on the call is translated into an INS number and recorded in the “INS” field of the CDR record. If a Call-By-Call Service Selection call is unsuccessful because of an administered trunk usage allocation plan, the INS number is recorded in the “INS” field of the report with a condition code of “E.’’

- **Call Coverage**

  When an incoming or intraswitch call is answered by a covering voice terminal, the extension number dialed by the originating party is recorded as the dialed number.

- **Call Forwarding All Calls**

  When a call is forwarded to another voice terminal, the extension number dialed by the calling party is recorded as the dialed number.

  There will be only one record generated for a forwarded intraswitch call. In this record, the dialed number will be the same as the extension dialed by the originating party.

If a DSI or ISDN-PRI trunk calls a station and the station is forwarded to another T1 trunk, the outgoing record will show the station being forwarded to the second T1 trunk rather than the first T1 trunk being forwarded to the second T1 trunk.

- **Call Park**

  When a voice terminal user parks an incoming or intraswitch call, that user's extension is recorded as the dialed number in the CDR record. Call duration in CDR reflects the entire time the incoming trunk is busy (incoming) or until the call ends (intraswitch).

- **Call Pickup**

  When an incoming or intraswitch call is answered by another voice terminal user in the pickup group, the extension number dialed by the calling party is recorded as the dialed number.

- **Call Vectoring**

  The 'CDR-related System Parameters' form can be administered so that the VDN extension will be used in place of the Hunt Group or Member extension. If administered to do so, this overrides the ''Call to Hunt Group - Record'' option of CDR for incoming Call Vectoring calls.

  For incoming calls to a VDN, the duration of the call is recorded from the time answer supervision is returned.

  — If answer supervision is returned by the vector (via an announcement, collect, disconnect, or wait with music command), and the call never goes to another extension, then the VDN extension is recorded as the called number in the CDR record.

  — If the call terminates to a hunt group, then the VDN, hunt group, or agent extension is recorded as the called number as per the administration discussed above.

  — If the call terminates to a trunk, then the following two CDR records will be generated:

    1. An incoming record with the VDN as the called number and the duration from the time answer supervision was provided to the incoming trunk.

    2. An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.

  Outgoing vector calls generate ordinary outgoing CDR records with the originating extension as the calling number.

  If "member extensions" is administered on the 'CDR System Parameters' form and the call successfully completes to a station via the "route-to" command, the CDR record will show an incoming call to that station.

  No Ineffective Call Attempt records will be generated for Call Vectoring `route to` commands that are unsuccessful.

If a vector interacts with an extension or group that has Call Forwarding All Calls active, normal Call Forwarding/CDR interactions apply.

Some calls may originally look like intraswitch calls, but result in trunk calls (for example, a call from a station administered for intraswitch CDR to a VDN, which ends up an outgoing call on an outgoing trunk). Such calls will not generate intraswitch CDR records; the CDR record will have a condition code A - outgoing.

- **Call Waiting Termination**

  Call duration timing starts when the voice terminal answers an incoming call.

- **CAS**

  If a CAS attendant extends a call for a user, and CDR is not assigned to the RLT trunk group, the user's extension is recorded as the originator of the call. If the RLT trunk group does have CDR administered, the RLT trunk is recorded. If a CAS attendant answers a call but does not extend the call, no CDR records are made.

- **CO Trunks**

  All incoming and outgoing calls on a CO trunk group will be recorded, if CDR is assigned and incoming calls are recorded.

- **Conference**

  For the purpose of CDR recording, a call is considered a conference call if it contains at least one trunk which is eligible for CDR recording plus two or more non-attendant parties, or if it contains at least one party optioned for intraswitch CDR. Condition Code C applies to each CDR record made for a conference call.

  For a conference call, a separate CDR record is produced for each outgoing/incoming trunk serving the conference call.

  For the outgoing portion of a conference call involving multiple voice terminals, the voice terminal which requested outside dial tone to bring an outside party into the conference is recorded as the calling party.

  For the outgoing/incoming portion of a conference call, the call duration in CDR reflects the entire time the trunk was on the conference call.

  Trunk-to-trunk transfer calls are treated like conference calls for CDR purposes. A separate CDR record is produced for each trunk used in a trunk-to-trunk transfer.

  If the originator of the conference call is optioned for intraswitch CDR, each time the originator dials a non-trunk party, a new CDR record is started. For example, Station A is optioned for intraswitch CDR and calls Station B. Station A conferences in Station C. Station A drops from the call. Station B or C drops from the call. Two CDR records are output with Condition Code C: one for the A to B call and one for the A to C call.

If the originator is not optioned for intraswitch CDR, but one or more parties brought into the conference are, one record with Condition Code C is generated for each dialed intraswitch party. For example, Station A calls Station B, which is optioned for intraswitch CDR. Station A conferences Station C. Station A drops from the call. Station B or C drops from the call. One CDR record is output with condition code C for the A to B call.

Intra-switch conference call CDR records are output when both the calling number (originator) and dialed number (terminator) of the call drop. The duration of the call will be from the time the terminator answers until both the originator and terminator drop from the call.

If the attendant originates the conference, only the dialed numbers corresponding to intraswitch optioned extensions will stimulate the creation of CDR records.

- **DCS**

  If the calling number is on a different switch within the DCS network, the actual calling number will be recorded in the “Calling Number” field, and the TAC of the trunk bringing in the call will be recorded in the “Incoming Trunk Access Code” field. DCS signaling messages do not generate CDR records.

- **DDC and UCD**

  Either the hunt group extension number or individual hunt group member extension number (depending on administration) is recorded as the called number.

- **DID**

  All incoming calls on the DID trunk group will be recorded if administered to record incoming CDR and if CDR is administered for this trunk group.

- **Emergency Access to the Attendant**

  No intraswitch CDR records will be generated for Emergency Access calls.

- **FX Trunks**

  NOTE:
  All calls made on an FX trunk group are recorded if administered to record CDR and if CDR is administered for this trunk group.

- **Expert Agent Selection (EAS)**

  A logical extension can be assigned to an agent who can log into a phone using that extension number. You can chose to record logical agent’s extension rather than the hunt group extension or hunt group member extension.

- **Hot Line Service**

  The stored number used on an outgoing or intraswitch Hot Line call is recorded by CDR the same as if it was manually dialed.

- **Intercept Treatment**

  If an outgoing or tandem call is routed to Intercept Treatment, the number dialed by the calling party is recorded as the dialed number, and Condition Code F is recorded.

- **Intercom — Automatic**

  Intercom calls can generate intraswitch CDR records.

- **Intercom — Dial**

  Intercom calls can generate intraswitch CDR records.

- **Inter-PBX Attendant Calls**

  If a user calls an Inter-PBX attendant and the trunk group used has CDR assigned, CDR records the following information:

  — Condition Code — A

  — Access Code Dialed — blank

  — Access Code Used — trunk access code of trunk used

  — Dialed Digits — Inter-PBX attendant access code

- **ISDN**

  When specific answer supervision is received from the network, an indication is sent to the CDR device to this effect. If an ISDN call has been interworked, the CDR record will not record the call as having answer supervision.

- **Loudspeaker Paging**

  When loudspeaker or chime paging is used, CDR may not correctly record the length of time a station was connected to an AUX trunk.

- **Manual Originating Line Service**

  If an attendant establishes an outgoing call for a voice terminal, designated as a Manual Originating Line, the CDR record for the call is the same as for any attendant-assisted outgoing call. The calling voice terminal extension number is recorded as the calling number, and Condition Code 1 applies.

- **Multiple LDNs**

  If incoming call information is recorded, the called number recorded for LDN calls is the extension number or trunk group access code to which the attendant completes the call. If the call terminates at the attendant console only, the called number recorded is 0, which is used to identify the attendants.

LDNs cannot be administered for intraswitch CDR. However, a call from an intraswitch optioned extension to a LDN will produce an intraswitch CDR.

- **Night Service — Night Station**

  The extension number assigned to the attendants is recorded as the dialed number. For G3vsV1/G3sV1 and G3iv1, the attendant extension number is always 0. For G3i-Global, G3rV1, G3V2, and later releases, the attendant extension number is administrable (the default is 0).

- **Night Service — Trunk Answer From Any Station**

  The extension number assigned to the attendants (0) is recorded as the dialed number.

- **Off-Premises Station**

  CDR data is recorded if the voice terminal is involved in an outgoing/incoming trunk call or it (or the other terminal involved in the call) has been optioned for intraswitch CDR.

- **PCOL**

  An outgoing PCOL call shows the dialed number in the "Dialed Number" field of the CDR record rather than a TAC. An outgoing PCOL call is recorded as a call from the originating extension number via the trunk group associated with the PCOL. On incoming PCOL calls the answering voice terminal's primary extension is recorded as the called number if incoming calls are recorded.

- **Planned Interchange on Duplicated G3r**

  When a planned interchange occurs (either demand or scheduled), it is possible for the CDR records on calls ending within 10-20 seconds after the interchange to report as "invalid long duration calls" (9:59:9 listed as call duration with Condition Code other than 4). This is caused by deviations in the clocks between the two processors and the short duration of the calls. These records should be considered invalid.

- **Private Network Access**

  Private Network Access calls will be recorded if CDR is administered for this trunk group. Private Network calls will be recorded if either an incoming and/or outgoing tie trunk is assigned CDR.

- **Remote Access**

  Remote Access calls will be recorded if Remote Access is provided on a per trunk group basis.

- **Ringback Queuing**

  Condition Code 8 is recorded for an outgoing call which is queued for a trunk before completion. The length of time the call is queued will not be recorded.

When an outgoing call is queued for a trunk and is unsuccessful (the queue times out or the calling party does not answer the callback) an CDR record is not generated for the call.

- **Security Violations Notification (SVN)**

  SVN calls will generate intraswitch CDR if the terminating extension is monitored. The originating extension for SVN calls cannot be administered for intraswitch monitoring.

- **Service Observing**

  No CDR records will be generated for Service Observing calls.

- **Tandem Tie Trunk Switching**

  The calling party on an incoming trunk can dial the CDR account code. The “Calling Number” field in CDR is the trunk access code for the incoming trunk group, the called number is the number dialed.

- **Temporary Bridged Appearance**

  A CDR record is not affected by any second or subsequent voice terminal bridging a call.

- **Temporary Signaling Connections (TSC)**

  TSCs and TSC requests associated with recorded calls will be recorded in the CDR record of the associated call, provided the switch is administered to use the ISDN version of CDR format layouts, which contain the necessary fields. Non-call-associated TSCs and TSC requests sent or received by the switch will generate their own CDR records if the switch is administered to record them. In either case, the “TSC Flag” field and the “Packet Count” field of the CDR record will be used to record TSC data.

- **Tie Trunk Access**

  Tie trunk calls will be recorded if CDR is administered for this trunk group. Tie trunk groups will be recorded if either incoming and/or outgoing tie trunks are assigned CDR.

- **Transfer**

  If a user **originates** a call on an outgoing trunk and then transfers the call to another voice terminal, the originating voice terminal will be recorded as the calling party.

  If a voice terminal user **receives** a call on an incoming trunk and then transfers the call to another extension, the extension that originally received the call is recorded as the dialed number.

  If a voice terminal user **receives an intraswitch call** and then transfers it to another extension, the extension that originally received the call is recorded as the dialed number.

  With the CDR call splitting option (available with G3i-Global V1 and V2 and later releases), if a user originates a call on an outgoing trunk and then transfers the call to another voice terminal, two CDR records are generated, one for each 'leg' of the call. For example, station 1 to outgoing number and station 2 to the same outgoing number. The call duration is appropriately assigned for each 'leg'.

Intra-switch CDR records are generated for each call to or from an intraswitch optioned extension. For example, Station A, which is intraswitch optioned, calls Station B. Station A then transfers the call to Station C. When either Station B or C drops, two CDR records with Condition Code 0 are output: one for the A to B call, and the second for the A to C call.

Intra-switch CDR transfer records are output when both the calling number (originator) and dialed number (terminator) drop from the call. The duration of the call is from the time the terminator answers until both the originator and terminator have dropped from the call.

- **Trunk-to-Trunk Transfer**

  Although they are not really conference calls, Trunk-to-Trunk Transfer connections are treated as such for CDR purposes. A separate CDR record is generated for each trunk in the connection.

  Unanswered Trunk Calls may or may not be recorded depending on administration. Each trunk group can be administered so that unanswered calls will be recorded if they remain unanswered for a specified period of time.

- **UDP**

  If one user calls another user via a Uniform Dial Plan extension number, and the trunk group used has CDR assigned, CDR records the following information:

  — Condition Code — 7

  — Access Code Dialed — blank

  — Access Code Used — trunk access code of trunk used

  — Dialed Digits — Uniform Dial Plan extension

- **VDN Return Destination**

  An incoming call does not generate a CDR record until the originator drops from the call. CDR creates a record when a call goes to the return destination VDN, the originator has not dropped, and vector processing — that is, the return destination VDN — has routed the call to an outgoing trunk. CDR does not create a record if vector processing routes a call from the return destination VDN to an internal call.

  If an incoming VDN call is routed to a station, CDR includes the station in the record.

  If an incoming VDN call is routed to an outgoing trunk, CDR includes the VDN in the record.

- **WATS and 800 Service**

  Calls made on a WATS or 800 Service trunk group will be recorded, if CDR is administered for the trunk group.

Administration

CDR is administered by the System Manager. The command is **change system-parameters CDR** for G3V2 and later releases. The following items can be administered.

System Parameters

- Type of CDR output format to be used. The type of output format must be assigned for both the primary and secondary output formats, if both the primary and secondary ports are used.
- Extension number assigned to the output format. The extension number must be assigned for both the primary and secondary output device. Before the extension number is assigned, the System Manager should check to make sure that it is not already assigned as a Property Management System (PMS) extension or a Permanent Switched Call (PSC) extension.
- Whether standard or ISDN formats are used.
- Whether enhanced formats are used.
- CDR account code length (from one to 15), the system defaults to two digits.
- The speed at which the CDR device connected to the direct RS-232C interface on the processor circuit pack will operate (300, 1200, 2400, 4800, or 9600 baud rate). This applies to G3s/vs/i only.
- Whether the reason for disconnect is recorded instead of the FRL.
- Whether an account code is required on a toll call.
- Whether the hunt group extension or the hunt group member extension is recorded by CDR.
- Whether or not the called VDN is recorded instead of the hunt group extension or hunt group member extension.
- Whether or not the called logical agent is recorded instead of the hunt group extension or hunt group member extension.
- CDR can be suppressed for Ineffective Call Attempts or for All Calls Excluding Outgoing Calls; system defaults to no. Ineffective call attempts are calls originated by a voice terminal user that are blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy. Ineffective call attempts include calls to incoming or

outgoing trunks that are unavailable due to trunk usage allocation for ISDN Call-By-Call Service Selection trunks and incoming calls rejected by the switch due to NSF mismatch.

- The number of trailing digits in the “CDR Dialed Number” field to be blanked on output for an outgoing call originating from a station with CDR Privacy enabled.
- Whether intraswitch CDR records will be generated for internal calls.
- Whether call splitting is activated.
- Whether attendant call recording is activated.
- Whether to record non-call associated and/or call associated TSC data. Defaults for both is ‘‘no.’’
- For G3r, what to do when record handling resources are exhausted (when 1900 archived records are full and a new one arrives). The options are: overwrite oldest record (warning), give reorder tone (reorder), redirect to attendant (attendant). A warning is the default value on the administration screen.
- Whether to record dialed or outpulsed (translated digits) for outgoing calls on the ‘‘Dialed Number’’ field.

Date and Time

The date and time should always be updated for events such as a leap year, daylight savings time, or a system restart after a power failure. If a time of day is not administered, CDR records will not be generated.

If the time is changed while a call is in progress, the actual duration for that call is not reflected in the CDR record. Instead, a special sequence of 9999 is recorded in the CDR record to indicate that the call was in progress during a time change.

Trunks, Loudspeaker Paging, and Code Calling Access

CDR can be assigned to all trunk groups, Loudspeaker Paging Access trunks, and PCOL trunks. The system defaults to yes for CDR. The System Manager must determine which types of trunks will be assigned CDR.

COR

Specify if CDR account code entry is forced.

Feature Access Codes

Assign CDR account code access code.

IXC Codes

- IXC access numbers
- Name of IXC (optional)

Data Modules and Modems

One or both of the CDR output devices can be connected to a PDM, Trunk Data Module, or a Modem. The following items must be administered:

- For G3vs/s/i, a netcon channel must be assigned using a ‘Data Module’ form and entering data-channel or netcon channel for the type. This channel provides a path for CDR data from the Switch Processing Element to the time-division bus.

  If the EIA port on the Processor circuit pack is used by the output device, the CDR output device extension should be administered as ‘‘eia’’ for G3s/vs or G3i. The SMDR/CDR EIA port is on the TN773 Processor.

- For G3r, a system port must be assigned using the "Data Module" form and entering system port for the type. The port is on a TN726B Data line and Associated PDATA Port is on a TN553 Packet Data Line, these two parts must be hardwired together at the “Cross-Connect” field. This provides a path for CDR data from the SPE to the packet-bus through the hardwired connection to the time-division bus.

  Ten system ports can be administered. The TN553 has 12 ports of which 10 can be used for system port applications. The TN726B has eight ports and would require two circuit packs to get the full 10 system ports. System ports are used by other applications as well (remote administration, system printer, PMS-link, etc.)

- For both G3r and G3i/s/vs, a TN726 Data Line circuit pack can also be used in conjunction with an ADU to connect to an output device. If the CDR output device is connected to a TN726 Data Line circuit pack via an ADU, administer a Data Line Type Data Module.

If the CDR output device is connected to a PDM, administer a PDM Type Data Module.

If the CDR output device is connected to a Trunk Data Module, administer a YDM Type Data Module.

If the CDR output device is connected to a 212A-type modem, a 2500 Voice Terminal a Pooled Modem must be administered. This allows circuit switched data connections between digital data communications equipment (data modules) and analog data communications equipment (modems).

Security

Call detail records should be monitored daily for unusual calling patterns, long calls, international calls, calls outside of normal business hours, and other indications of toll fraud. Call accounting systems such as the AT&T Hacker Tracker provide automatic monitoring for fraudulent calling patterns.

Call Forwarding All Calls

Feature Availability

This feature is available with all Generic 3 releases. Call Forwarding Override and the List Call Forwarding command are only available with G3V4 and later releases.

Description

Allows all calls to an extension number to be forwarded to a selected internal extension number, external (off-premises) number, the attendant group, or a specific attendant. This feature is activated or deactivated by dial access code or by a Call Forwarding button.

Call Forwarding All Calls can be activated or deactivated by voice terminal users and data terminal users. Also, an attendant or voice terminal user with console permission can activate or deactivate the feature for a particular extension number, TEG, DDC, UCD group, or ACD split (but not vector-controlled splits; see Call Vectoring for more information).

Voice terminal users activate Call Forwarding All Calls by dialing a feature access code or pressing a Call Forwarding button and then dialing the designated (forwarded-to) number. The feature is deactivated by dialing a different feature access code or pressing the Call Forwarding button again.

An attendant activates Call Forwarding All Calls by dialing a feature access code, followed by the forwarding extension number plus the forwarded-to number. The attendant deactivates the feature by dialing a different access code, followed by the extension number for which the feature is to be canceled. The attendant cannot have a Call Forwarding button assigned to the console.

A voice terminal user with console permission activates Call Forwarding All Calls for another user by dialing a feature access code, followed by the forwarding extension number plus the forwarded-to number. A voice terminal user with console permission can also activate Call Forwarding All Calls for himself or herself by dialing the feature access code or pressing the Call Forwarding button.

When a Call Forwarding button is used to activate the feature, the status lamp associated with the button remains lighted until the feature is deactivated.

Calls can be forwarded only once. Calls forwarded to a designated (forwarded-to) number do not forward again. These calls ring the designated number, if possible; redirect if the forwarding party’s Call Coverage criteria are met; or return busy tone to the calling party.

Only one call can be forwarded off-premises from a given voice terminal every 30 seconds. This prevents the possibility of the first call getting forwarded back from a remote forwarding device and then being forwarded again, thus ultimately using all available trunks.

When Call Forwarding All Calls is activated at a voice terminal and a call for that terminal is forwarded, the terminal can (if administered to do so) receive a redirection notification signal that a call is being forwarded.

If ‘‘save translation’’ is run after call forwarding is activated for a terminal, forwarding is saved to tape.

Call Forwarding Override

With G3V4 and later releases, Call Forwarding All Calls can be administered to allow Call Forwarding Override. With the override feature, the forwarded-to terminal user can override call forwarding by placing a call to the forwarding terminal. In this way, the forwarded-to user can call the forwarding user for private consultation, to conference the calling party, or to transfer the call back to the forwarding terminal. Call Forwarding Override is automatically invoked when the system-wide override option is set, and the forwarded-to station attempts to call the forwarding station.

Call Forwarding Override cannot be used when calls forward to an external number. Call Forwarding All Calls from a data user or a hunt group cannot be overridden with Call Forwarding Override.

Security Measures

G3V4 and later releases provide a **list call-forwarding** command. The command lists stations that have Call Forwarding All Calls and Call Forwarding Busy/Don’t Answer active. It displays the number and name of the extensions that have the feature active as well as the forwarded to destination. Use the **list call-forwarding** command to identify unauthorized activation of the Call Forwarding features.

Users who do *not* have permission to call out of the building may do so with Call Forwarding. The users could forward their work phone to their home and forward their home phone to work. Proper security precautions should be taken in this case.

Considerations

With Call Forwarding All Calls, users can have their incoming calls forwarded to another extension number. This allows users to have their calls follow them when they know they are temporarily near another extension. A user can also forward calls to an outside number when temporarily at an off-premises location. There is no maximum number of calls that can be forwarded simultaneously. For TEG, UCD groups, and DDC, Call Forwarding All Calls can only be activated by the attendant or voice terminal user with console permission.

If an incoming call on a CO trunk is forwarded to an external number, any other calls to the same extension within the next 30 seconds receives busy tone or redirect to coverage if Send All Calls is assigned.

When a call is forwarded to an off-premises location, the forwarding-to number can have a maximum of 16 digits. When counting the 16 digit limit, count the TAC or ARS feature access code (usually a “9”). Do not count the “#” sign that might append to the end of the terminate the number.

Calls to attendants cannot be forwarded. However, calls can be forwarded to the attendant group.

Only the attendant, or any station with console permissions, can activate Call Forwarding All Calls for a data module.

If a user attempts to make a call to an extension that he or she is restricted from calling, and the called extension has activated Call Forwarding All Calls, the call is not completed.

A user cannot forward calls to an extension that he or she is normally restricted from calling.

Interactions

The following features interact with the Call Forwarding All Calls feature.

- **Attendant Override of Diversion Features**

  If an attendant uses redirection override to call a user who has Call Forwarding active, the call is not forwarded and remains at the user’s voice terminal.

- **Automatic Callback and Ringback Queuing**

  Automatic Callback cannot be activated toward a voice terminal that has Call Forwarding activated. If Automatic Callback was activated before the called voice terminal user activated Call Forwarding, the callback call attempt is redirected to the forwarded-to party.

- **Call Coverage**

  If the principal’s (forwarding extension number) redirection criteria are met at the designated (forwarded-to) extension number, the forwarded call is redirected to the principal’s coverage path and the designated extension gets a temporary bridged appearance, which remains active after the call is answered so the designated extension can bridge onto the call if desired. The temporary bridge appearance remains until the caller hangs up.

If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding is active at a voice terminal, incoming Priority Calls forward to the designated extension number and all other calls redirect according to the Call Coverage path.

When a covering user has activated Call Forwarding, a coverage redirected call does not forward to the designated extension number. Instead, the call is redirected to the next point in the principal's coverage path, if available. If no other coverage point is available, the call remains at the principal's voice terminal.

- **Call Detail Recording (CDR)**

  When a call is forwarded to an off-premises number, the call is recorded in CDR records as a call from the forwarding station.

- **CDR Account Code Dialing**

  If forced entry of account codes is required, calls cannot be forwarded to off-premises destinations.

- **Call Forward Busy/Don't Answer**

  Call Forwarding All Calls and Call Forward Busy/Don't Answer cannot be active for the same terminal at the same time.

- **Call Park**

  Calls can be parked on a forwarded-from extension even though Call Forwarding is active for that extension. If a forwarded-to extension user parks a call that had been forwarded to that extension, the call is normally parked on the forwarded-to extension, not the forwarded-from extension.

- **Direct Inward Dialing (DID)**

  If an incoming DID call is forwarded to an external number, any other calls to the same DID extension within the next 30 seconds receives busy tone or redirect to coverage, if assigned.

- **Expert Agent Selection (EAS)**

  Agents logged in with EAS enabled *cannot* activate Call Forwarding. The physical extension where the EAS agent is logged in can be forwarded, but the EAS agent must first log out. Then, the phone can be forwarded.

- **Hot Line Service and Manual Originating Line Service**

  Voice terminals assigned these features cannot activate Call Forwarding. However, calls can be forwarded to these terminals. Other users can active Call Forwarding for the hotline terminal.

- **Interflow**

  The Interflow feature allows ACD calls to be redirected from one split to a split on another switch or to another external location. This is accomplished by forwarding calls that are directed to the split extension to an off-premises location via the Call Forwarding All Calls feature. For details on the Interflow feature, see the Intraflow and Interflow feature description elsewhere in this document.

- **Intraflow**

  Call forwarding can be used to unconditionally redirect ACD calls from a split to another destination on the same switch.

- **Leave Word Calling (LWC)**

  LWC cannot be activated toward a voice terminal that has Call Forwarding activated. If (LWC) was activated before the called voice terminal user activated Call Forwarding, the callback call attempt is redirected to the forwarded-to party.

- **Personal Central Office Line (PCOL)**

  PCOL calls cannot be forwarded.

- **Send All Calls**

  If an extension has both Send All Calls and Call Forwarding All Calls activated, calls to that extension that can immediately be redirected to coverage are redirected. However, other calls, such as priority calls, are forwarded to the designated extension.

  Activation of Send All Calls at the forwarded-to extension does not affect calls forwarded to that extension.

Administration

Call Forwarding All Calls is assigned on a per-extension number basis by the COS. The following items require administration by the System Manager:

- Voice Terminals
  - Class of Service
  - Call Forwarding Buttons
  - Redirection Notification
- Feature Access Codes for activation and deactivation of Call Forwarding All Calls
- Enable Call Forward Override if desired (G3V4 and later releases only)

Hardware and Software Requirements

No additional hardware or software is required.

Call Forward Busy/Don’t Answer

Feature Availability

Call Forward Busy/Don't Answer is available with G3V4 and later releases.

Description

Allows calls to an extension to be forwarded when the called extension is busy or when the call is not answered after an administered number of rings. If the called extension is busy, the call forwards immediately. If the called extension is not busy, the incoming call rings the called extension, then forwards only if it remains unanswered longer than the administered "Coverage Subsequent Redirection/CFWD No Answer Interval" (1-99 ring cycles).

Calls can be forwarded to an internal extension number, external (off-premises) number, the attendant group, or a specific attendant. The feature is activated or deactivated with a Feature Access Code (FAC) or Call Forward Busy/Don't Answer feature button.

An attendant or voice terminal with console permission also can activate or deactivate the feature for a particular extension by using a Feature Access Code (FAC).

Calls can be forwarded only once. Calls forwarded to a designated (forwarded-to) number do not forward again. These calls ring the designated number, if possible; redirect if the forwarding party's Call Coverage criteria are met; or return busy tone to the calling party.

Only one call can be forwarded off-premises from a given voice terminal every 30 seconds. This prevents the possibility of the first call getting forwarded back from a remote forwarding device and then being forwarded again, thus ultimately using all available trunks.

When Call Forward Busy/Don't Answer is active at a voice terminal and a call for that terminal is forwarded, the terminal can (if administered to do so) receive a redirection notification signal that a call is being forwarded.

Call Forward Busy/Don't Answer cannot be activated for hunt groups, data extensions, or Terminating Extension Groups (TEG). Calls to an attendant or EAS agent cannot be forwarded.

Call Forwarding Override

With the override feature, the forwarded-to terminal user can override call forwarding by placing a call to the forwarding terminal. In this way, the forwarded-to user can call the forwarding user for private consultation, to conference the calling party, or to transfer the call back to the forwarding terminal. Call Forwarding Override is automatically invoked when the forwarded-to station attempts to call the forwarding station.

Call Forwarding Override cannot be used when calls forward to an external number.

End User Operation

Users activate the feature by dialing the Call Forward Busy/Don't Answer FAC or by pressing the Call Forward Busy/Don't Answer feature button. Once the feature is active, the user receives dial tone and dials the phone number that calls should be routed to when they are forwarded. When forwarding is successful, the user receives confirmation tone. In addition, when a feature button is used to activate Call Forwarding, the status light associated with the button remains lighted until the feature is deactivated. If access to the feature is denied or activation fails, the user receives intercept tone.

The Call Forward Busy/Don't Answer feature is deactivated by dialing the deactivation FAC or pressing the feature button. Confirmation tone is given to indicate that the feature has been successfully deactivated.

The attendant or user of a terminal with console permissions activates the feature by dialing the FAC followed by the forwarding extension number and then the forwarded-to extension number. The feature is deactivated by dialing the deactivation FAC followed by the extension number for which the feature is to be deactivated.

To activate Call Forward Busy/Don't Answer for their own extension, the user of a station with console permission must dial the FAC, then dial their own number and finally dial the forwarded-to extension number.

Security Measures

G3V4 and later releases provide a **list call-forwarding** command. The command lists stations that have Call Forwarding All Calls and Call Forwarding Busy/Don’t Answer active. It displays the number and name of the extensions that have the feature active as well as the forwarded-to destination. Use the **list call-forwarding** command to identify unauthorized activation of the Call Forwarding features.

Considerations

If an incoming trunk call is forwarded to an external destination and answered by the forwarded-to destination, any other incoming trunk calls to the forwarding extension within the next 30 seconds will not forward. The call will ring or get busy tone depending on the status of the terminal.

When a call is directed to an external number, the forwarded-to number can have a maximum of 16 digits.

If a user attempts to make a call to an extension that he or she is restricted from calling, the call will not complete even if Call Forwarding Busy/Don’t Answer is activated on the called station. A user cannot forward calls to a destination that he or she is normally restricted from calling.

Interactions

The following features interact with Call Forward Busy/Don’t Answer:

- **Attendant Override of Diversion Features**

  If an attendant uses redirection override to call a user who has Call Forwarding Busy/Don’t Answer active, the call is not forwarded and remains at the user’s voice terminal.

- **Automatic Callback and Ringback Queuing**

  Automatic Callback cannot be activated toward a voice terminal that has Call Forwarding Busy/Don’t Answer activated. If Automatic Callback was activated before the called voice terminal user activated Call Forwarding, the callback call attempt is redirected to the forwarded-to party.

- **Bridging**

  Calls will not terminate to bridged appearances when Call Forward Busy/Don’t Answer is active.

- **Call Coverage**

  If the principal’s (forwarding extension number) redirection criteria are met at the designated (forwarded-to) extension number, the forwarded call is redirected to the principal’s coverage path and the designated extension gets a temporary bridged appearance, which remains active after the call is answered so that the designated extension can bridge onto the call if desired. However, if the principal (forwarding extension number) redirection criteria is busy only and the designated (forwarded-to) extension number is busy, then calls that terminate at the principal will not be forwarded because the forwarded-to station is not available and the principal is idle (does not meet coverage criteria).

  If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding Busy/Don’t Answer is active at a voice terminal, incoming Priority Calls forward to the designated extension number and all other calls redirect according to the Call Coverage path.

When a covering user has activated Call Forwarding, a coverage redirected call does not forward to the designated extension number. Instead, the call is redirected to the next point in the principal's coverage path, if available. If no other coverage point is available, the call remains at the principal's voice terminal.

- **Call Detail Recording (CDR)**

  When a call is forwarded to an off-premises number, the call is recorded in CDR records as a call from the forwarding station.

- **Call Detail Recording (CDR) Account Code Dialing**

  If forced entry of account codes is required, calls cannot be forwarded to off-premises destinations.

- **Call Forwarding All Calls**

  Call Forwarding All Calls and Call Forward Busy/Don't Answer cannot be active for the same terminal at the same time.

- **Call Park**

  Calls can be parked on a forwarded-from extension by an attendant even though Call Forwarding Busy/Don't Answer is active for that extension. If a forwarded-to extension user parks a call that had been forwarded to that extension, the call is parked on the forwarded-to extension, not the forwarding extension.

- **Direct Inward Dialing (DID)**

  If an incoming DID call is forwarded to an external number, any other trunk calls to the same DID extension within the next 30 seconds receives busy tone or redirect to coverage, if assigned.

- **Expert Agent Selection (EAS)**

  Agents logged in with EAS enabled cannot activate Call Forwarding unless the station has a Class Of Service (COS) with console permissions (console permission set to "y"). Dialing the Feature Access Code (FAC) for Call Forwarding then requires that the station being forwarded is entered first.

- **Hot Line Service and Manual Originating Line Service**

  Voice terminals assigned these features cannot activate Call Forwarding. However, calls can be forwarded to these terminals. Another user can activate Call Forwarding for a hotline terminal.

- **Personal Central Office Line (PCOL)**

  PCOL calls cannot be forwarded.

- **Send All Calls**

  If an extension has both Send All Calls and Call Forward Busy/Don't Answer activated, calls to that extension that can immediately be redirected to coverage are redirected. However, other calls, such as priority calls, are forwarded to the designated extension.

Activation of Send All Calls at the forwarded-to extension does not affect calls forwarded to that extension.

Administration

Call Forward Busy Don't Answer is assigned on a per-extension number basis by the COS. The following items require administration by the System Manager:

- Voice Terminals
  - Class of Service
  - Call Forward Busy/Don't Answer Buttons
  - Redirection Notification
- Feature Access Codes for activation and deactivation of Call Forward Busy/Don't Answer
- Coverage Subsequent Redirection/CFWD No Answer Interval — the number of rings before a call forwards

Hardware and Software Requirements

No additional hardware or software is required.

Call Management System (CMS)

Feature Availability

CMS is an adjunct and is optionally available with all Generic 3 releases as an adjunct.

Description

Provides real-time and historical reports for monitoring ACD facilities and personnel. Unlike BCMS, the CMS software resides in a computer (usually referred to as an adjunct) that connects to the switch via a data link. For more information, refer to the appropriate CMS documentation.

Call Park

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows users to put a call on hold and then retrieve the call from any other voice terminal within the system.

When a voice terminal user, active on a call, needs to go to another location for information, the call can be placed in Call Park and retrieved at the other location.

Conference calls can also be placed in Call Park.

Call Park can be activated by any of the following:

- A single-line voice terminal user — Flash the switchhook, dial the Call Park access code, and hang up. The call is parked on the user's extension number.
- A multiappearance voice terminal user — Press the Transfer or Conference button, dial the Call Park access code, and press the Transfer or Conference button again, or simply press the Call Park button (if assigned). The call is parked on the user's extension number.
- An attendant — Press Start, dial the Call Park access code followed by any extension number, and press Release. The call is parked on the number dialed. An attendant can use the Direct Extension Selection With Busy Lamp Field feature instead of dialing the extension number.
- The system — When Code Calling Access is used, the call is automatically parked on the paged party's extension number.

Calls are retrieved by dialing the Call Park Answer Back access code and the extension number where the call is parked or, by pressing the same Call Park button used to park the call.

A system-wide expiration interval can be set for parked calls. When the interval expires, the parked call redirects to an attendant console (or, if administered, to the parking user) and no longer is parked on the extension number. However, if the parked call has already been retrieved when this interval expires, the call does not redirect. If two parties are connected on a parked call, a third party can also answer the call before the interval expires, creating a three-way conference. This option is tied to the default paging option. Timed out parked calls redirect to the attendant if the default ''Loudspeaker Paging'' option is assigned and to the parking user if ''Deluxe Paging and Call Park Timeout to Originator'' option is assigned.

If no attendant (this includes CAS, local attendants, and Individual Attendant Access) or night service extension is administered, and if Night Service — Trunk Answer From any Station is not administered, the expiration interval is ignored and the call remains parked.

The attendant console group can have common shared extensions numbers used exclusively for Call Park. These extension numbers are not assigned to a voice terminal, but are stored in system translations and used to park a call. These extension numbers are particularly useful when one party is paged at the request of another party. The calling party is parked and the extension number is announced. The lamp associated with the extension number identifies call parked or no call parked (instead of active or idle status).

Considerations

Call Park can be used whenever a voice terminal user who is on a call needs to go elsewhere and obtain information, and wishes to complete the call at another extension. Call Park also allows users to answer a call from any station after being paged by a voice terminal user or an attendant.

Only one call per extension number can be parked at a time, even if the extension number has multiple appearances. However, a conference call with five parties can be parked. The sixth conferee is the retrieving party.

Calls cannot be parked on a group extension number. If a group member places a call in Call Park, the call is parked on the member extension number. Group members include the following:

- A Coverage Answer Group member
- A DDC group member
- A TEG member
- A UCD group member

Interactions

- Abbreviated Dialing

  An Abbreviated Dialing button can be assigned so that parking calls or retrieving parked calls can be done by pressing a button, instead of using the buttons and access codes normally used. This operation reduces the number of steps required to park a call or retrieve parked calls.

- Automatic Wakeup

  Automatic Wakeup calls cannot be parked.

- Bridged Call Appearance

If a user, active on a bridged call appearance, activates Call Park, the call is parked on the primary extension associated with the bridged call appearance.

- Call Vectoring

  A call cannot be parked on a VDN extension. Also, a call that is undergoing vector processing cannot be parked.

- Data Privacy and Data Restriction

  These features are automatically deactivated when a call is parked.

- Loudspeaker Paging Access

  Calls to paging zones cannot be parked.

- Loudspeaker Paging Access — Deluxe

  If the system is administered to have Deluxe Paging, parked calls are redirected to the parking user when the Call Park timeout interval expires.

- Music-on-Hold

  If a parked call involves only one party, the parked user hears music-on-hold.

  — With a Call Park feature access code

  — By pressing the Transfer button, the Call Park button, and Transfer button again

  If a call involves only one party but was parked by pressing the Conference button, the Call Park button, and then the Conference button again, the parked user hears music-on-hold.

  If a call involves multiple parties (such as a conference call), none of the parties hears music-on-hold.

  If Music-on-Hold is provided, the user activating Call Park also hears music after the call is parked and confirmation tone is heard.

- Remote Access

  A Remote Access caller cannot park a call. However, the Code Calling Access feature, an answering attendant, or a voice terminal user can park an incoming Remote Access call.

Administration

Call Park is administered on a per-system basis by the System Manager. The following items require administration:

- Call Park access code
- Answer Back access code
- Call Park Time-out interval (from one to 90 minutes)

- Call Park button (multiappearance voice terminals only). A Call Park button should have a lamp so that the voice terminal user can tell when a call is parked on his or her extension.
- Common shared extension numbers for the attendant group.
- The “Deluxe Paging and Call Park Timeout to Originator” field on the Feature Related System Parameters form must be administered as ‘‘yes’’ to have parked calls return to the parking user (the originator) when the Call Park Timeout interval expires. If this field is administered as ‘‘no’’ parked calls go to the attendant when the Call Park Time-out interval expires.

Hardware and Software Requirements

No additional hardware or software is required.

Call Pickup

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows voice terminal users to answer calls to other extension numbers within the user's specified Call Pickup group.

Call Pickup groups are established so that when one member of a group is away, other members of the group can answer that member's calls. A Call Pickup group usually consists of users who are located in the same area or have similar functions.

When a member of a Call Pickup group is away and receives an incoming call, any member of the Call Pickup group can answer the call. A member simply goes off-hook and dials the Call Pickup access code or presses a Call Pickup button. That group member is then connected to the calling party.

A Temporary Bridged Appearance is maintained at the called voice terminal. This allows the called party to bridge onto the call after it has been picked up by another member of the Call Pickup group.

If a Call Pickup group member's voice terminal is equipped with a Call Pickup button and status lamp, then:

— The status lamp has a steady light when Call Pickup is used to answer a call.

— The status lamp has a flash feature light when the ''Call Pickup Alerting'' option on the ''System Parameters Feature'' form is enabled and there is an incoming call to any extension in the Call Pickup group, including the called station. Group members, other than the principal called station, can answer such calls using Call Pickup; the principal can answer such calls on the ringing call or bridged appearance.

NOTE:
Call Pickup Alerting for a voice terminal only takes effect when the Call Pickup status lamp is not lit. If Call Pickup is used to answer a call, the status lamp displays a steady light and does not flash if there are additional calls to this Call Pickup group.

For the called station only, both the Call Pickup and the Call Appearance status buttons flash.

If calls are ringing at two or more voice terminals in a Call Pickup group, and another group member presses his or her Call Pickup button, then a distribution algorithm determines which call is answered. In this way, equal treatment is given to all Call Pickup group members. Specifically, when a voice terminal's Call Pickup button is pressed, the algorithm ‘‘cycles’’ through the group extension numbers until it finds an extension with a call eligible for Call Pickup. The next time a Call Pickup button is pressed, the algorithm resumes cycling from the *next* extension number.

For example, if extension A has two calls ringing and extension B has one call ringing, and if one of extension A's calls is answered using Call Pickup, then extension B's call is answered the next time Call Pickup is activated. After extension B's call is answered, Call Pickup can be used to answer the second call to extension A.

When multiple calls are ringing on a single voice terminal, and another member of the Call Pickup group activates Call Pickup, then:

- All switches (except G3i-Global), the call associated with the lowest call appearance number is answered. For example, if calls are ringing on the second and fourth call appearance button on a voice terminal, and a user at another voice terminal activates Call Pickup, then the call on the second call appearance button is answered.
- For G3i-Global, the call type and call appearance number determine which call is answered. The precedence order for call types is:
  1. Priority Calls
  2. Attendant or Trunk Calls
  3. Intercom Calls
  4. All other calls

  If multiple calls of the same call type are eligible for Call Pickup, the call associated with the lowest call appearance number is answered. For example, if priority calls are ringing on the second and fourth call appearance button on a voice terminal, and a user at another voice terminal activates Call Pickup, then the call on the second call appearance button is answered.

Considerations

With Call Pickup, a user does not have to leave his or her voice terminal to answer a call at a nearby voice terminal. Instead, the user simply lifts the handset and dials an access code or presses a Call Pickup button. This allows unanswered calls to be handled quickly and efficiently.

A voice terminal can be a member of only one Call Pickup group.

Setting the Call Pickup Alerting option on the “System Parameters Features’’ form to yes provides voice terminals equipped with a Call Pickup button and status lamp with a visual notification that an incoming call is eligible for Call Pickup. Specifically, the voice terminal Call Pickup status lamp flashes when a call to another extension is available for Call Pickup.

NOTE:
Call Pickup Alerting for a voice terminal only takes effect when the Call Pickup status lamp is not lit. If Call Pickup is used to answer a call, the status lamp displays a steady light and does not flash if there are additional calls to this Call Pickup group.

When a member of a Call Pickup group is away from his or her voice terminal and receives an incoming call, the other members of the Call Pickup group voice terminals do not ring. As a result, the Call Pickup feature is only useful if either:

- The Call Pickup Alerting option is enabled and the Call Pickup group members have voice terminals equipped with a Call Pickup button and status lamp.
- The members of a Call Pickup group are located close to one another and can hear each other’s voice terminal ring.

Interactions

The following features interact with the Call Pickup feature.

- **Automatic Callback and Ringback Queuing**

  Callback calls cannot be answered by Call Pickup group members.

- **Bridged Call Appearance**

  The interaction between Call Pickup and Bridged Call Appearance depends on whether the Call Pickup Alerting option on the “System Parameters Features’’ form is enabled:

  - If the Call Pickup Alerting option is set to ‘‘yes,’’ activating Call Pickup, while a bridged call appearance is ringing on a Call Pickup group member’s station, does not pickup the call.

    For example, if Adrian’s voice terminal receives ringing on a call to a bridged extension number that is not in Adrian’s Call Pickup group, then the incoming call cannot be picked up by members of Adrian’s Call Pickup group.

  - If the Call Pickup Alerting option is set to ‘‘no,’’ and if a voice terminal receives ringing on a bridged call appearance, then the incoming call can be picked up by members of that voice terminal’s Call Pickup group.

- **Call Waiting Termination**

  A Call Waiting call cannot be picked up by a Call Pickup group member.

- **Conference**

  If Call Pickup Alerting is enabled and a call is picked up and conferenced into a conference call, the Call Pickup status lamp flashes if additional calls are available for Call Pickup.

- **Hold**

  A call, picked up and placed on hold at an extension, remains on that extension, even if the called party answers the call.

  If Call Pickup Alerting is enabled and a call is picked up and placed on hold, the Call Pickup status lamp flashes if additional calls are available for Call Pickup.

- **Hot Line Service and Manual Originating Line Service**

  Voice terminals assigned these features can be Call Pickup group members so their incoming calls can be answered. However, voice terminal users with these features assigned cannot answer calls for other group members.

- **Intercom—Automatic**

  Call Pickup can be used to answer an Automatic Intercom call.

- **Internal Automatic Answer (IAA)**

  Internal calls to a voice terminal in a Call Pickup group are eligible for IAA. If the called extension in a Call Pickup group is IAA-active, the call is answered automatically. An extension that is IAA-active is not able to automatically answer calls to other voice terminals in its Call Pickup group. IAA-eligible calls to an IAA-active extension can't be answered by Call Pickup (because they are automatically answered at the called station), but any non-IAA-eligible calls such as external calls that ring the IAA-active station can be answered by members of that station's call pickup group.

- **Transfer**

  If Call Pickup Alerting is enabled and a call is picked up and transferred, the Call Pickup status lamp flashes if additional calls are available for

  Call Pickup.

Administration

Call Pickup is administered by the System Manager. The following items require administration:

- Call Pickup group number
- Members (extension numbers) of each Call Pickup group
- Call Pickup access code
- Call Pickup buttons

Also, to enable Call Pickup Alerting for voice terminals equipped with a Call Pickup button and status lamp, the ‘‘Call Pickup Alerting’’ option on the ‘‘System Parameters’’ form must be set to yes

Hardware and Software Requirements

No additional hardware or software is required.

Call Prompting

Feature Availability

This optional feature is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Uses specialized vector commands to provide flexible handling of incoming calls based on information collected from the calling party.

Call Prompting can be used in various applications to achieve better and more flexible handling of incoming calls. This description describes four call prompting applications. A brief description of each of the four sample applications is given below. (A more detailed description of each application is given later in this chapter.)

- Automated Attendant — Allows the calling party to enter the extension of the party that he/she would like to reach. The call is then routed to that desired extension.
- Data In/Voice Answer (DIVA) Capability — Allows the calling party to hear an announcement based on the digits that he or she enters, or to be directed to a hunt group or another system extension.
- Data Collection — Allows the calling party to enter data which can then be used by a host/adjunct to assist in call handling. This data, for example, may be the calling party's account number.
- Message Collection — Gives the calling party the option of leaving a message or waiting in queue for an agent.

Since the Call Vectoring feature is used with the Call Prompting feature, it is recommended that the Call Vectoring feature be read and understood in order to more easily comprehend this Call Prompting feature description. Call Vectoring Basic is not a prerequisite for vector prompting. This means that stand-alone call prompting implementations cannot queue calls, or do any conditional checks based on queue or agent status, time of day, or day of week.

As in the Call Vectoring feature, VDNs are used to access Call Prompting vectors. Each VDN routes to a single Call Prompting vector but several VDNs may route to the same Call Prompting vector.

Call Prompting Vector Commands

The following list shows the complete set of vector commands that are used with the Call Prompting feature.

- Announcement

- Collect Digits
- Goto step
- Goto vector
- Messaging Split
- Route-to Digits
- Route-to Number
- Stop
- Wait-time

See the *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 555-230-520, for a complete description of each of these commands. The following paragraphs describe how the following vector commands are used with the Call Prompting feature:

- Collect Digits
- Goto Step
- Goto Vector
- Route-to Digits
- Route-to Number... if digits
- Stop

Collect <*# of digits*> digits after announcement <*extension*>

In this command, ***<# of digits>*** is an administered number of digits from one through 16, and ***<extension>*** is an administered announcement extension or ***none*** (digits are collected without a prompt).

With this command, the switch can collect up to 24 touch-tone digits from the calling party. Up to 16 of these digits can be used immediately, while any remaining digits are stored as dial-ahead digits (explained later). An optional announcement (prompt) may be used to request the calling party to enter these digits. In addition, the announcement can instruct the user to enter an asterisk (***) if incorrect data is entered. When the calling party enters an asterisk (***), the digits collected for the current **collect** command are deleted, digit collection is restarted, and the announcement is not replayed.

When programming this command, the maximum number of digits requested of the calling party must be specified in the administration of the command. If the calling party can enter less digits than the maximum specified, the announcement should instruct the calling party to terminate the entry with a pound (*#*) sign digit as an end-of-dialing indicator. If less digits than the maximum specified are entered and the calling party does not complete the entry with a pound (*#*) sign, an inter-digit timeout occurs. The timeout terminates the command and any digits collected prior to the timeout are available for subsequent vector processing.

When digit collection for the current ***collect*** command completes, vector processing continues at the next vector command. However, the switch continues to collect any subsequently dialed digits. These ‘‘dialed-ahead’’ digits are to be saved for use by subsequent **collect** commands and provide the calling party with a means to bypass subsequent unwanted announcement prompts. For example, a frequent caller to a service may not want to listen to all of the announcements in full. When the dial-ahead digits are provided by ASAI, the **collect** command collects these first.

The sum of the digits collected for the current **collect** command plus the dial-ahead digits must not exceed the switch storage limit. Any additional digits dialed are discarded until storage is freed up by a subsequent **collect** command. These additional digits are only available for use by subsequent **collect** commands and are never used by other vector commands that operate on digits (such as **route-to digits**, **goto...if digits,** and so on). In addition, these digits are not displayed as part of the ‘Caller Information’ button operation.

This command functions as follows:

1. A DTMF receiver (TTR) is required for the collection of digits (unless the digits are available from ASAI). A TTR may already be connected to the call (due to the processing of a previous **collect** command). If answer supervision has not been previously returned to the calling party as a result of a previous vector step, then answer supervision is returned at this point. Also, if a TTR is required for the collection of digits, the call (with inter-digit timing disabled) is connected to a TTR. Incoming rotary trunks and internal hybrid and rotary station sets, for example, do not require connection of a TTR. Note that external users with rotary sets cannot use the Call Prompting feature.
   - If a TTR is not available and the TTR queue is not full, the call is queued for the next available TTR. Processing of this command does not continue until a TTR becomes available and is connected. Answer supervision is not returned at this time. The caller typically hears whatever feedback has previously been provided by the vector. Therefore, it is a good idea to begin each prompting vector with a wait hearing ringback step.
   - If a TTR is not available and the TTR queue is full, then vector processing continues at the next vector command (the **collect** command is unsuccessful).
2. If dial-ahead digits are available, the system reacts as follows:
   - The connection of the announcement prompt (if administered) is skipped.
   - If no TTR is connected, no action is taken to connect one.
   - If a TTR is connected, the inter-digit timer is started.
   - The dial-ahead digits are analyzed up to the maximum number specified for this **collect** command or up to the first pound (***#***) digit. If an asterisk (*******) is found, the digits up to and including the ******* are

deleted and any additional digits are again analyzed. Any remaining dial-ahead digits are to be saved for use by a subsequent **collect** command.

3. Once a TTR has been connected to a call, and a valid announcement extension has been administered for the **collect** command, the system reacts as follows:
   - If the announcement to be connected is busy (there are no available announcement ports) and the queue for the announcement is full or there is no queue, the calling party continues to hear the current feedback. The system then waits five seconds and tries again to connect the call to the announcement. This process continues until the call is successfully queued or connected to the announcement, or the calling party disconnects from the call.
   - If the announcement to be connected is busy (there are no available announcement ports) and the queue for the announcement is not full, the call is queued for the announcement.
   - If an announcement port is available (either initially or after system retry), or if the queued request for the announcement has been fulfilled, any previous calling party feedback is removed from the call, and the calling party is connected to the announcement.
4. If no announcement extension is administered for the **collect** command or if the announcement extension is not administered or recorded within the system; the calling party receives silence (the system attempts to collect digits without the prompt), and the inter-digit timer is started and answer supervision is returned.
5. If an announcement was connected in one of the previous steps, the system reacts as follows:
   - If the calling party enters a touch-tone digit prior to the completion of the announcement, the call is disconnected from the announcement, the inter-digit timer starts, and the collected digit is analyzed.
   - If the announcement completes before the calling party enters a touch-tone digit, the system starts the inter-digit timer.
6. At this point the system is doing digit collection and analysis. In addition the inter-digit timer is activated. The system continues digit collection for this command until one of the following occurs:
   - The maximum number of digits specified has been collected
   - A pound (#) digit is collected (signifying end of dialing)
   - The inter-digit timer expires.

   The system then analyzes the collected digits and reacts as follows:

- If the digit is an * (signifying an error was made while entering digits), the system deletes all digits collected for the current **collect** command and restarts the inter-digit timer. The announcement is not replayed.
- If the calling party has entered the maximum number of digits specified, if the digit is a pound (*#*), or if the inter-digit timer expires, then the inter-digit timer is disabled, and vector processing continues at the next vector command (the **collect** command is successful). However, the switch continues to collect any subsequent dialed digits (including *#* and *** digits) to allow for the dial-ahead capability. These additional ''dialed ahead'' digits are saved for use by subsequent **collect** commands.
- When the TTR is disconnected due to a **route-to number**, **route-to digits**, or an **adjunct routing** step, all dial-ahead digits are discarded. This means that, following a failed **route-to** or **adjunct routing** step, a subsequent **collect** *digits* step always requires the caller to enter digits (unless digits are provided by ASAI when routing the call).

  Dial-ahead digits are available for use only by subsequent **collect digits** commands. The digits are never used by other vector commands that operate on digits (for example, **route-to digits**, **goto...if digits**, etc). In addition, these digits are not displayed as part of the CALLR-INFO button operation until they are collected via a **collect digits** command.
- The TTR required by the touch-tone phone user to collect digits is disconnected. This occurs under the following conditions:
  - Successful or unsuccessful **route-to number** step is encountered during vector processing except where the **route-to** ***number*** is a VDN extension.
  - Successful or unsuccessful **route-to digits** step is encountered during vector processing except where the **route-to** ***number*** is a VDN extension.
  - Successful or unsuccessful **adjunct routing** step is encountered during vector processing.
  - Successful or unsuccessful **converse-on** step is encountered during vector processing.
  - Timeout occurs, during which time the caller does not dial any digits, asterisks (***) or pound signs (*#*).

Goto step <*step #*> if digits <*comparator*> <*digits*> or if unconditionally

In this command, ***<step #>*** is an assigned step number. ***Digits*** (in the **if digits** ***<comparator>*** part of the command) refers to the digits entered by the calling party (via the last **collect digits** command), and ***<digits>*** is a digit string from one through 16 digits in length.

The **if digits *<comparator> <digits>*** part of this command is optional. The unconditional form of the command causes vector processing to continue at the specified step (command). With the **if digits *<comparator> <digits>*** part, this command causes vector processing to continue at the specified step (command) if the digits entered by the calling party for the last **collect digits** command meet the conditions described by the comparator in reference to the digits administered in the digit string of this command. This command is used for both skipping vector commands and looping through vector commands.

If the digits entered by the calling party for the last **collect digits** command do not meet the conditions described by the comparator in reference to the digits in the administered digit string, then vector processing continues at the next vector step.

Goto vector <*vector #*> if digits <*comparator*> <*digits*> or if unconditionally

In this command, ***<vector #>*** is an assigned vector number. **Digits** (in the **if digits *<comparator>*** part of the command) refers to the digits entered by the calling party (via the last ***collect digits*** command), and ***<digits>*** is a digit string from 1 through 16 digits in length. This command allows for the chaining of vectors in such a way that queuing or feedback treatment applied in the initial vector continues as vector processing proceeds in the subsequent vector.

The **if digits *<comparator> <digits>*** part of this command is optional. The unconditional form of the command causes vector processing to continue at the specified vector. With the **if digits *<comparator> <digits>*** part, this command causes vector processing to continue at the specified vector, if the digits entered by the calling party for the last **collect digits** command meet the conditions described by the comparator in reference the digits administered in the digit string of this command.

If the digits entered by the calling party for the last **collect digits** command do not meet the conditions described by the comparator in reference to the digits in the administered digit string, then vector processing continues at the next vector step.

Route-to number <*#*> with cov <*y/n*> if digit <*comparator*> <*digits*>

In this command, ***<#>*** is the number to route to if the digit comparison succeeds, **with cov *<y/n>*** refers to whether coverage should apply when routing, ***<digit>*** (in the **if digit *<comparator>*** part of the command) refers to the digit entered by the calling party (via the last **collect digits** command), and ***<digit>*** is a single administered digit from 0 through 9 or #.

This command is simply a conditional **route-to number** command. It allows a call to be conditionally routed to a specified destination based on a single digit entered by the calling party. In other words, if the digit entered by the calling party in the last **collect digits** command is the same as the administered digit, then the command attempts to route the call to the specified destination. If more than one digit was collected when this command is encountered, it fails and vector processing continues at the next command. Everything that applies to the **route-to number** command also applies to this command.

This command can only be used for routing based on single-digit comparisons. If a customer application requires comparisons based on more digits, then the appropriate combination of a **goto** and **route-to number** command must be used. If coverage applies and the digits are a system extension, vector processing terminates as soon as this command is encountered.

Route-to digits with coverage <*y/n*>

In this command, **with coverage *<y/n>*** refers to whether coverage should apply when routing, and **digits** refers to the digits entered by the calling party (via the last **collect digits** command). These digits represent a destination which may be any of the following:

- An internal extension (such as a split/hunt group, voice terminal, announcement, and so on)
- A VDN extension
- An attendant
- A remote extension
- An external number such as a TAC or AAR/ARS FAC followed by a public or private network number (however, routing to just such a TAC or to the "AAR/ARS Partitioning" feature access to return dial tone is never allowed)

This command allows a call to be routed to the destination specified by the digits entered by the calling party as a result of the last **collect digits** command. This command can be used to implement an automated attendant function. The optional coverage parameter determines whether coverage should apply when routing. If coverage applies and the digits are a system extension, vector processing terminates as soon as this command is encountered.

Stop

This command is used to stop processing of any subsequent vector commands in the vector. In addition, if a TTR is allocated to the call when the **stop** command is encountered, the TTR is disconnected since there is no purpose in allowing digits to be entered when subsequent commands (such as **collect digits**) are no longer processed. The calling party continues to hear the current feedback that was on the call when the stop was encountered. In addition, the **stop** command does not affect the queued status of a call (for example, if the call was in a split's queue when the stop was encountered, it remains in the split's queue after the stop has been processed). An implicit stop command is processed, if necessary, following the last administered command in a vector. If a **stop** command or an implicit stop command is executed when the call is not queued, the call is dropped.

Vector Processing and Calling Party Feedback

Vector processing starts when a call routes to a VDN. The called VDN then passes control of the call to its assigned vector. Processing begins at Step 1 in the vector and proceeds sequentially through the vector unless a **goto** command is encountered. Unadministered steps are skipped and an implicit **stop** command is processed after the last administered command within a vector.

Calling party feedback is provided by the vector. However, incoming CO, FX, and WATS calls hear CO ringback until answer supervision is supplied (at which point vector feedback is applied). The initial calling party feedback for non-CO/FX/WATS (DID type) calls is silence. This silence continues until the vector processes a command which alters this feedback. Some commands that change feedback are **announcement** commands, which apply a recorded announcement, **wait-time**, and **messaging** and **route-to** commands which apply ringback. More details on calling party feedback for vectoring are found in the *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 555-230-520.

Agent Set Caller Display Information

The customer may choose to include the CALLR-INFO button at agents' display stations to help process calls that are serviced by the Call Prompting feature. However, in G3V2 and later releases, if the agent has a two-line display set, such as a 7407 or Callmaster, the collected digits are automatically displayed on the second line. These digits remain on this line until they are overwritten, even after the call is released by the agent. For other display sets, the agent must press the CALLR-INFO button to display the collected digits.

The CALLR-INFO button displays information in the following format: ''x=Info:1234567890'' where x is a call appearance letter (a, b, c, and so on) and 1234567890 are the digits collected for the last **collect digits** command. Any digits that were ''dialed ahead,'' and not asked for explicitly by the most recently executed **collect digits** command, are not displayed as a result of a CALLR-INFO button depression.

When digits have been collected via the Call Prompting feature, and an attendant or display-equipped voice terminal user presses the CALLR-INFO button, the display is updated with the information described in the previous paragraph. This information remains displayed for 10 seconds, unless an incoming call is received, or the active call changes status (for example, another party is added to a conference). In this case, the display changes to show the new call identification information.

If the answering agent needs to display the collected digits again, the CALLR-INFO button can be depressed again to repeat the above operation as long as the agent is active on the call, the call is still alerting (for display-equipped voice terminals except CALLMASTER), or if the agent attempts to transfer or conference the call. If the call is on hold, or if the agent attempts to conference the call, the CALLR-INFO button cannot be used to display information for an alerting call until the call is answered.

The CALLR-INFO button works when a call is alerting a voice terminal or attendant console and after a voice terminal user or attendant has answered a call. If no digits were collected, the attempt is denied.

Call Prompting Applications

Automated Attendant

The Automated Attendant application allows customers, particularly those who do not have DID trunks, to route incoming calls to a desired location without the use of an attendant. This allows the customer to reduce costs by reducing the need to employ live attendants. A sample prompting vector that implements the Automated Attendant application is shown in Screen 3-27.

In the following sample vector, the calling party is prompted to enter the destination extension (up to five digits) of the party he/she would like to reach. For example, the announcement at extension 300 might say, ‘‘Enter the five-digit extension of the party you wish to reach.’’ The illustrated vector collects the digits and then routes to the destination. If the **route-to digits** command fails (for example, the calling party is an outside rotary user, the calling party does not enter any digits, etc.), the **route-to number** command executes and routes the call to the attendant (default). On the other hand, as long as the destination is a valid system extension, the **route-to digits with coverage y** succeeds, coverage applies, and vector processing terminates. (Even if the destination is busy, vector processing terminates since call coverage processing takes effect.) If an invalid extension number is entered, Step 3 fails and Step 4 is executed.

```
1. collect 5 digits after announcement 300
2. route-to digits with coverage y
3. route-to number 0 with cov n if unconditionally
4. stop
```

Screen 3-27. Call Vector/Automated Attendant Application

Data In/Voice Answer (DIVA)

The Data In/Voice Answer (DIVA) application allows a calling party to receive information on a topic selected at a prompt. In addition, DIVA provides a means to partition different groups of users so that their calls may be handled more efficiently. A sample prompting vector that implements the DIVA application is shown in Screen 3-28.

In the following example, the calling party is prompted, by ***announcement extension 310***, to enter a ***1*** or ***2*** to receive information on two different subjects. Then one of the following occurs:

- If the calling party fails to enter a ***1*** or ***2***, an announcement (extension 320) is connected (with an explanation of options) and the calling party is routed to the attendant.
- If a **2** is entered, the calling party receives the chosen information (**announcement ext. 313**) and vector processing terminates.
- If a ***1*** is entered, the calling party is again prompted by **announcement extension 315** to enter a ***1*** or ***2*** to further identify the type of information that is being requested. At this prompt, if either a ***1*** or ***2*** is entered, the calling party receives an announcement on the chosen topic. If anything else is entered, the calling party is routed to an agent to receive additional information or assistance. (Extension 50000 is a split extension.)

  In addition, the calling party may utilize the Dial-Ahead capability and enter the digit string ***1 1*** or ***1 2*** at the initial prompt. This allows the calling party to bypass the secondary prompt and get to the information he or she desires more quickly.

```
 1. collect 1 digits after announcement 310
 2. goto step 6 if digits = 1
 3. goto step 15 if digits = 2
 4. announcement 320
 5. route-to number 0 with cov y if unconditionally
 6. collect 1 digits after announcement 315
 7. goto step 11 if digits = 1
 8. goto step 13 if digits = 2
 9. route-to number 50000 with cov y if unconditionally
10. stop
11. announcement 311
12. stop
13. announcement 312
14. stop
15. announcement 313
```

Screen 3-28. Prompting Vector/DIVA Application

Data Collection

The Data Collection application provides the system with a method to collect digits from a calling party which can be used by an adjunct to assist in the handling of the call. For example, the calling party could be prompted to enter an account number which could be used by an adjunct to retrieve calling party account information. This information could then be displayed on a data terminal screen to be used by an agent. This capability can be fully automated using an integrated adjunct via the CallVisor ASAI, "Inbound Call Management (ICM)" feature. Alternatively, the Caller Information CALLR-INFO button is available, which allows the prompted information to be displayed on the answering party's voice terminal. The answering party can then use this information to manually retrieve calling party account information via an external adjunct or host computer. A sample prompting vector that implements the Data Collection application is shown in Screen 3-29.

In the following example, the calling party is prompted to enter a 10-digit account number representing his or her customer account. The vector then routes the call to a split to be answered by an agent. Using the CALLR-INFO button, the answering agent can display this account number and use it to retrieve calling party account information from the customer's host computer.

```
1. collect 10 digits after announcement 300
2. queue-to main split 3 pri m
3. stop
```

Screen 3-29. Prompting Vector/Data Collection Application

Message Collection

The Message Collection application gives the calling party the option of not waiting to be serviced by an agent, but instead leaving a message for the agent or the agent's associated split. For example, a customer with both the Call Prompting and Call Vectoring features enabled can let a calling party waiting in a split queue be given a choice (via a prompt) of remaining in queue or leaving a message for that split (or agent).

Message collection allows a calling party who does not have time to wait to be serviced to leave a message (name, phone number, reason for calling, and so on) so that an available agent may return the call at a later time. If the customer prompt requested that certain information be left in the message, a return call may not even be necessary. A sample prompting vector that implements the Message Collection application is shown in Screen 3-30.

In the following example, the calling party is prompted to enter a ***1*** to leave a message, or enter a ***2*** to speak with an agent for more information about the company. If anything other than a ***1*** or ***2*** is entered, an announcement is connected (''You have not selected a valid option'' or ''Operator will be with you momentarily to handle your call.'') The caller is then sent to the attendant via Step 5.

If a ***1*** is entered, the calling party is connected to AUDIX so that he or she may leave the pertinent information and be placed on the company mailing list. If the switch cannot connect to AUDIX (for example, if the AUDIX link is down), an announcement is played informing the caller to try and call back later. If a ***2*** is entered, the call is routed to a split to be answered by an agent to receive additional information about the company.

```
 1. collect 1 digits after announcement 300
 2. goto step 6 if digits = 1
 3. goto step 9 if digits = 2
 4. announcement 301
 5. route-to number 0 with cov n if unconditionally
 6. messaging split 20 for extension 50000
 7. announcement 302
 8. stop
 9. queue-to main split 3 pri m
10. stop
```

Screen 3-30. Prompting Vector/Message Collection Application

Considerations

With the Call Prompting feature, a caller is prompted to enter information from his or her DTMF phone. The system then uses this information to route the call (if necessary) to the right person or group of persons. The call may also be routed to an announcement, if desired. In addition, a caller may be asked to enter an account number or some other type of number to be used in handling the call. Since the system retrieves all of this type of information, and processes the call accordingly, time is not wasted trying to determine the type of call and to whom it is supposed to go. Also, agents spend less time gathering information and can handle more calls.

The **route-to number *<#>* if digit** command can only be used to conditionally route a call based on a single digit comparison.

Vector processing executes a maximum of 1,000 vector commands for a given call. After executing the 1,000th command, an implied **stop** command is executed. There are TTR considerations. The default timer results in TTR always being dropped if no digits are dialed in a specified period. With G3V3 and earlier releases this specified period is ten seconds. With G3V4 and later releases the time is administrable. Quantities of TTRs on the TN744 have to be appropriately sized for each application.

Interactions

The following features and functions interact specifically with the Call Prompting feature. Interactions with the Call Vectoring feature are found in the Call Vectoring feature description.

- Answer Detection

  Call prompting competes with call classifier for ports on the TN744.

- Answer Supervision

  Answer supervision is only returned once during the life of a call. With respect to prompting commands, answer supervision is returned in response to a **collect digits** or an **announcement** command. In addition, if a call is answered and answer supervision hasn't previously been sent, it is then sent.

- AUDIX

  If a **route-to** command in a vector calls AUDIX, the call is treated as a direct call to AUDIX and the calling party may retrieve his or her messages.

- Authorization Codes

  Authorization codes are disabled with respect to routing via VDNs. In other words, if authorization codes are enabled, a **route-to** command in a prompting vector accesses AAR or ARS and the VDN's FRL does not have the permission to utilize the chosen routing preference, then no authorization code is prompted for and the **route-to** command fails.

- AAR/ARS

  Any **route-to** command in a vector can dial an AAR/ARS FAC followed by other digits. A **route-to** command cannot just specify the AAR/ARS FAC and return feature dial tone.

- AAR/ARS Partitioning

  When a **route-to** command in a vector dials an AAR or ARS FAC, the COR associated with the VDN is used to determine the AAR/ARS PGN. The PGN then determines the appropriate routing tables to use for the call.

- Automatic Incoming Call Identification

  When a call terminates to a display station via a **route to number with coverage n** or **route to digits with coverage n** command, the station displays the following information: "Originator Name (trunk or station name) to VDN name."

- Automatic Call Back

  It is recommended that no AUTOMATIC CALL BACK (ACB) buttons be assigned to stations involved in third party calls (including domain control).

- Bridging

  If a principal extension with bridged appearances receives a call via a **route-to** command, that extension's bridged appearances are be updated to reflect the state of the call on the principal extension's call appearance.

- CallVisor ASAI

  Call prompting competes with CallVisor ASAI switch-classified calls for ports on the TN744 or TN2182. ASAI-provided digits can be collected by the Call Vectoring feature via the "collect" vector step as dial-ahead digits.

- Call Coverage

  In G3v2 and later releases, a VDN may be administered as the last point in a coverage path.

  When a **route-to digits with coverage y** or **route-to number with coverage y** command terminates to a system extension (not a VDN), then vector processing terminates and normal routing, termination and coverage apply to this call.

- Call Forwarding All Calls

  A **route-to** command in a vector cannot dial the Call Forwarding All Calls FAC.

  If a **route-to** command in a vector calls a system extension (split/hunt group, voice terminal extension, and so on) that has Call Forwarding All Calls active, the call is forwarded to the designated (forwarded to) destination.

  Calls that are forwarded to a VDN are considered successfully forwarded and call coverage is disabled. (For example, if a **route to digits with coverage y** command is performed within a vector that has been accessed via call forwarding, coverage does not apply and the command is treated as a **route to digits with coverage n** command.)

- Call Pickup

  If a **route-to** command calls a voice terminal extension that is a member of a pickup group, that call can be picked up by another pickup group member.

- Call Waiting Termination

  When calls are routed to analog voice terminals via a **route-to with coverage n** command, the call is considered successful if the analog extension is idle. If it is not idle, the **route-to** is considered unsuccessful and call waiting does not apply. In other words, call waiting is disabled for these **route-to** commands. On the other hand, for a **route-to with coverage y** command, call waiting does apply if appropriate.

- Call Vectoring

  Call Prompting is administered through Call Vectoring administration. If only Call Vectoring is enabled, vectors can be administered using only Call Vectoring commands. If only Call Prompting is enabled, vectors can be administered using only Call Prompting commands. When both features are enabled, all the commands associated with vectoring and prompting are available.

- CAS

  If a **route-to** command calls the attendant and CAS is enabled, the call completes to the CAS attendant if an RLT trunk can be seized.

- Coverage All Calls

  The **route-to with coverage y** command works like any other call. All other types of **route-to** commands fail.

- Coverage Callback

  Coverage Callback only operates successfully for **route-to with coverage y** commands that terminate successfully to a coverage point. The covering user is able to initiate coverage callback for the principal.

- Coverage Incoming Call Identification

  If a coverage call terminates at a covering user's display-equipped voice terminal, via a **route-to with coverage y** command, the voice terminal user that the vector routed to is displayed as the called party instead of the VDN.

- DCS

  A **route-to** command in a prompting vector can route to a UDP extension and provide DCS transparency (Distinctive Alerting). In addition the DCS Call Forwarding, "Call Waiting Termination", "DCS Leave Word Calling", and "DCS Multi-Appearance Conference/Transfer" features apply, where appropriate, to calls routed via a **route-to** command.

- DOD

  DOD can be provided via a **route-to** command within a vector. The COR of the VDN is used to determine calling party permissions/restrictions. A digit string must follow the TAC in the **route-to** command. A **route-to** command cannot be used to return trunk dialtone.

- FRL

  If a **route-to** command dials an external number via AAR/ARS, the FRL associated with the VDN COR is used to determine the accessibility of a routing preference in an AAR/ARS pattern.

- Go To Cover

  Go To Cover operates correctly when a **route-to with coverage y** command terminates successfully at an internal destination. At all other times, if the GO TO COVER button is depressed, the feature is denied.

- Hold

  If a call is put on hold during the processing of a collect command, the collect command is restarted, beginning with the announcement prompt, when the call is taken off hold. All dialed-ahead digits are lost. Similarly, if a call to a vector is put on hold, vector processing is suspended when a collect step is encountered. When the call becomes active, the collect step resumes.

- Hunting

  A **route-to** command can call a hunt group.

- Inbound Call Management

  The Call Prompting feature can be used to collect information from a calling party which may later be used by an adjunct to assist in the handling of the call.

- Individual Attendant Access

  A **route-to** command can dial an individual attendant extension.

- Inter-PBX Attendant Calls

  If a **route-to** command calls the attendant and this feature is enabled, the call completes to the Inter-PBX attendant.

- ISDN-PRI

  A **route-to** command in a prompting vector can route calls over ISDN-PRI trunks.

- LWC

  LWC operates correctly when a **route-to with coverage y** command terminates successfully at an internal destination. At all other times, if the LWC button is depressed, activation is denied.

- Night Service

  Route-to commands that route to destinations with night service activated, redirect to the night service destinations.

- Priority Calling

  A **route-to** command cannot dial the priority calling FAC.

- Queuing

  Queuing applies, where appropriate, to any calls that route to an attendant or hunt group via a **route-to** command.

- Recorded Announcements

  Recorded Announcements can be accessed via a VDN through the use of the **announcement** command or the **route-to** command, if the destination is an announcement extension. In addition, the **collect digits** command has the option to connect an announcement when prompting for digits.

- Redirect Notification

  Redirect notification applies when a call is about to redirect to coverage via a **route-to with coverage y** command.

- Ringback Queuing

  External call attempts made via **route-to with coverage n** commands are not queue via Ringback Queuing when all trunks are busy.

- Rotary Dialing

  Outside users using rotary dialing are not able to enter digits requested via a **collect digits** command. The inter-digit time-out takes effect and a **collect digits** command is skipped for these users. A default **route-to** command (such as **route-to attendant**) should always be provided.

- SAC

  When a **route-to with coverage y** command terminates to a system extension with SAC active, the call is treated as a normal internal call to a station having SAC active (call coverage via SAC applies). When a **route-to with coverage n** command terminates to an extension having SAC active, SAC is ignored. If the station has an idle appearance, the call terminates and the **route-to** is successful. Otherwise, the **route-to** command is considered unsuccessful and vector processing continues at the next vector command.

  If a **route to with coverage y** command terminates to a system extension and the station user then activates SAC, the system attempts to redirect the call to coverage (due to SAC coverage criteria). When a **route-to with coverage n** command terminates to a system extension, activation of SAC by the station user (after termination) is ignored.

- CDR Account Codes

  A **route-to** command cannot dial the CDR account code FAC.

- Subnet Trunking

  Subnet trunking applies to any AAR/ARS call dialed via a **route-to** command.

- Temporary Bridged Appearance

  A Temporary Bridged Appearance is maintained at the principal's extension when a **route to with coverage y** command terminates to a system extension and redirects to coverage. However, if coverage is to AUDIX or the principal is an ACD agent, no Temporary Bridged Appearance is maintained at the principal.

- TEG

  A **route-to** command can call a TEG.

- Transfer

  If a call to a VDN is transferred during a **collect** command, the **collect** command restarts when the transfer is complete, and all dialed-ahead digits are lost. Similarly, if a call to a vector is transferred, vector processing is suspended when a **collect** step is encountered. When the transfer is complete, the **collect** step resumes. This also applies to attendant extended calls

- TCM

  A TCM is sent when a **route-to** command dials a seven-digit ETN or 10-digit DDD number via AAR/ARS. This TCM is the FRL associated with the VDN COR.

- UDP

  A **route-to** command can call a UDP extension.

Administration

The Call Prompting feature is administered on a per-system basis by the system manager. The Call Vectoring feature must be administered as described in the Call Vectoring feature. In addition, the following items must be administered specifically for the Call Prompting feature.

- The Call Prompting feature must be enabled on the 'System-Parameter Customer-Options' form and must be done by an authorized AT&T employee.
- Any display-equipped voice terminal or attendant can be administered with a Caller Information CALLR-INFO button.
- With G3V4 and later releases the Call Prompting feature timeout is administrable on the 'Feature-Related System Parameters' form.

Hardware and Software Requirements

Each the "Call Prompting" feature announcement requires:

— One port on a TN750 Integrated Announcement circuit pack or

— Announcement equipment and one port on a TN742 or TN746B (Mu-law) Analog Line circuit pack.

DTMF receivers are required to accept the touch-tone digits entered by the call prompting users. The TN744 Call Classifier circuit pack or the TN2182 Tone Clock circuit pack (required for the "Call Prompting" feature) provides eight DTMF receivers. G3i allows a maximum of 80 TN744 DTMF receivers (10 circuit packs); G3r allows a maximum of 200 TN744 DTMF receivers (25 circuit packs). Beginning with G3V4, TN744 DTMF receivers can also be used as general purpose DTMF receivers in the system. Other DTMF receivers such as those on the TN748C circuit pack (TN420C, TN744 support A-law) are still required for normal call processing and maintenance features.

"Call Prompting" software is required. "Call Vectoring" software may be required depending on the application. If external CMS is being used, "Call Vectoring" software for CMS is always required if call prompting (with or without call vectoring) is active on the switch.

Call Vectoring

Feature Availability

This optional feature is available with G3vs/G3s PBP, G3i, and G3r, and is not available with G3vs/G3s ABP.

Description

This section gives a basic description of the capabilities of the "Call Vectoring" feature. For instructions for creating and troubleshooting vectors, see the *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 555-230-520.

The "Call Vectoring" feature provides processing of incoming and internal calls according to a programmed set of commands. The commands, called Vector commands, determine the type of processing that specific calls receive. Vector commands may direct calls to on-premise or off-premise destinations, to any hunt group or split, or to a specific call treatment such as an announcement, forced disconnect, forced busy, or delay treatment.

It is possible for the system to collect digits from the user, route calls to a destination specified by those digits, and/or do conditional processing according to those digits ("Call Prompting" feature). The "Call Prompting" feature utilizes the "Call Vectoring" feature and a set of specialized vector commands. Also, the Lookahead Interflow feature uses the "Call Vectoring" feature for its operation. Expert Agent Selection (EAS) uses vectors to direct calls to agents with specific skills. Also see Voice Response Integration (VRI) for information about integrating call vectoring with the capabilities of the VRUs and the CONVERSANT Voice Information Service.

Vector Directory Numbers and Vectors

Calls access vectors using VDNs. A VDN is a ‘‘soft’’ switch extension number that is not assigned to a physical equipment location. Access to a VDN may occur in many ways. Since a VDN is an extension, it can be accessed in almost any way that an extension can be accessed. The primary ways that a VDN can be accessed are:

- Internal call — The VDN extension can be dialed from another extension on the switch.
- CO trunk — A CO trunk may be mapped to a VDN (as an incoming destination or night service extension).
- Non-CO trunk — A call may come into a DID or PRI trunk and connect to a VDN extension.
- LDN — A VDN may be the night destination for an LDN.

- From a *route-to* step within a vector.
- As a coverage point

A Call Vector is a set of vector commands (described later in this chapter) to be performed for an incoming, outgoing or internal call. As previously described, vectors are accessed by VDNs. Each VDN maps to one vector. However, several VDNs may map to the same vector. The answering user sees the information (such as the name) associated with the VDN on his or her display and can respond to the call with knowledge of the dialed number. This operation provides DNIS (described later in this description).

Applications

There are many different applications for the "Call Vectoring" feature. However, call vectoring is primarily used to handle the call activity of ACD splits/skills. Call Vectoring can also manage a queue by keeping calls queued in up to three splits/skills (with four different priority levels) while also providing a series of other processing options. Other common applications include:

- Special treatment for selected callers.

  For example, calls from preferred credit card customers may receive priority treatment, but they do not have to be handled by a separate split. Agents in the same split can handle both preferred customers and all other customers. A call can be queued into one of four priority levels, and calls to different VDNs (and vectors) could go to different levels, with preferred customers having top priority. This means that when all agents are busy in this split, calls from preferred customers would go to the top of the queue ahead of other callers already in the queue.

- Night treatment.

  During non-business hours, the Call Vector could route calls to a specified destination such as an announcement and a disconnect. During business hours, the vector could send calls to splits/skills for connections with agents, or queue normally.

- Off-loading periodic excess calls resulting from promotions, seasonal trends, or regular daytime fluctuations in calls.

  A vector can test a split for the number of calls already in queue. If the number is above a certain threshold, the vector bypasses that split and route the call to someone else, such as an attendant. However, if the number of calls queued to the split is below the threshold, the vector would queue the call to that split.

- Information Announcements for the Calling Party

  The human intervention needed to distribute common messages can be minimized with information announcements. A group of people with a common interest can be instructed to call a specific number (VDN) that terminates to a specific announcement vector. The vector's announcement can be periodically updated to provide current information to the callers. Vectors providing information announcements are easily programmed by the System Manager.

What Happens When a Call is Processed by a Vector

General

When an incoming or internal call goes to a VDN, the VDN directs the call to a specific call vector. When the call goes to a vector, the call's routing and treatment is determined by the commands in that vector. Processing starts at Step 1 and proceeds sequentially through the vector unless a *goto* command is encountered. Any steps that have been left blank during administration are skipped. The process automatically stops after the last step in the vector.

Call vectoring allows the chaining of vector steps and vectors through the use of a *goto* command. In other words, one vector can direct the call to another vector or VDN, which can in turn direct the call to yet another vector, and so on.

The "Call Vectoring" feature has an execution limit of 1,000 steps. Once a call enters vector processing, a loop counter keeps track of the number of vector steps executed. If the loop counter exceeds 1,000, a *stop* command is executed. The loop counter remains in effect across *goto vector* commands. This execution limit is provided as a means of system recovery from vector programmed infinite loops.

Three examples of how the "Call Vectoring" feature functions are shown in Figure 3-1. An outside call to a VDN is processed based on the various properties of the VDN form. Handling of the call is based on the commands associated with the vector. Displayed information is dependent on the setting of the “VDN Override” field.

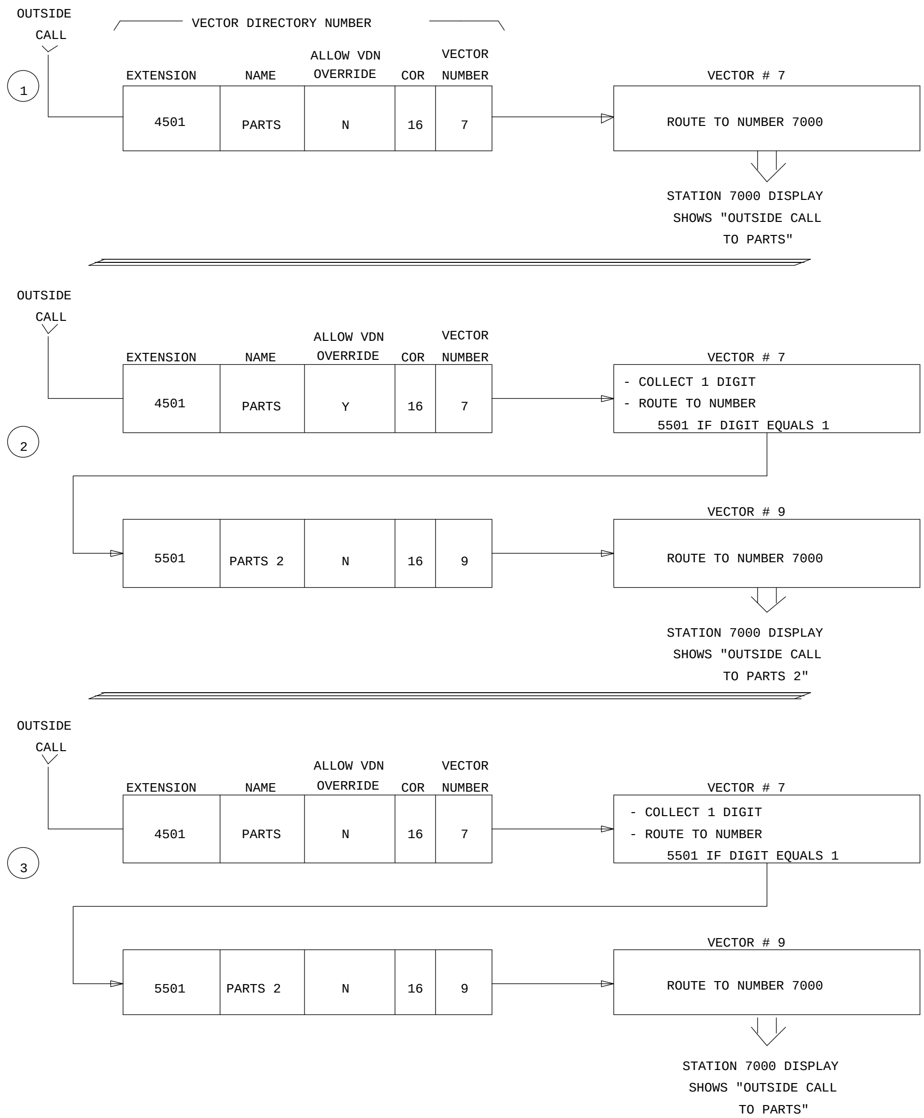


Figure 3-1. Typical VDN Call Processing Examples

Calling Party Feedback

The initial feedback that a caller hears as a call is processed by a vector depends on how the call originates. The call may originate as one of the following:

- An internal call from another system user

- An incoming call over a non-CO (DID) trunk
- An incoming call over a CO trunk

If the call is an internal or non-CO call, the calling party hears silence until one of the following occurs:

- An announcement is played
- A *wait-time* command with system music, ringback, or alternative audio/music source is processed
- A *busy* command is processed
- The call is alerting at an station

If the call is a CO call, the calling party hears CO ringback until one of the following occurs:

- An announcement is played
- A *wait-time* command with system music or alternative audio/music source is processed
- The call is answered

Additional Information

If the call is a CO call and answer supervision has been previously supplied (by processing of an *announcement* or *wait-time* command), the caller hears feedback specified in the vector commands:

- An announcement when any announcement command is processed
- Ringback, silence, system music or an alternative audio/music source when a *wait-time* command is processed
- Busy when a *busy* command is processed
- Ringback when the call is alerting an extension

After a *wait-time* command is processed, the calling party continues to hear that specified treatment unless an announcement is being played, another *wait-time* command with a different treatment is processed, the call is alerting at an extension, a *busy* command is processed, or vector processing terminates as whenever an agent becomes available or a successful *route-to* command is executed.

If a step cannot be executed, the calling party continues to receive the feedback to which they are currently listening and vector processing continues at the next step.

When a call terminates to an agent or extension, the calling party hears ringback. Any announcement or other treatment that is currently being heard is disconnected and ringback is supplied.

After an announcement completes, the calling party hears silence until another command that specifies treatment is processed (for example, *wait-time*, *announcement*, *busy*), or the call terminates at an extension.

If a call is routed off-premises via a trunk, the calling party hears the standard call progress tones and/or far-end feedback only if answer supervision has been passed or the answer detection timer has expired.

If the calling party disconnects at any time while the vector is being processed, vector processing is terminated, the call is removed from any queues, and the call is torn down.

Vector Commands

A Call Vector is a set of vector commands. Vector commands determine the type of processing that specific calls receive. The basic vector commands are listed below and described in the paragraphs that follow. Please note that most of these commands require one or more parameters (for example, an extension number is required with the Announcement command). For more information on these commands and how they should be used, refer to the *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 555-230-520.

- Adjunct Routing (used only with the ASAI feature)
- Announcement
- Busy
- Check-Backup Split/Skill
- Collect Digits (used for Call Prompting)
- Converse-on split/skill
- Disconnect
- Goto Step
- Goto Vector
- Messaging
- Queue-To Main Split/Skill
- Route-To Number
- Route-To Digits (used for Call Prompting)
- Stop
- Wait-Time

Adjunct Routing

The *adjunct routing* command is only available with the ASAI feature. This command causes a message to be sent to an adjunct that is requesting routing instructions. This command is described in detail in the "Inbound Call Management (ICM)" description in this chapter.

Announcement

The *announcement* command is used to let callers hear a given announcement. All calls hear the announcement from the beginning. This may result in a delay because the call may have to wait in an announcement queue before the announcement is heard. While a caller is waiting to hear the announcement, the caller continues to hear whatever treatment he or she is currently hearing.

If the announcement's queue is currently full, the call retries the announcement step indefinitely before processing any new vector steps. If the requested announcement is not administered (in the case of external or integrated announcements) or recorded (in the case of integrated announcements only), the current step is skipped and vector processing continues at the next vector step.

When a call is connected to the announcement, the previous treatment is discontinued and answer supervision is sent if it has not already been supplied. During an announcement, if the call is moved from waiting in a split's queue to alerting at an agent's extension, the announcement is disconnected and the caller hears ringback. When the announcement completes and is disconnected, the caller hears silence until a vector step with an alternate treatment is processed or the call reaches an agent's extension.

Busy

The *busy* command causes termination of vector processing and gives the caller a busy signal. The busy takes effect on non-CO trunk calls whether or not answer supervision has been sent; but if the call is a CO (auto-in) trunk call and answer supervision (via *announcement* command or *wait-time* command) has not been sent, the CO ignores the busy and continues giving the caller CO ringback. Non-CO trunks are dropped approximately 45 seconds after busy tone is applied.

Check-Backup Split/Skill

The *check-backup* command checks the status of a split/skill for possible termination of the call to that split/skill. When termination is not possible, queuing at the specified priority is attempted. Termination and/or queuing is attempted if the split meets certain conditions specified as part of the command. The possible conditions are shown below:

- Unconditionally (always attempts to queue)
- Number of available (idle) agents in the specified split
- Number of staffed agents in the specified split
- Number of calls queued at a given priority to the specified split.

This condition tests for calls in queue at the specified priority level and higher.

- Age of the oldest call waiting in the specified split's queue
- The average time it takes for a call to be answered in the specified split. (G3V4 and later releases only)
- The amount of time a call could expect to wait in the queue of the specified split before it is answered (G3V4 and later releases only)

A call may be queued in up to three different splits/skills at the same time. A call remains queued until either vector processing terminates or the call reaches an agent.

Collect Digits

The *collect digits* command lets the caller enter up to 16 digits from a touch-tone phone. An optional announcement may be played first. This step is part of the Call Prompting feature and is described in greater detail in the Call Prompting feature description elsewhere in this document.

Converse-On Command

The *converse-on* command is an enhancement to the Basic Call Vectoring customer option. The *converse-on* Call Vectoring step is specifically designed to integrate a VRU with the DEFINITY switch. Voice Response Integration allows VRU capabilities to be used while keeping control of the call in the switch. The inclusion of VRUs with vector processing provides the following advantages:

- Access to local and host databases
- Validation of caller information
- Text to speech capabilities
- Speech recognition
- Increased recorded announcement capacity
- Audiotex applications
- Interactive Voice Response (IVR) applications
- Transaction processing applications

One of the advantages of Voice Response Integration is that it allows users to make more productive use of queuing time. For example, while the call is waiting in queue, the caller can listen to product information by using an audiotex application or by completing an interactive voice response transaction. In some cases, it may even be possible to resolve the customer's questions while the call is in queue. This can help reduce the queuing time for all other callers during peak intervals.

If the *converse-on* command is successful, it delivers the call to a pre-determined split (skill), which is referred to as the converse split (skill). A converse split is administered and behaves exactly as any other split in the system. The term non-converse split is used to refer to traditional ACD splits accessed by means of *queue-to main* and *check-backup* vector steps.

Once the call is answered by the VRU, the *converse-on* command may or may not pass data to the VRU (depending upon the parameters of the command). Up to two items of data may be passed. This data may be used to select the VRU script to be executed or it may be information passed to the script. See the *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 555-230-520, for more information about the types of data that can be passed to the VRU.

Once the VRU script is completed, the VRU may or may not return data to the DEFINITY system. Whether or not data is returned, once the VRU drops the line to DEFINITY system, vector processing is then re-activated on the switch starting with the vector step which follows the *converse-on* step.

Digits returned by the VRU are treated as dialahead digits. Call prompting must be optioned to collect and use these digits. The rules for collecting and processing VRU-returned digits are identical to those for Call Prompting.

Digits returned from the VRU can be used in the following way:

- Displayed on the answering agent's display set (automatically for 2-line displays (such as a 7407 or Call Master) using the ''callr-info'' button for other display sets).
- Treated as an extension in a ''route-to digits'' vector step.
- Used for vector conditional branching in an ''if digits'' vector step.
- Tandemed to an ASAI host.
    - Voice Response Integration also provides the integration of VRUs with ASAI hosts. Since collected digits are also passed to ASAI hosts in ''call offered to domain'' event reports and in ''route request'' messages, caller digits or database information returned from the VRU may also be tandemed through the DEFINITY system to ASAI hosts.

Based on administration, with G3V4 and later releases, a DTMF tone can be used as an answer feedback signal and sent to the VRU when the VRU-to-ACD call is answered by the agent. Another DTMF tone can be used as a disconnect feedback signal and sent to the VRU when the incoming call to the VRU disconnects before it is transferred by the VRU.

Refer to *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 555-230-520, for additional information.

Disconnect

The *disconnect* command ends treatment of a call and removes the call from the switch. This command also allows the optional assignment of an announcement that plays immediately before the disconnect. If the switch has not yet sent answer supervision, it does so immediately before disconnecting a call so there is a charge for calls that terminate with the disconnect command. The exception is on ISDN calls where disconnect can occur without returning answer supervision when an announcement is not played.

During playing of the announcement, if the call is moved from waiting in a split's queue to alerting at an agent's extension, the announcement is disconnected, the caller hears ringback, and the *disconnect* command is aborted.

Goto Step

The *goto step* command is a branching step that provides conditional or unconditional movement to later or earlier steps in the Call Vector.

Conditional branching may be used to compensate for heavy traffic or for night/weekend service. Calls in queue and/or any wait treatment in effect before a *goto* remain in effect after the branch.

The conditions of branching are as follows:

- Unconditionally (always branches)
- Number of available (idle) agents in a specified split
- Number of staffed agents in a specified split
- Number of calls queued at a given priority (or higher) to the specified split
- Age of the oldest call waiting in the specified split's queue
- Time of Day, start time, and end time specified
- Rolling average speed of answer for the specified split or VDN. (G3V4 and later releases only)
- Expected wait time for the specified split or for the call being processed (G3V4 and later releases only)
- Number of calls active in the VDN (G3V4 and later releases only)
- Collected Digits from a *collect digits* step (Call Prompting feature only)
- ANI (caller identity) of the call being processed (G3V4 and later releases only)
- II-digits (type of originating line) of the call being processed (G3V4 and later releases only)

With G3V4 and later releases, ANI, II-digits, and digits collected from a *collect digits* step can be tested against entries in a Vector Routing Table for conditional branching with the *goto* step.

Goto Vector

The *goto vector* command is a branching step that provides conditional or unconditional movement to another vector.

The *goto vector* command is used for chaining more than one vector together. Calls in queue and/or any wait treatment in effect before a *goto* remain in effect after the branch.

The conditions of branching are:

- Unconditionally (always branches)
- Number of available (idle) agents in a specified split
- Number of staffed agents in a specified split
- Number of calls queued at a given priority (or higher) to the specified split
- Age of the oldest call waiting in the specified split's queue
- Time of Day, start time, and end time specified
- Rolling average speed of answer for the specified split or VDN. (G3V4 and later releases only)
- Expected wait time for the specified split or for the call being processed (G3V4 and later releases only)
- Number of calls active in the VDN (G3V4 and later releases only)
- Collected Digits from a *collect digits* step (Call Prompting feature only)
- ANI (caller identity) of the call being processed (G3V4 and later releases only)
- II-digits (type of originating line) of the call being processed (G3V4 and later releases only)

With G3V4 and later releases, ANI, II-digits, and digits collected from a *collect digits* step can be tested against entries in a Vector Routing Table for conditional branching with the *goto* command.

Messaging Split/Skill

This command causes the calling party to be connected to the AUDIX or Message Center split so that he or she may leave a message for the specified extension (call answering service). If the calling party is successfully connected to the AUDIX or a Message Center agent, then vector processing terminates and a message may be left for the specified extension.

Priority can be given to the call by assigning a high priority to the messaging split extension via the COR.

This command functions as follows:

1. If the split number specified in the command is a valid message service split (such as AUDIX or Message Server Adjunct), and either the extension is a valid assigned extension or is administered as “none” (defaults to current VDN), then the system attempts to terminate the call to the message service split for call answering service.
   - If the call is queued to the message service split or if the call terminates to an available message service agent or AUDIX voice port, then the calling party is connected to ringback (successful termination), and vector processing terminates. For DEFINITY AUDIX the VDN number must be included in the VDN name on the switch.
   - If the split queue is full, the AUDIX link is down, all AUDIX voice ports are out of service, or the message service split is DCS-AUDIX and all DCS trunks are busy, then vector processing continues at the next vector command (unsuccessful termination).
2. If call termination was successful and the administered extension (or default VDN) is a message service subscriber, the calling party can leave a message for the specified extension.

   If the extension or VDN was not a subscriber of the message service, one of the following may occur:
   - If the message service split is AUDIX, the calling party receives ringback until he or she disconnects.
   - If the message service split is a Message Server Adjunct, the calling party may be answered by an message service agent but no message is taken since the specified extension (default VDN) is not a Message Server Adjunct subscriber.

Queue-To Main Split/Skill

The *queue-to main split* command sends a call to a split and assigns a queuing priority level to the call if all agents are busy. A call sent with this command either connects to an agent in the split or enters the split’s queue.

A call may be queued in up to three different splits/skills at the same time. A call remains queued either until vector processing terminates or the call reaches an agent. When an agent becomes available in any split to which the call is queued, the following actions are performed:

- The call begins alerting the agent
- The call is removed from any other queues
- Vector processing terminates

If the desired split is one of the splits/skills to which the call is already queued, the call is requeued at the new priority level. This step is skipped and vector processing continues at the next step if any of the following are true:

- The desired split’s queue is full

- The desired split is unstaffed
- All staffed agents are in the auxiliary work mode
- The desired split is not vector controlled
- The call is already queued in this split at the specified priority level
- The call has already been queued to three different splits/skills
- The desired split has no queue and no agents are available

Route-to Number

This command routes a call to the specified number. This digit string represents a destination number which may be any of the following:

- An internal extension (such as a split/hunt group, voice terminal, announcement, and so on)
- A VDN extension
- An attendant
- A remote extension
- An external number such as a TAC or AAR/ARS FAC followed by a public or private network number.

This command allows a call to be routed to a specified destination. The COR associated with the VDN is used to determine any calling party restrictions (such as lack of calling permission, FRL restriction, and so on). If this command is successful (that is, it successfully terminates to a station, seizes a trunk, and so on) then vector processing terminates. Otherwise, vector processing continues at the next vector command.

This command functions as follows:

1. If the number is a system extension or attendant group (not a VDN) the system attempts to terminate the call to the endpoint if one of the following conditions occur:
   - The endpoint is alerted.
   - The endpoint has call forwarding or night service (hunt group) enabled, and the forwarded to (night service) destination is alerted (or in the case of off-premises call forwarding, a trunk is seized).

   The system then provides ringback to the calling party, and vector processing terminates. However, if the call cannot complete successfully (for example, no idle appearance is available), and the *route-to number* step is specified with coverage set to no, vector processing continues at the next vector command. If coverage is set to yes, the call routes to the destination’s coverage path. *Route-to number* with coverage is only available when G3V4 and later release software has been installed either when purchased or as a bugfix release.
2. If the number is a VDN extension, then the following events occur:

- Vector processing terminates within the current vector.
- If the current VDN is administered with VDN override, then the new VDN overrides the current VDN as the VDN to be displayed at the terminating extension.
- Processing of the vector associated with the VDN extension begins.
- All previous vector queue and wait treatment ends.

3. If the number is an AAR/ARS FAC plus digits, or a remote (UDP) extension, standard AAR/ARS processing is performed to select the trunk group and outpulse the digits. If a trunk is seized, vector processing terminates, and the calling party hears feedback provided by the far end. Otherwise, the call cannot complete successfully (no trunks available, FRL/COR restricted, and so on), and vector processing continues at the next vector command.
4. If the number is a TAC plus digits, and a trunk is seized, then vector processing terminates and the calling party hears feedback provided by the far end. Otherwise, the call cannot complete successfully (no trunks available, COR restricted, and so on), and vector processing continues at the next vector command.
5. If the number is any other number (such as an FAC plus digits), vector processing continues at the next vector command (unsuccessful *route-to* command). Note that with G3V4 and later releases, a call can route to a Service Observing FAC.

Route-To Digits

The *route-to digits* command is a Call Prompting command. It attempts to route the call like the *route-to number* command, except that the number routed to is a set of digits collected from the caller (via a *collect digits* step). For more details see the "Call Prompting" feature description elsewhere in this chapter.

Stop

The *stop* command terminates the processing of any subsequent vector steps. After the *stop* command, any calls that are already queued remain queued, and any wait treatment is continued.

Wait-Time

The *wait-time* command is generally used to set a length of time for a call to wait in the main split's queue and to specify what the caller will hear while the call advances in queue. Prior to G3V4 the caller could hear silence, ringback or system music. G3V4 and later releases also allow the caller to hear an alternative audio/music source. A wait time of 0 seconds can be used to change the call treatment without executing a delay. Anytime during a delay if an agent becomes available *wait-time* treatment is cancelled and the call is connected to the agent.

ACD Split/Hunt Group Operation with Call Vectoring

A split is considered Vector Controlled if yes has been selected in the “Vector” field on the Hunt Group Administration screen. Only vector controlled splits/skills are accessible from the following call vectoring commands: *check-backup split*, *queue-to main split*, and *converse-on split.*

Overview

Throughout this description, split is used to specify a hunt group or ACD split. ACD split is used when the group has ACD optioned. Hunt group is used when the group does not have ACD optioned. When EAS is optioned only non-ACD hunt groups and Skills may coexist. Splits and Skills may not coexist.

This section describes the interactions between the Hunting, ACD, and Call Vectoring features. Some key points to note are:

1. Hunt groups and ACD splits/skills may coexist with one another as well as with vector-controlled splits/skills.
2. Vector-controlled splits/skills may be called directly. However, these calls do not receive any announcements, can be forwarded, can be redirected to coverage, or intraflow/interflow to another hunt group.
3. ACD is required to obtain any CMS and/or BCMS split measurements.

Vector Controlled Splits/Skills

Vector Controlled Splits/Skills have the following properties:

- Agents may be logged into multiple controlled splits/skills simultaneously.
- Agents’ voice terminals are not locked while they are logged into a vector controlled split.
- A vector controlled split may also be adjunct controlled. If it is, the more restrictive properties of each controlling function apply.
- A vector controlled split is not mapped to an adjunct via system administration, unless it is also adjunct controlled.
- Request Notification is allowed (as long as the split is not adjunct-controlled). The adjunct receiving notification is the same one requesting it.
- When a vector controlled split is removed via system administration, any active notification request is terminated.
- A vector controlled split cannot be assigned a coverage path. VDNs cannot be assigned a coverage path but can be assigned *to* a coverage path.
- Calls queue when agent is logged out or in AUX work mode.
- When EAS is optioned all ACD hunt groups (skills) must be vector controlled.

- Call Forwarding cannot be activated for a vector controlled split.
- No announcements are associated with vector controlled splits/skills.

Split/Skill Queue Priority Levels

Each call in a split queue can have one of four possible priority levels: Top, High, Medium, and Low. Calls are processed initially by priority level. Within each priority level, calls are processed in first-in, first-out (FIFO) order. A vector can be administered to process queued calls using any of the four priority levels. Calls not controlled by a vector can use only two priority levels: High and Medium.

The default priority level for non-vector controlled calls queued to a split is Medium. An ACD split may assign certain calls to the high-priority level. The high-priority level can be assigned to a call by intraflowing from one ACD split to another ACD split. This is achieved using the “Call Coverage Option” field and “Setting the Priority on Intraflow” field on the Hunt group form of the principal ACD split. The high-priority level can also be assigned to a call from a voice terminal user whose associated COR has priority queuing optioned.

Direct Agent Calls (DACs) have the highest priority of any calls (including those mentioned in the previous paragraph).

Split/Skill Thresholds

The split thresholds, available-agents, and staffed-agents, used by the *check-backup* and *goto* vector commands, have slightly different meanings for hunt groups and ACD splits/skills. For ACD splits/skills, staffed-agents is the number of agents logged-in. Available-agents is the number of agents logged-in and ready to receive an ACD call (for example, Auto-in or Manual-in work mode and not currently on a call). Since hunt groups have no log-in, log-out, or work modes, a staffed-agent is merely an administered agent. All that is checked is the number of administered agents. Similarly, available-agents is the number of agents ready to receive a hunt group call.

When there is no queue, the oldest call waiting is always 0 and the number of queued calls is always 0. When the system time is not set, the time of day and oldest call waiting threshold always fail.

Call Vectoring and PBX Toll Fraud

Abuse of PBX remote access is one way in which unauthorized users obtain services illegally. Call Vectoring features can be used to block unauthorized use of the DEFINITY Communications System remote access feature.

The following are ways Call Vectoring features are used to alleviate toll fraud:

1. By reaching the remote access extension via call vectoring.
2. By replacing the function of the remote access extension by one or more call vectors.

Front Ending Remote Access

Authorized external callers are given a VDN extension to call as opposed to the remote access extension, which is kept private. The corresponding call vector then implements a number of security checks before routing callers to the remote access extension.

The following advantages are possible with this feature.

1. A recorded announcement is played to callers, which declares that use of PBX services by unauthorized callers is illegal and that this call is subject to monitoring and/or recording.
2. Call prompting is used to prompt for passwords.
3. Use of the remote access extension is limited to certain times of the day or days of the week.
4. Vectoring introduces a delay before dial-tone is provided to the caller.

   NOTE:
   Immediate dial-tone is often a searched for by hacker's programs when trying to break into a system.

5. If ANI is available on incoming calls, different VDNs are administered for different geographic areas, such that once a security breach occurs, only callers from that specific region have the capability to abuse the remote access capabilities.
6. If the Information Indicator (II) digits are available on the incoming call, additional security checks based upon the origination information associated with the calling parties telephone line can be made. The II digits may indicate that the inbound call originated from a prison, hotel/motel, or even cellular telephone line and as such would require unique security treatment.
7. Real-time and historical reports on the use of the remote access feature are available from BCMS and/or CMS.
8. Different passwords are used on different days of the week or times of day.
9. Many VDNs are defined which call the remote access extension. In this way, individuals or groups are given their own VDN with unique passwords, permissions and reports. Any abuse of the system or security leak is then attributed to an individual or group.
10. The caller can be routed to a VRU using the *converse-on* step, where more sophisticated security checking takes place, such as speaker recognition.
11. Any call failing any of the security checks is routed to a VDN that routes the caller to security personnel with a display set or to a VRU. Such calls show security violation and possibly also the attempted password on the

display. If the call is passed to a VRU, the VDN, the ANI and/or the prompted digits are captured. BCMS/CMS reports on this VDN give information on how often and when security violations occur.

Replacing Remote Access

In this case, the remote access extension is not used. One or more VDNs are defined which access call vectors employing all the security checks described for *Front Ending Remote Access*. The same reports and monitoring/recording capabilities described for *Front Ending Remote Access* are also possible. Instead of routing the call to the remote access extension, the call vector collects digits from the caller and then routes to the given destination.

Again, multiple VDNs are created for individuals or groups with different security checks and different permissions/restrictions. Destination numbers provided by callers are screened by the vectors and denied if the user does not have permission to access that destination. For example, an individual user is restricted to placing calls to numbers beginning with area codes 303 and 908.

Call Vectoring and the Call Management System (CMS)

The CMS provides the following capabilities to be used with Call Vectoring:

- The CMS can request full translations to initialize its database. The PBX supplies data for VDNs and measured trunk groups which map to VDNs.
- The CMS can retrieve and change vectors on the PBX. However, no information is sent to the CMS when vectors are administered via the Management Terminal.
- The CMS can change the vector the number assigned to VDNs.
- The PBX notifies the CMS when the incoming destination of a trunk group is changed. This happens when a trunk group maps to either a split or VDN.
- The PBX notifies the CMS when a VDN is called. State change messages are also sent as a vector is processed.
- The PBX notifies the CMS when an agent transfers a call to a VDN.

DNIS

Call Vectoring provides DNIS, which allows agents with display-equipped voice terminals to receive visual displays that specify the name of the called VDN.

In traditional ACD arrangements, groups of agents are organized into splits/skills. With this type of arrangement, an agent is trained to answer calls for one specific purpose in an efficient and professional manner. However, ACD managers are recognizing the need to relax this concept of limiting each split to a single call-answering task.

The alternative is to provide splits/skills where each group of agents is proficient with several types of calls. The desired gain is to provide adequate service for the several call types with fewer agents and with less administrative intervention by the ACD manager. Using this approach, the changing staff needs of the several call types are averaged in time, and enough agents are staffed to provide adequate service for the prevailing average load. Where five agents might be needed in each of three smaller splits/skills (15-agent total) to handle three types of calls, only 11 or 12 agents might be needed in the single (more general) split.

The DNIS function provided by Call Vectoring allows each answering agent to know the purpose of each incoming call as the call terminates to the agent's voice terminal. As a result, the natural efficiencies of the single-split/single-call type arrangement are not compromised. With the display information provided by DNIS (for example, the name of the called VDN), agents are aware of each call's purpose, and can answer each incoming call with the appropriate greeting. Agents do not have to waste time trying to determine the purpose of calls.

Vector Routing Tables (G3V4 and later releases)

A *goto* vector step can test digits collected as part of Call Prompting, or ANI (Automatic Number identification) against numbers in a Vector Routing Table. Vector Routing Tables contain a list of numbers that can be used to test a *goto...if digits* or a *goto...if ani* command. The values can be tested to see if they are either in or not-in the specified table. Tables can have up to 100 entries.

Considerations

Call Vectoring is a valuable feature that provides a highly flexible way of processing incoming calls to the switch. With Call Vectoring, one can define a separate set of call processing steps for different types of incoming calls. Maximum benefits are realized when Call Vectoring is combined with Call Prompting and ACD.

A maximum of 1,000 vector steps are processed per call.

A maximum of 16 adjunct route steps are supported.

A call can be queued to multiple splits/skills and/or hunt groups at the same time to minimize the waiting time. The maximum number of queues that a call can be queued at is three.

A queue size must be assigned for vector-controlled splits/skills (on the Hunt Group form). If vector processing attempts to queue a call to a split, and if there are no queue slots available, the step fails, and vector processing continues at the next step.

It is recommended that you not change a vector while it is processing calls because the calls already in the vector might experience problems. Instead, add a new vector and change the VDN to go to the new vector.

Agents should not be used for hunt group calls and ACD split calls simultaneously. Otherwise, all of the calls from one split (either ACD or hunt group) are answered first. For example, if the ACD calls are answered first, none of the hunt group calls are answered until all of the ACD calls are answered.

The oldest call waiting termination is only supported for agents who are servicing ACD calls only.

Call Vectoring operations in DEFINITY Generic 3 systems are different from those in DEFINITY Generic 2 systems. For Call Vectoring information related to the DEFINITY Communications System Generic 2, see *DEFINITY Communications System Generic 2 and System 85 Feature Descriptions*, 555-104-301. The *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide*, 555-230-520, also discusses differences between G2 and G3 Call Vectoring.

Interactions

The following features interact with the Call Vectoring feature.

- Abbreviated Dialing

  A VDN extension can be used in a personal, group, system, or enhanced number list.

- Administered Connections

  The PSC administration screen blocks entry of a VDN.

- Answer Supervision

  Answer Supervision is only returned once during the life of a call. With respect to Call Vectoring commands, answer supervision is returned in response to an announcement, waiting with music, and disconnect. In addition, if a call is answered, answer supervision is sent if it has not previously been sent.

- AP Demand Print

  A VDN cannot be used as an argument to the feature access code for AP Demand Print.

- Attendant Control of Trunk Group Access

  If a *route-to* step in a vector dials a controlled trunk group, vector processing continues at the next step.

- Attendant DXS with Busy Lamp Field

  A DXS lamp for a VDN is always off. A DXS button can be used to place a call to a VDN.

- Attendant Display

  A button may be assigned to the attendant to display an internal caller's COR. The restriction identifiers are: orig, otwd, toll, code, and none. This operation is available for calls originated by a *route-to* step in a vector. The COR of the originator is displayed.

- Attendant Recall

  Attendant Recall to a VDN is blocked.

- AUDIX Interface

  A *route-to* step in a vector may call the AUDIX extension. If a voice port can be seized to that adjunct, vector processing is terminated. The system sends a message to AUDIX requesting retrieval of messages for the originating extension (not the VDN).

  AUDIX may also be accessed by the *queue-to main split* and *check-backup split* commands. Also, the messaging step may use an AUDIX hunt group in its operation.

- Authorization Codes

  Authorization codes are disabled with respect to routing via VDNs. In other words, if authorization codes are enabled, a *route-to* command in a prompting vector accesses AAR or ARS and the VDN's FRL does not have the permission to utilize the chosen routing preference, then no authorization code is prompted for and the *route-to* command fails.

- AAR/ARS

  Any *route-to* command in a vector can dial an AAR/ARS FAC followed by other digits. Specifying only the FAC in a *route-to number* command is not allowed. The *route-to* step is skipped.

- Automatic Callback

  Automatic Callback cannot be used for calls placed to a VDN.

- Automatic Call Distribution

  Splits/Skills which are not translated as vector-controlled may use all of the existing ACD features. Splits/Skills which are translated as vector-controlled lose certain ACD properties.

- Automatic Incoming Call Display

  The information displayed for the current call is replaced by the identity of the incoming call. A *route-to number* step in a vector can initiate a call to an extension. The name of the originator (or the name of the trunk group) and the name of the VDN are displayed for that extension.

- Automatic Message Waiting Notification

  The Message button can specify a VDN. The associated lamp is lit when messages are left for that VDN.

- Automatic Wakeup

  A wakeup call cannot be programmed for a VDN.

- Bridged Call Appearance

  VDN extensions cannot be assigned to bridged appearance buttons. A *route-to* command to an extension with bridged appearances updates bridged appearance button lamps.

- Busy Verification of Terminals and Trunks

  Busy verification of VDNs is denied and intercept tone is returned.

- Call Coverage

  A VDN may be administered as the last point in a coverage path.

  Vectors and VDNs do not have coverage paths associated with them.

- Call Forwarding

  Calls can be forwarded to a VDN. Calls placed by a *route-to* command to an extension that has call forwarding activated are forwarded.

  An attendant or voice terminal with console permission can activate/deactivate call forwarding for a particular extension number, TEG, or hunt group. However, activation/deactivation of call forwarding for a VDN is blocked.

  If Call Vectoring is enabled, activation/deactivation of call forwarding for a vector-controlled hunt group is blocked. The user receives intercept tone.

- Call Detail Recording (CDR) Account Code Dialing

  If a *route-to number* command in a vector specifies an CDR account code, vector processing continues at the next step.

- Call Detail Recording (CDR)

  The Feature Related System Parameters form can be administered so that the VDN extension is used in place of the Hunt Group or Agent extension. If administered to do so, this overrides the ‘‘Call to Hunt Group - Record’’ option of CDR for Call Vectoring calls.

  If a vector interacts with an extension or group that has Call Forwarding All Calls active, normal Call Forwarding/CDR interactions apply.

  For incoming calls to a VDN, the duration of the call is recorded from the time answer supervision is returned.

  — If answer supervision is returned by the vector (via an *announcement, collect digits, disconnect*, or *wait-time* command), and the call never goes to another extension, then the VDN extension is recorded as the called number in the CDR record.

  — If the call terminates to a hunt group, then the VDN, hunt group, or agent extension is recorded as the called number as per the administration described above.

— If the call terminates to a trunk, then the following two CDR records are generated:

1. An incoming record with the VDN as the called number and the duration from the time answer supervision was provided to the incoming trunk.
2. An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.

Outgoing vector calls generate ordinary outgoing CDR records with the originating extension as the calling number.

No Ineffective Call Attempt records are generated for Call Vectoring *route-to* commands that are unsuccessful.

- Call Park

  Calls cannot be parked on a VDN. If the call park access code is dialed and a VDN is entered, the user receives intercept tone. If the call park answer back access code is dialed and a VDN is entered, the user receives intercept tone.

- CMS

  See the section in this feature description entitled ‘‘Call Vectoring and the CMS’’ for CMS interactions.

- Call Pickup

  A VDN cannot be administered as a member of a pickup group. However, a vector call that routes to an extension can be picked up, if that extension is part of a pickup group.

- Call Prompting

  Call Prompting is administered through Call Vectoring administration. If only Call Vectoring is enabled, vectors can be administered using only "Call Vectoring" commands. If only Call Prompting is enabled, vectors can be administered using only "Call Prompting" commands, described elsewhere in this chapter. When both features are enabled, all the commands associated with vectoring and prompting are available.

  Enabling both vectoring and prompting together provides the capability to prompt the caller to enter pertinent data while the caller is waiting in queue at an ACD split. For example, prompting and vectoring can be used together to enhance the message collection capability to provide a caller who is waiting in queue at an ACD split with the choice of remaining in queue or leaving a message for that split. In addition to queuing, vectoring provides the capability to change calling party feedback. In addition to silence, which both prompting and vectoring supply, vectoring can provide ringback or music feedback to the calling party.

  The implications of having both features enabled are:

— When a Call Prompting command (such as a successful *route-to* command) terminates vector processing, the call must be dequeued and dropped from all queues that it is currently residing in as part of the termination process.

— If a call is waiting in an announcement queue, waiting to be connected to an announcement or an announcement queue (announcement retry), or is currently connected to an announcement, and the call is dequeued from a split's queue and terminates to an agent's voice terminal extension, the announcement is disconnected and ringback is connected to the call.

— If a *collect digits* command is being processed for a call and the call is dequeued from a split's queue and terminates to an agent's voice terminal extension, the *collect digits* command is terminated and ringback is connected to the call.

- Call Waiting Termination

  A **route-to number** command in a vector can dial a single-line voice terminal. If the extension is busy and has call waiting termination administered, the **route-to with cov n** operation is considered unsuccessful and vector processing continues at the next step. **Route-to with cov y** is successful (call will wait) and vector processing terminates.

- CAS

  A *route-to number* command in a vector can dial CAS. If a release link trunk can be seized, the *route-to number* operation is considered successful and vector processing is terminated.

- COR

  Each VDN in the system has a COR associated with it. This VDN COR is used to determine the calling permissions/restrictions, the AAR/ARS PGN, and the priority queuing associated with a given vector.

- Code Calling Access

  A VDN cannot be used as the argument to the code calling access feature access code.

  If a *route-to number* command in a vector specifies the code calling feature access code, vector processing continues at the next step.

- Conference

  A call to a VDN can be included as a party in a conference call only after vector processing terminates for that call (for example, after a successful *route-to* command).

- Coverage Callback

  A vector call does not follow any coverage paths; therefore, coverage callback is not available.

- DCS Attendant Control of Trunk Group Access

  This allows an attendant at any node in the DCS to exercise control over an outgoing trunk group at a different node in the cluster. If a *route-to number* step in a vector dials a controlled trunk group, vector processing continues at the next step.

- DCS Attendant Display

  Calls to/from a system in a DCS environment have calling/called party identification transparency. The name of the originator is displayed on the called terminal's alphanumeric display. The name of the party to whom the call is directed is displayed on the originator's display.

  The same display operations occur even if the originator, VDNs, and terminator are on different nodes in a DCS network.

- DCS Automatic Callback

  Automatic Callback calls do not apply for VDNs.

- DCS Call Forwarding

  Calls can be forwarded to a VDN anywhere in the DCS network. An attendant cannot activate/deactivate call forwarding for a VDN.

- DCS LWC

  LWC messages cannot be generated for a VDN.

- DCS Voice Terminal Display

  This feature allows calling and called name information (plus miscellaneous identifiers) to be sent from a terminal on one node to a terminal on another node. The name of the originator is displayed on the called terminal's alphanumeric display. The name of the party to whom the call is directed is displayed on the originator's display.

  The same display operations occur even if the originator, VDNs, and terminator are on different nodes in a DCS network.

- DID

  DID trunks can dial a VDN and be subject to the treatment of its associated vector.

- DOD

  DOD can be provided via a *route-to* command within a vector. The COR of the VDN is used to determine calling party permissions/restrictions.

- Do Not Disturb

  Do Not Disturb cannot be activated for a VDN.

- Emergency Access to the Attendant

  When night service is in effect, emergency calls to the attendant route to the night destination. The night destination can be a VDN.

The extension number where emergency queue overflow redirects can be a VDN.

- Facility Busy Indication

  The facility busy lamp indication for a VDN is always off. A facility busy button may be used to call a VDN.

- FRL

  If a *route-to* command dials an external number via AAR/ARS, the FRL associated with the VDN COR is used to determine the accessibility of a routing preference in an AAR/ARS pattern.

- Facility Test Calls

  Provides a voice terminal user with the capability of making test calls to access specific trunks, touch-tone receivers, time slots, and system tones. The test call is used to make sure the facility is operating properly. A local voice terminal user can make a test call by dialing an access code. An Initialization and Administration System (INADS) terminal user can also make test calls.

  If a *route-to number* command in a vector specifies a Facility Test Call, vector processing continues at the next step.

- Forced Entry of Account Codes

  Requires users to dial an account code when making certain types of outgoing calls. To maximize system security, it is recommended that the Forced Entry of Account Codes feature be enabled and administered on the System. The Forced Entry of Account Codes option must be enabled on the System-Parameters Customer-Options form before associated forms and fields on forms can be administered.

  If a COR requiring entry of account codes is assigned to a VDN, the *route-to number* commands executed by the associated vector are unsuccessful and vector processing continues at the next step.

- Individual Attendant Access

  Each attendant console can be assigned an individual extension number. That extension number can be used as the argument to a *route-to number* command in a vector.

  Each attendant has a queue that allows two incoming calls to wait. This individual attendant queue has priority over all other attendant-seeking calls. A call established by a *route-to number* command in a vector can wait in this queue and is removed from vector processing.

- ISDN-PRI

  A VDN may be accessed by an ISDN-PRI trunk. ISDN-PRI calls may be disconnected without providing answer supervision.

  A vector may initiate calls over ISDN-PRI facilities via a *route-to number* command.

- Integrated Directory

  VDN names and extensions are not available in the Integrated Directory feature.

- Intercept Treatment

  A VDN cannot be used for Intercept Treatment.

- Intercom — Automatic

  A VDN cannot be included in an intercom group.

- Intercom — Dial

  A VDN cannot be included in an intercom group.

- Inter-PBX Attendant Calls

  A *route-to number* command in a vector can dial the Inter-PBX Attendant. If the call attempts to access a controlled trunk group, vector processing continues at the next step.

- Intraflow and Interflow

  The functionality of intraflow and interflow may be obtained using the *check-backup split* and *goto* Call Vectoring commands.

  Calls may intraflow from an ACD split which is not vector-controlled into one that is vector-controlled.

- Last Number Dialed

  After a voice terminal user with a Last Number Dialed button dials a VDN extension, the VDN extension is stored as the Last Number Dialed and may be redialed by pressing the button.

- LWC

  LWC messages cannot be stored, canceled, or retrieved for a VDN.

- Manual Message Waiting

  Message Waiting buttons may point to a VDN.

- Manual Signaling

  A manual signaling button can point to a VDN. However, activation of a Manual Signaling button which points to a VDN is ignored.

- Multiple Listed Directory Numbers

  The incoming destination for each CO trunk group and each FX trunk group used for LDNs can be a VDN. The DID LDN night extension can be a VDN.

- Music-on-Hold Access

  The system-wide Music-on-Hold Access feature must be administered in order for music to be heard when a *wait-time* command with system music is executed. If it is not administered, the calling party receives silence.

- Network Access — Private

  A *route-to number* command in a vector can access private networks.

- Network Access — Public

  A *route-to number* command in a vector can access public networks.

- Night Service

  A VDN can be administered as a night service destination.

  Route-to commands that route to destinations with night service activated redirect to the night service destinations.

- Priority Calling

  A VDN cannot be used with the priority calling access code. Intercept tone is supplied to the user. If a *route-to number* in a vector specifies the priority calling access code, vector processing continues at the next step.

- Property Management System Interface

  VDNs cannot be used with the following features and functions: Message Waiting Notification, Check-In, Check-Out, Room Status, and Automatic Wakeup.

- Recorded Announcement

  The first announcement extension, second announcement extension, first announcement delay, second announcement delay, and recurring second announcement do not exist for a hunt group translated as vector-controlled. Recorded announcements may be accessed via the *announcement, collect digits, disconnect*, and *route-to number* vector commands.

- Remote Access

  A remote access user can access a VDN.

- Ringback Queuing

  External call attempts made via *route-to* commands with coverage no are not queue via Ringback Queuing when all trunks are busy. External call attempts made via *route-to* commands with coverage yes are.

- Send All Calls

  If the destination of a *route-to with coverage no* command has the Send All Calls feature active, the feature activation is ignored. If there is an idle appearance, the call is terminated and vector processing terminates. If not, vector processing continues at the next step.

  If the Send All Calls button is pressed after a vector call is terminated, button activation is denied.

- Subnet Trunking

  Subnet trunking applies to AAR/ARS calls placed via a *route-to number* command.

- System Measurements

  Hunt group measurements need different interpretation when Call Vectoring is used. For example, If a call queues to splits/skills one, two, and three, but the caller abandons the call, the measurements show only split 1 as having an abandoned call (on BCMS and R3 CMS).

- TEG

  A VDN cannot be administered as a TEG member.

  A *route-to number* command in a vector can specify a TEG extension.

- Timed Reminder

  The attendant Timed Reminder is not available for calls placed, transferred, or extended to a VDN. Vectoring causes all other timers to be ignored.

- Time of Day Routing

  Since a *route-to number* command in a vector can specify the AAR or ARS access codes, the TOD routing algorithm can be used to route the call.

- Transfer

  Stable trunk or internal calls (such as those that are currently in a talking state) can be transferred to a VDN. Calls in which vector processing is active are treated the same as ringing calls for transfer purposes.

- TCM

  A TCM is sent when a *route-to* command dials a seven-digit ETN or 10-digit DDD number via AAR/ARS. This TCM is the FRL associated with the VDN COR.

- Trunk Identification by Attendant

  This feature is available for incoming or outgoing calls routed through a VDN.

- UDP

  A *route-to* command can call a UDP extension.

- Voice Message Retrieval

  If a *route-to number* command in a vector specifies the Voice Message Retrieval access code, vector processing continues at the next step.

Administration

Call Vectoring is administered on a per-system basis by the System Manager. For a detailed description of the required administration see *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653. The following items require administration:

- ‘Call Vector" form (one form per vector)
- ‘Vector Directory Number" form (one form per VDN)
- ‘Hunt Group" form
  - Whether or not each hunt group is vector-controlled.
- ‘System-Parameters Customer-Options" form

  The following list describes the features that are available with each selection on the System-Parameters Customer-Options form. The system administer can view this form. However, it can only be changed by AT&T personnel.

  - Basic Call Vectoring, Call Prompting, or both must be enabled.

    Basic Call Vectoring allows the vector to place calls in a queue for a split and to check the size of a queue for a particular split. The latter facility is useful for directing a call to the split where it receives the quickest response.

    Vectoring with prompting allows the vector to collect digits from a caller and use these digits for routing the call. This facility can be used to program an automatic answering service to direct the caller to the required department.

  - Vectoring (G3V4 Enhanced)

    Allows for the use of enhanced comparators, wildcards, Vector Routing Tables, and alternative audio/music sources for the *wait-time* step.

  - Vectoring (G3V4 Advanced Routing)

    Allows for the use of the Average Speed of Answer, VDN Calls, and Expected Wait Time conditionals.

  - Vectoring (ANI/II-Digits)

    Allows for the use of ANI and II-Digits for conditional routing.

    For a complete description of the commands and features available with each customer option, see the *DEFINITY Communications System Generic 3 Call Vector/Expert Agent Selection (EAS) Guide,* 555-230-520.

Administration software *does not* allow a VDN extension to be entered as data in the listed fields of the following screen forms:

- Recorded Announcements
- Call Coverage Answer Group
  - Group Member Assignments

- Call Coverage Paths
  - G3v1—Coverage point assignments
  - G3v2 and later—Coverage Point Assignments, other than the last coverage point
- Console Parameters
  - CAS Back-up Extension
- Feature-Related System Parameters
  - ACA Long Holding Time Originating Extension
  - ACA Short Holding Time Originating Extension
  - Extensions With System wide Retrieval Permission
  - Controlled Outward Restriction Intercept Treatment
  - Controlled Termination Restriction (Do Not Disturb)
  - Controlled Station-to-Station Restriction
- Hospitality-Related System Parameters
  - Extension of PMS Log Printer
  - Extension of Journal/Schedule Printer
  - Extension of PMS
  - Extension to Receive Failed Wakeup LWC Messages
- Hunt Groups
  - Supervisor Extension
  - Member Extensions
- Intercom Group
  - Member Extensions
- Listed Directory Numbers
  - LDN Extensions
- Loudspeaker Paging and Code Calling Access
  - Extension Numbers Assigned to Codes
- Pickup Groups
  - Member Extensions
- Remote Access
  - Remote Access Extension
- Terminating Extension Group
  - Member Extensions

Administration software *does* allow a VDN extension to be entered as data in specific fields of the following screen forms:

- Abbreviated Dialing Lists
- Call Coverage Paths (G3v2 and later allow it as the last coverage point only)
- Hunt Groups
  - Night Destination
- Listed Directory Numbers
  - Night Destination
- Trunk Groups
  - Night Destination
  - Incoming Destination

Administration software *does not* allow a VDN extension to be entered as auxiliary data for the following buttons:

- Bridged Appearance (brdg-app)
- Data Call Setup (data-ext)

Administration software *does* allow a VDN extension to be entered as auxiliary data for the following buttons:

- Remote Message Waiting Indicator (aut-msg-wt)
- Facility Busy Indication (busy-ind)
- Manual Message Waiting (man-msg-wt)
- Manual Signaling (signal)

For more detailed information on the administration of Call Vectoring, see *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

Hardware and Software Requirements

Announcement hardware is required (see information on the Call Prompting feature). Call Vectoring software is required.

Call Waiting Termination

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides for calls to busy single-line voice terminals to wait and sends a distinctive call waiting tone to the called party.

Generally, the called party hears one quick burst of tone when a call from another voice terminal user is waiting, two quick bursts of tone when an attendant-handled or an outside call is waiting, or three quick bursts of tone when a Priority Call is waiting. The called party hangs up on the current call and immediately receives ringing from the waiting call. For G3i-Global, G3V2, and later releases, the defaults for the number of ring bursts may be changed by the system administrator.

The call in progress at the voice terminal can be placed on hold in order to answer the waiting call. After answering the waiting call, the voice terminal user can return to the held call or toggle back and forth between the two calls. The single-line voice terminal user can only be connected to one call at a time.

The calling party hears special audible ringback tone if the call is allowed to wait. If Call Waiting is denied, the calling party hears busy tone. Only one call can wait at a time.

NOTE:
Special ring types are not supported over DID facilities.

The burst(s) of tone heard by the called voice terminal user is not heard by other parties on the call.

An internal caller can activate LWC or Automatic Callback after Call Waiting has been activated.

A Priority Call and an attendant-handled call can wait for the voice terminal to become idle even if the Call Waiting Termination feature is not assigned.

Calls to a DDC or UCD group voice terminal cannot wait. However, such calls can enter the group queue (if provided) unless the queue is full.

Considerations

With Call Waiting Termination, the party who calls a busy single-line voice terminal does not have to hang up and try the call again later. Instead, the call waits at the called voice terminal until the called party hangs up on the current call.

Call Waiting Termination applies only to busy single-line voice terminals. Calls to multi-appearance voice terminals are routed to an idle call appearance and do not wait.

An analog voice terminal user must place the active call on ‘‘soft’’ hold (see Hold feature) and dial the Answer Hold-Unhold feature access code to answer the waiting call. The soft held call at that time becomes a ‘‘hard’’ held call.

If an analog single line set has the Call Wait feature enabled and has created a conference call, the Call Wait feature is denied. For example, when caller A on an analog set is talking to caller B, then flashes and is talking to caller C, and flashes to conference B and C, then, if caller D calls caller A, the Call Wait feature is denied.

Interactions

Call Waiting is denied when the following features are activated at the single-line voice terminal:

- Automatic Callback (to or from the voice terminal)
- Data Privacy
- Data Restriction
- Another Call Waiting Call

A Call Waiting call cannot be picked up by a Call Pickup group member.

Administration

Call Waiting Termination is assigned on a per-voice terminal basis by the System Manager. The only administration required is the assignment of Call Waiting Termination to the desired voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Calling Line Identification (CLI) Prefix

Feature Availability

The Calling Line Identification (CLI) Prefix feature is available with G3V4 Release 3 and later releases.

Description

The Calling Line Identification (CLI) Prefix feature allows DEFINITY to modify calling party number (CPN) information for ISDN/PRI calls that come in to the switch on a given trunk group. These calls may terminate locally, or may be tandemed out to other private switches or central offices. This feature only applies to incoming calls, and is administered for and applied to *all* calls on a trunk group. It does not apply to calls originated locally.

Based on the values in the Calling Number - Delete, Insert and Numbering Format fields, DEFINITY will delete up to 15 of the most significant digits from the incoming CPN information. If there is a digit string in the Insert field, DEFINITY appends these numbers to the CPN after deleting the specified number of digits. Only the first 15 digits are passed on when the call is routed.

This feature is a requirement of the Unisource Network, but anyone may use the CLI Prefix for call accounting or redirection purposes. By using this feature in conjunction with Call Vectoring, it is possible to accomplish call routing required by Unisource, as well as Unisource-specific error handling. For example, if ANI information is required, a vector step can check whether it is present, and if the ANI is exactly the same as the inserted digits, this call may be rejected.

Considerations

For switch- or adjunct-based features that require database lookup of CPN/ANI information, you will need to change the appropriate databases to reflect the CLI Prefix modifications. For example, if you use the CLI Prefix feature in conjunction with Call Vectoring, you may need to update the Vector Routing Table.

Because Call Vectoring can determine whether an incoming call has Automatic Number Identification or not, if you use the CLI Prefix to add digits to incoming CPNs, you may need to program ANI detection differently.

If you administer the feature to delete incoming digits, this may lead to imprecise or ambiguous CPN/ANI lookups for features such as Malicious Call Trace.

Interactions

Because this feature affects the calling party information for all calls coming in on a given trunk group, other features that use the CPN will receive modified calling number information. However, the functionality of these features will not change. Terminal displays for calls that have been modified with the CLI Prefix will reflect the modifications.

The Message Sequence tracer continues to record CPN information as it came in on the ISDN link, and does not record the modified number.

The following features receive CPN information that may be changed as a result of the CLI prefix:

- Attendant Display
- Automatic Incoming Call Display
- Coverage Incoming Call Identification
- Call Detail Recording
- Calling Number Delivery to CMS (SID/ANI)
- Calling Number Display
- Call Pickup
- Call Vectoring:
  - ANI/II Digits Routing
  - Converse Command (VRU Integration) - if digits were added to the CPN, a slightly longer time may be required when the CPN is passed inband to a Voice Response Unit.
  - Error Trace Log (Vector Events)
- Commonwealth of Independent States - Outbound Call MFR ANI Transmission
- DS1 Interface
- DCS Alphanumeric Display for Terminals
- ISDN/BRI
- ISDN Call-by-Call Service Selection
- ISDN QSIG Name and Number
- ISDN QSIG Call Diversion
- ISDN QSIG Call Transfer
- ISDN Gateway - 1-Way BRI Interface
- Integrated R2-MFC Signaling (DID/CO)
- Inter-PBX Attendant Calls (IA Service)

- Malicious Call Trace - If the incoming CPN is affected by this feature, accurate lookup of the number may not be possible.
- Network Access - Public
- Network Access - Private Premises Bases
- Network Access - Private Public Net Based (SDN)
- PC/PBX Connection
- PC/ISDN Connection
- Transfer (single-line and multi-appearance)
- Trunk-to-Trunk Transfer
- Voice Terminal Display - Calling Number Display (SID/ANI/Extn ID)

Administration

This feature can be implemented for incoming calls on a per trunk group (incoming or two-way) basis. The administration required includes the assignment of digits to delete and/or append to CPN information, and the numbering format to use. Send Calling Number must be set to “y." All of these fields are found on the ISDN/PRI Trunk Group form.

NOTE:
If you have an ISDN trunk group with incoming calls that are mixed between those that need this feature and those that don’t, partition the calls into different trunk groups so that the CLI Prefix feature can be administered only for those calls that need it.

Existing G3MA Bulkload scripts may need to be modified to accommodate the CLI Prefix Administration fields.

Hardware Requirements

None.

Call-By-Call Service Selection

Feature Availability

This optional feature is available with G3vs/G3s PBP, Generic 3i, and Generic 3r, and is not available with G3vs/G3s ABP.

Description

Call-By-Call Service Selection allows a single ISDN-PRI trunk group to carry calls to many services or facilities (such as a SDN, MEGACOM telecommunications service, MEGACOM 800 service, and so on) and/or to carry calls using different Inter-exchange Carriers.

NOTE:
The Call-by-Call Service is only applicable for Country Protocol option 1 (U.S.).

Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by AAR, ARS, and GRS. The service or facility used on an outgoing Call-By-Call Service Selection call is determined by information assigned in the AAR/ARS/GRS routing patterns.

As an example of how Call-By-Call Service Selection works, see the following figure. Without Call-By-Call Service Selection, each trunk group must be dedicated to a specific service or facility. Call-By-Call Service Selection eliminates this requirement by allowing a variety of services to use a single trunk group. These services are specified on a call-by-call basis. Trunking efficiency is immediately obtained with Call-By-Call Service Selection by distributing traffic over the total number of available trunks.

Services Used With Call-By-Call Service Selection

The services used on incoming and outgoing Call-By-Call Service Selection calls are assigned after an ISDN-PRI trunk group is assigned a service type of Call-By-Call Service Selection. A Call-By-Call Service Selection trunk group can be administered to carry calls to many services. The available services in the US are as follows:

- ACCUNET Digital Service — AT&T's digital network services for various high-volume, high-speed data transmission requirements.
- INWATS — Provides OUTWATS-like pricing and service for incoming calls.
- AT&T Long Distance Service — A shared use, two-way, premises-to-premises service that uses the public-switched network to transmit and receive voice, data, and graphics communications.

- OUTWATS Band — WATS is a voice-grade service providing both voice and low-speed data transmission capabilities from the user’s location to defined service areas commonly referred to as bands. Currently, the widest band is five.
- MEGACOM Service — Provides an AT&T service that provides unbanded long distance services using special access (PBX to 4ESS switch) from an AT&T node.
- MEGACOM 800 Service — Provides an AT&T service that provides unbanded 800 service using special egress (4ESS switch to PBX) from an AT&T node.
- Network Operator — Provides access to the network operator.
- SDN — An AT&T offering that provides a virtual private network using the public-switched network. SDN can carry voice and data between customer locations as well as off-net locations.
- Presubscribed Common Carrier Operator — Provides access to the presubscribed common carrier operator.
- Maximum Banded WATS — A WATS-like offering for which a user’s calls are billed at the highest WATS band subscribed to by the user.
- International 800 — Allows a subscriber to receive international calls without a charge to the call originating party. The subscriber of the service is charged for the calls.
- MULTIQUEST. Telecommunications Service — Service between callers at switched-access locations and service providers directly connected to the AT&T switched network. Callers access the service providers by dialing a 700 number.
- Other User-Defined Services — New service types can be assigned as they are developed and defined.
- Call-By-Call services by Bellcore NI-2 PRI are Foreign Exchanges (FX), Tie, OUTWATS, and INWATS. The DEFINITY switch does not support the NI-2 FX, Tie, and Bonded OUTWATS services.

  NOTE:
  When the DEFINITY switch is connected to 5ESS, all of the services supported under the AT&T and NI-2 country options are available.

A call-by-call service selection example is shown in Figure 3-2.

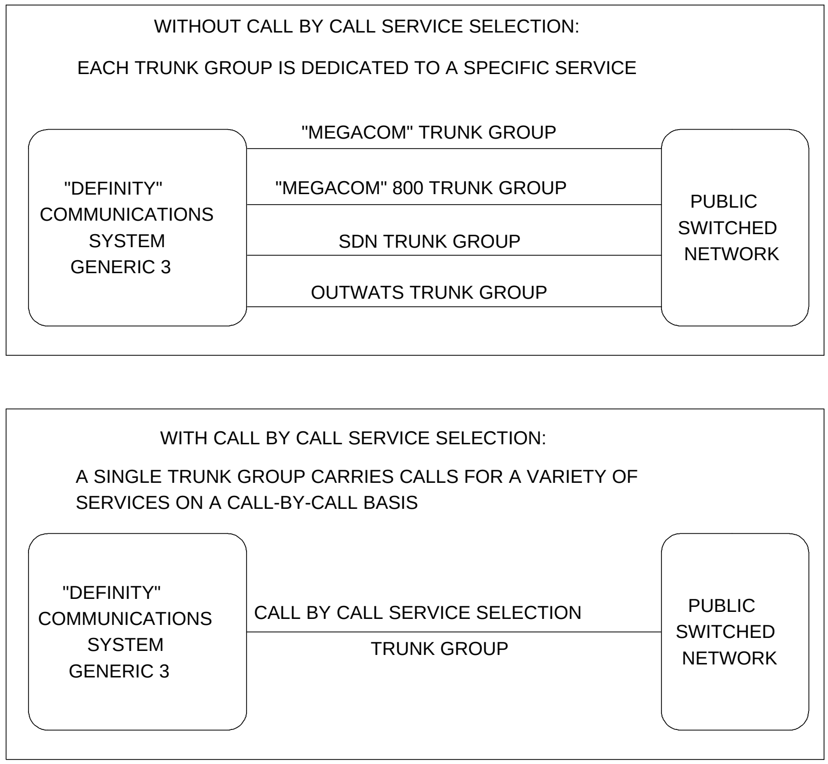


Figure 3-2. Call-By-Call Service Selection Example

ISDN-PRI Messages and Information Elements Used for Call-By-Call Service Selection

Although the technical details of ISDN-PRI messages and information elements are not critical to implementing the ISDN-PRI application, the following details may aid in the understanding of some readers and are therefore included in this description.

Call-By-Call Service Selection allows the system to specify one of the preceding service types on a call-by-call basis. This is done via a SETUP message that indicates the intent of the originating system to initiate a call using the specified service or facility. The SETUP message contains units called information elements which specify call-related information. The information elements used with Call-By-Call Service Selection are as follows:

- Network Specific Facility — Indicates which facilities or services are to be used to complete the call.

The system also checks all incoming ISDN-PRI calls for the presence of a Network Specific Facility information element. If this information element is present, the system makes sure that the requested service is compatible with the administration of the trunk. If the requested service is not compatible with administration, the switch rejects the call.

For an outgoing call on a Call-By-Call trunk group, the Network Specific Facility information element is constructed using the Service/Feature specified on the routing pattern preference selected for the call.

- Transit Network Selection — Indicates which Inter-exchange Carrier is to be used on an inter-LATA call.

  If a call requires both the Service/Feature and the Inter-exchange Carrier to be specified, the Inter-exchange Carrier information is sent in the Network Specific Facility information element rather than the Transit Network Selection information element.

Usage Allocation Plan

Optional Usage Allocation Plans (UAPs) may be assigned to provide more control over a Call-By-Call Service Selection trunk group. Up to three Usage Allocation Plans can be assigned for each Call-By-Call Service Selection trunk group. A Usage Allocation Plan allows the customer to set the following:

- A maximum number of trunk group members that each specific service can use at any given time.
- A minimum number of trunk group members that always is available for each specific service.

The sum of the allocation plan maximums may exceed the total number of trunk group members. For example, if a trunk group has 15 members and provides access to MEGACOM service, MEGACOM 800 service, and SDN, the maximum number of trunks to be used for each of these services could possibly add up to more than 15. In this case, for example, you could administer a maximum of seven MEGACOM service calls, six MEGACOM 800 service calls, and eight SDN calls. This ensures that all trunk group members are not dominated by a specific service, yet allows for periodic fluctuations in demand.

The sum of the allocation plan minimums may not exceed the total number of trunk group members. For example, if a trunk group has 10 members and provides access to MEGACOM service, MEGACOM 800 service, and SDN, the minimum number of trunks to be used for each of these services cannot add up to more than 10.

If a UAP has been defined for a Call-By-Call Service Selection trunk group, and the type of the incoming call exceeds one of the plan’s limits, the system rejects the call, even if a trunk is available. If a UAP has been defined for a Call-By-Call Service Selection trunk group, and a system user makes an outgoing call of a type that exceeds one of the plan’s limits, the user receives reorder tone unless other preferences are available.

As previously mentioned, each Call-By-Call Service Selection trunk group can have as many as three UAPs. The customer can assign either fixed or scheduled allocation plans for each Call-By-Call Service Selection trunk group, (see Screen 3-30, Screen 3-31, and Screen 3-32) which are used to administer the Usage Allocation Schedule and plans).

- Fixed

  One plan applies at all times. The minimum and maximum usages specified in this plan are in effect for the trunk group at all times.

- Scheduled

  Two or three plans can be administered to apply at different times based on the time of day and day of the week. As many as six activation times and associated plans can be assigned for each day of the week. At the specified activation time, the associated plan goes into effect for the Call-By-Call Service Selection trunk group.

```
               CBC SERVICE TYPE USAGE ALLOCATION PLAN ASSIGNMENT SCHEDULE
                                                                Page x of x

Usage Method:
        Fixed? _y       Allocation Plan Number: _1
    Scheduled? _n
       Time   #    Time   #    Time   #    Time   #    Time   #     Time   #
Sun    00:00  1    __:__  _    __:__  _    __:__  _    __:__  _     __:__  _
Mon    00:00  1    __:__  _    __:__  _    __:__  _    __:__  _     __:__  _
Tue    00:00  1    __:__  _    __:__  _    __:__  _    __:__  _     __:__  _
Wed    00:00  1    __:__  _    __:__  _    __:__  _    __:__  _     __:__  _
Thu    00:00  1    __:__  _    __:__  _    __:__  _    __:__  _     __:__  _
Fri    00:00  1    __:__  _    __:__  _    __:__  _    __:__  _     __:__  _
Sat    00:00  1    __:__  _    __:__  _    __:__  _    __:__  _     __:__  _
```

Screen 3-31. CBC Service Type Usage Allocation Plan Assignment Schedule

```
                     CBC SERVICE TYPE USAGE ALLOCATION
                                                               Page 3 of 8
Trunk Allocation Plan 1       Trunk Allocation Plan 2    Trunk Allocation Plan 3
  Service/     Min#  Max#       Service/     Min#  Max#     Service/     Min#  Max#
  Feature      Chan  Chan       Feature      Chan  Chan     Feature      Chan  Chan
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
_____________   __    __     _____________   __    __     _____________   __    __
```

Screen 3-32. CBC Service Type Usage Allocation

System administration allows the customer to have anything from a simple fixed usage allocation plan to a very flexible plan with many scheduling options. The customer can even start out with no allocation plan and build the plan as the need arises. This allows the customer to respond to periodic fluctuations in the environment in a more timely manner. The customer does not have to involve the network to fine-tune the trunk group administration. To ensure that administration complexity is kept to a minimum, the following steps should be followed when assigning UAPs.

1. Assign a Usage Allocation Plan for a Call-By-Call Service Selection trunk group.
2. If scheduling is desired, add one or two more Usage Allocation Plans for that trunk group.
3. Administer the scheduling information for the trunk group's UAPs.

Incoming Call Handling Treatment

Call-By-Call Service Selection provides special Incoming Call Handling Treatment for ISDN-PRI trunk groups. An incoming call on an ISDN-PRI trunk group is handled according to a treatment table that is administered for that trunk group. An example of the screen that contains this table follows.

```
TRUNK FEATURES                                                Page 2 of x
   Service/    Called      Called        Del       Insert      CPN/BN     Night
   Feature      Len        Number                                          Serv
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
 _____________  ___  _______________  ___  ______________ __________  _______
```

Screen 3-33. Trunk Features

As shown above, a variety of specifications can be administered in the table for the treatment of incoming calls. The table allows for as many as 12 different combinations of call treatments. Seven columns are provided for seven different specifications. These specifications are as follows:

- Selection Criteria
  - **Service/Feature** — Specifies the ISDN-PRI Services/Features to which the specified call treatment may apply. These services/features are those discussed in the "Services Used With Call-By-Call Service Selection" section of this feature description. The identifier "other" can be used for all Services/Features not explicitly specified.
  - **Called Len** — Specifies the number of digits contained in the Called Party Number (the digits received for the incoming call). The number of digits contained in the Information Element must exactly match the number of digits specified in this field. Allowable entries are 0 through 16 or blank. A blank entry indicates a wild card, meaning that a called number of any length matches.
  - **Called Number** — Specifies the leading digits contained in the Called Party Number. For this row to be chosen for a call, the data is this field must exactly match the leading digits in the IE. A blank entry indicates that the incoming digits are not significant for this entry. Allowable entries are 1 through 16 digits or blank.

- Action Taken Based on Selection Criteria
  - **Del** — Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type may be administered to be routed to a single destination by deleting all incoming digits and then administering the “Insert” field with the desired extension.
  - **Insert** — Specifies the digits to be prepended to the front of the Called Party Number. The new number is used to route the call. Allowable entries are up to 16 digits or blank.
  - **CPN/BN** — Specifies your preference of a Calling Party Number (CPN) or Billing Number (BN) request for this type of call. A blank or ‘‘none’’ indicates that the switch is not requested either CPN or BN for any incoming calls of this type. Allowable entries are BN only, prefer BN but accept CPN, CPN only, and prefer CPN but accept BN.
  - **Night Serv** — Specifies a night service extension per Service/Feature. An entry other than blank overrides the night destination entry on Page 1 of the form. Allowable entries are an assigned extension, attendant, or blank.

    NOTE:
    Bellcore NI-2 Call-By-Call service does not support the delivery of CPNI B-N on a per call basis.

The treatment for an incoming call is selected based on the “Service/Feature, Called Len”, and “Called Number” fields in the table. When the attributes of an incoming call match these specifications, the call is treated according to the corresponding Del, Insert, CPN/BN, and Night Service specifications. If an incoming call matches more than one set of specifications, the most restrictive case is selected. The following table lists the possible cases in order of most restrictive to least restrictive:

Table 3-47 lists the most/least restrictive specifications.

Table 3-47. Call-by-Call Service Selection Most/Least Restrictive Specifications

	Service/ Feature	Called Len	Called Number
Most Restrictive	Specified	Specified	X Number Of Leading Digits Specified
	Specified	Specified	Y Number Of Leading Digits Specified, Where Y < X
	Specified	Specified	Not Specified
	Specified	Not Specified	Not Specified
	Specified as ‘‘other’’	Specified	X Number Of Leading Digits Specified
	Specified as ‘‘other’’	Specified	Y Number Of Leading Digits Specified, Where Y < X
	Specified as ‘‘other’’	Specified	Not Specified
Least Restrictive	Specified as ‘‘other’’	Not Specified	Not Specified

Considerations

Call-By-Call Service Selection provides the following benefits:

- Cost Reduction — Since many services share the same trunks, the total number of trunks can be reduced.
- Improved Service — Call-By-Call Service Selection trunks can reduce the probability of features and services being blocked.
- Simplified Networking — Network engineering is simplified because analysis of trunking needs can be done based on total traffic instead of on a per-service basis.
- The ability to respond to changes in a more timely fashion. The network does not have to be consulted because of the flexibility provided by the usage allocation plans.
- Measurement of Call-By-Call Service Selection calls.

Interactions

The following features interact with the Call-By-Call Service Selection feature.

- AAR

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by AAR.

- ARS

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by ARS.

- GRS

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by GRS.

- CDR

  On successful incoming and outgoing Call-By-Call Service Selection calls, the Network Specific Facility specified by the call's Network Specific Facility Information Element is recorded by CDR. CDR refers to this information as the INS (ISDN Network Service).

  If an outgoing Call-By-Call Service Selection call uses an Inter-exchange Carrier (IXC) other than the presubscribed common carrier, CDR records the 3- or 4-digit IXC code. CDR may not record the IXC code properly if the dialed code format differs from the US IXC code formats.

  When a Call-By-Call Service Selection call is rejected because of a trunk group usage allocation plan, CDR records the cause as an ineffective call attempt.

- Time of Day Routing

  Any Time of Day Routing administration that affects routing preference also affects Call-By-Call Service Selection.

  The Time of Day Routing feature can be used to vary the IXC based on the time of day and day of the week.

- Traffic Measurements

  The system provides traffic measurements for each individual service administered as part of the ISDN Call-By-Call Service Selection trunk group.

Administration

Call-By-Call Service Selection is administered by the System Manager on a per trunk group basis. The following items require administration:

- ISDN-PRI Trunk Group — Must be administered with a Service Type of Call-By-Call Service Selection. The trunk group administration also includes the following:

- — Incoming Call Handling Treatment
- — Whether or not UAPs are required
- — UAPs
- — UAP Assignment Schedule
- — Group Member Assignments

- AAR/ARS Routing Patterns — Routing Patterns can be administered to include a Network Specific Facility and/or IXC.
- Network Specific Facilities Encoding — New Network Specific Facilities can be added as needed by the system technician.

Hardware and Software Requirements

A TN767 24-port DS1 circuit pack or TN464B/C/D DS1 interface (24-port T1/32-port E1) circuit pack is required for assignment of a signaling link and up to 23 or 30 ISDN-PRI Trunk Group members. For G3vs, a TN765 Processor Interface circuit pack is required in conjunction with the DS1 circuit pack, for ISDN D-Channel Links. For G3s and G3i, a TN675 Processor Interface or a TN778 Packet Control circuit pack are required for ISDN D-Channel Links.

NOTE:
The TN767 and the TN464C and later revisions of the DS1 circuit packs provide a DSX1 interface.

CallVisor Adjunct/Switch Applications Interface (ASAI)

Feature Availability

This optional feature may be purchased with any Generic 3 release except G3vs/G3s ABP and G3i-Global.

Description

Provides one or more links between the DEFINITY switch and one or more adjuncts. CallVisor ASAI improves the call handling efficiency of ACD agents and other system users by allowing an adjunct to monitor, initiate, control, and terminate calls on the switch. The CallVisor ASAI interface may be used for Inbound Call Management (ICM), Outbound Call Management (OCM), and office automation/messaging applications.

CallVisor ASAI services are provided using either Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) transport (CallVisor ASAI-BRI), or LAN Gateway Transmission Control Protocol/Internet Protocol transport (CallVisor ASAI-LAN Gateway).

Regardless of transport type, CallVisor ASAI messages and procedures are based on the ITU-T Q.932 international standard for supplementary services. A special information element, the Q.932 Facilities Information Element (FIE), carries the CallVisor ASAI requests and responses across the interface. An application program can access CallVisor ASAI services or features by supporting the ASAI protocol, by using a computer vendor, or by a third-party provider application programming interface.

The system can have up to eight (four for G3s PBP) CallVisor ASAI links.

G3V4 ASAI Enhancements

- **ASAI Accessed Internally Measured Data**

  This feature allows an ASAI adjunct application to access and retrieve Internally Measured Data used to provide VuStats information to display-equipped digital voice stations. The application accesses this data via a new Value Query message that specifies whether agent, split/skill, trunk group, or VDN information is being requested. The new message also specifies a particular observed object of the type (for example, agent or split) specified in the request.

  By providing Internally Measured Data via ASAI, applications can offer enhanced VuStats-like presentations (for example, using graphics, color, or multimedia) on a PC or can further process this data to provide enhanced statistics for better call center management. Also, VuStats-like service can be provided without requiring a digital display-equipped station.

- **Send DTMF Signals**

  This feature is an ASAI-accessed service that, when invoked by a client application, causes the DEFINITY system to send a DTMF sequence on behalf of one party on an active call. The digit sequence to be transmitted is contained in the ASAI service message.

  The DTMF tones can be heard by any endpoint connected to the sending talk path. Through such a tone sequence, an adjunct can interact with far-end applications such as automated bank tellers, automated attendants, voice mail systems, various databases, and paging services. An application could provide certain convenience features such as automated entry of passwords or service-access sequences.

- **Flexible Billing**

  This feature allows an ASAI adjunct to change the rate at which an incoming 900-type call is billed.

- **Redirect Call**

  This feature allows an ASAI adjunct to direct the switch to move an already alerting call away from an extension (at which it is alerting) to another endpoint. Prior to G3V4. such routing was possible only if switch features, rather than an ASAI-provided service, were used (for example, call forwarding or Send All Calls). With this service, an application could, for example, determine, based on call-related information or time of day, etc., whether to answer the call or re-route it to some other number.

  This service does not support removing a call from a queue. Only calls *alerting* at extensions can be manipulated by this service.

- **ASAI-Accessed Integrated Directory Database Service**

  This feature allows an ASAI adjunct to access and retrieve administered name-extension associations stored in the switch (all administered names associated with station extension numbers, trunk groups, and VDNs). The application accesses this data via a new Value Query message containing the target extension. If a valid Query Message is submitted, the switch responds with a message containing the administered name.

- **Event Report Capabilities**

  Enhancements to Event Reports allow ASAI adjuncts to receive Event Reports when specified events occur at monitored objects. In addition, in the Alerting Event Report, new cause values are provided to specify a reason for redirection.

  The following are descriptions of new/changed Event Report for G3V4. These reports are provided only if Link Version 2 is active.

- **Agent Login Event Report** — This is a new Event Report provided when an agent logs into a domain controlled ACD group or EAS skill. Notification of manual login events allows adjunct applications to maintain accurate views of current agent login/logout status.
- **Call-Originated Event Report** — This is a new Event Report provided when a call is manually or otherwise originated from a domain-monitored station. It also provides the dialed digits to an application.
- **Alerting Event Report** — This existing Alerting Report has been enhanced to provide a new set of cause code values that map one-to-one to each existing G3V4 reason for redirection that appears on display-equipped stations when a redirected call is offered.

  The new cause codes are provided in an existing Information Element of the Report. An adjunct application can use this information to provide an enhanced information display (for example, on a computer monitor) to called parties or to otherwise determine how to best handle an incoming call.

In addition to these GV4 enhancements, a new transport option, ASAI-LAN Gateway, is available. Also, CPN/BN on BX.25 links has been removed in G3V4.

See the *DEFINITY Communications System Generic 3 CallVisor ASAI Technical Reference,* 555-230-220, for more detail on these enhancements.

CallVisor ASAI Capabilities

The following capability groups are provided by CallVisor ASAI:

- Third Party Call Control Group
- Third Party Domain (**Station/ACD Split**) Control Group
- Notification Group
- Set Value Group
- Value Query Group
- Request Feature Group
- Routing Group
- Maintenance Group

The following paragraphs provide an overview of these capability groups.

NOTE:
The Third Party Domain (Station/ACD Split) Control Capability Group is a actually a subsection of the Call Control Capability Group. For the purpose of this document, the Third Party Domain (Station/ACD Split) Group is treated as a separate capability group. This section represents the DEFINITY Communications System implementation of ASAI, as described

in the *DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference*, 555-230-221, and the *DEFINITY Communications System Generic 3 CallVisor ASAI Technical Reference*, 555-230-220.

Computer vendors may not implement all of the ASAI capabilities with their respective Application Programming Interface (API). Specific computer vendors determine which aspects of these capabilities are supported within their APIs.

Third Party Call Control Capabilities Group

Third Party Call control capabilities are used by the adjunct to initiate, control, and terminate calls to which it is not a party. The following capabilities make up the Third Party Call Control capabilities:

- **Initiating Capabilities**:
  - — **Third Party Make Call** — Used by the adjunct to set up a call. This capability supports the following types of adjunct-controlled calls:
    - — Switch-Classified
    - — User-Classified
    - — Direct-Agent
    - — Supervisor-Assist

    These call types are described in the next section of this feature description, "CallVisor Adjunct/Switch Applications Interface (ASAI)".
    - — User-to-User Information (UUI) Transport can be included in Third Party Make Call.
    - — Third Party Take Control — Used to take control of a call that is already in progress if other third party call control functions (such as hold, drop, and so on) are desired.
- **Control Capabilities**:
  - — **Third Party Selective Drop** — Used by the adjunct to request that a party be dropped from the specified adjunct-controlled call.
  - — **Third Party Selective Hold** — Used by the adjunct to request that a party on a specified adjunct-controlled call be placed on hold.
  - — **Third Party Reconnect** — Used by the adjunct to request that a party be reconnected to a specified adjunct-controlled call.
  - — **Third Party Merge** — Used by the adjunct to request that two adjunct-controlled calls be merged. Merging two calls means conferencing the calls or transferring a held call to an active call. The adjunct specifies the desired type of merge.
  - — **Redirect Alerting Call** — Used by the adjunct to reroute a call that is alerting at a station.

- **Terminating Capabilities**:
  - **Third Party Relinquish Control** — Used by the adjunct to request that a specified adjunct-controlled call no longer send call events. The call must have been previously controlled by the same adjunct.
  - **Third Party Clear Call** — Used by the adjunct to request that a specified adjunct-controlled call be dropped (all parties on that call are dropped and all resources are released from the call).
  - **Third Party Call Ended** — Invoked by the switch as a result of an adjunct-controlled call being dropped. This capability lets the switch terminate the exchange between the switch and the adjunct of messages related to CallVisor ASAI.
- **Event Reporting Capability**:
  - **Event Report** — Used by the switch to convey call feedback and event information in the form of event reports to the adjunct. Call events are sent to the adjunct for monitored and controlled calls.

CallVisor ASAI Call Types

Table 3-48 shows the CallVisor ASAI call types that the Third Party Make Call, the Third Party Auto Dial, and the Route Select capabilities may request.

Table 3-48. CallVisor ASAI Call Types Supported

	Supported by...		
Call Type	**Third Party Make Call**	**Third Party Auto Dial**	**Adjunct Route or Route Request**
Switch Classified Calls	X		
User Classified Calls	X	X	X
Direct Agent Calls	X		X
Supervisor Assist Calls	X		

These call types are described next.

Switch-Classified Calls

A ‘‘Switch-Classified Call’’ is primarily used in Outbound Call Management (OCM) applications with AutoPace or Predictive Dialing.

An adjunct initiates Switch-Classified calls on behalf of the originator (extension number associated with a split, hunt group, or announcement) with only the answered calls are transferred to the originator; other calls are dropped by the switch.

With this type of call, the destination is alerted first and the call is then classified using Call Classification. The destination can be either a valid switch extension or an external number.

It is recommended that trunks with answer supervision be used for switch-classified outbound calls. Answer supervision from the network over the trunk facility provides an accurate and quick indication of when the far end answers.

Answering Machine Detection (AMD)

Answering Machine Detection for Outbound Calls provides Outgoing Call Management (OCM) customers the ability to differentiate between calls answered by a human (‘‘real’’ voice energy) and calls answered by an answering machine.

An ASAI adjunct can request answering machine detection for a switch classified call. When AMD is requested, the reporting of the call outcome is delayed because detection of voice energy is no longer sufficient to classify a call. Instead, further classification is required to distinguish between a live answer and a machine answer.

The range for talk duration is from 0.1 to 5.0 seconds in increments of 0.1. The default value is 2.0 seconds. The range for pause duration if from 0.1 to 2.0 seconds in increments of 0.1. The default value is 0.5 seconds. Customers can alter the default values to suit their needs. For example, to bias the algorithm so that it errors on the side of a line error, the customer could lengthen the call duration.

This feature can be disabled on the System-Parameters Customer-Options form.

User-Classified Calls

A ‘‘User-Classified Call’’ is set up by the adjunct on behalf of an originator to an internal or external destination. It is a call whose originator is alerted first. Such a call is ‘‘user classified’’ because the originator (station user) listens to the call origination and network provided tones, determines the call outcome (for example, busy, reorder, answered), and decides what to do with the call: allow it to continue ringing, drop the call, or speak to the destination or answering party.

This type of call is typically used in ViewFirst or Preview Dialing applications, and can also be used to place and conference/transfer a new call with an existing call. When priority calling is requested, the destination behaves as specified in the "Priority Calling" feature description later in this chapter.

The adjunct must provide the switch with the call destination address and notify the switch of whether priority calling is desired. For Third Party Make Calls, the adjunct must also provide the originator’s address or extension. Valid originators for user classified calls are all voice terminal extensions. Valid destinations are either internal or external extensions [including Vector Directory Numbers (VDNs)] or external numbers.

When initiating user-classified calls, the switch selects an originator's call appearance to set up the call to the destination. If the originator is on-hook, the switch takes the station off-hook if it can (for example, speaker users) and originates the call. Otherwise, the switch waits five seconds for the originator's station to go off-hook on a idle call appearance. If the originator is off-hook with a dial tone, the switch completes the dialing with the destination digits provided by the adjunct.

In the case of an incoming call to a VDN, the Route Select just provides the new destination for the call.

Direct-Agent Calls

A ''Direct Agent'' Call is a special type of ACD call that can be set up via CallVisor ASAI on behalf of any voice terminal user (or VDN caller) to a *specific* ACD agent. Such calls differ from typical ACD calls in that the agent is specified, not selected by the ACD.

NOTE:
This feature is supported through CallVisor ASAI and is not available as a feature from the voice terminal.

Direct Agent calls may be used whenever an adjunct or ACD agent decides that the caller/customer should talk to a specific ACD agent rather than being put in the split queue for an available agent. For instance, in ''adjunct directed'' call routing applications, an adjunct routes incoming calls to a VDN to specific agents based on agent availability and the call's DNIS, CPN/BN, Look-Ahead Interflow information, or digits collected via Call Prompting.

Valid originators for Direct Agent Calls are all voice terminal extensions and VDN callers.

Valid destinations are the following:

- ACD agent extensions logged into an ACD split.
- Station extension of EAS agent (logged in).

  Direct Agent calls are delivered with the highest ACD priority level. That is, Direct Agent calls have priority over all non-Direct Agent ACD calls and are inserted ahead of all non-Direct Agent calls in the split queue.

  NOTE:
  A new Direct Call call agent is queued behind previously queued Direct Agent calls.

  An agent becomes available to receive a Direct Agent call by becoming available to receive ACD calls in the split or (EAS skill) that the Direct Agent call is associated with (that is, going into the ''manual-in'' or ''auto-in'' work mode for the split). When the direct agent call is delivered to the agent and the auto-answer option is enabled on the station set, zip tone is applied to the agent as for any ACD call. If the station set is optioned for manual answering, the call rings the destination agent's station. After disconnecting from a Direct Agent call, the agent's work mode follows the same rules as if the agent disconnects from any other ACD call (refer to the "Automatic Call Distribution (ACD)" feature description earlier is this chapter).

If an agent is not available to receive a Direct Agent call because the agent is busy on another ACD call, or is in the ''auxiliary-work'' or ''after-call-work'' mode for the split associated with the direct agent call, the call queues for the agent on the specified ACD split. Agents are notified of queued Direct Agent calls only once for each Direct Agent call that queues with either a ring ping (for multifunction sets and on-hook analog sets) or a three burst call waiting tone (for analog sets that are off-hook and active on a call).

- If the agent has an analog set, is off-hook, and not active on a call, then the Direct Agent call queues without a call warning tone.
- If the agent has a multifunction set, the active work mode button lamp, if administered for the set, indicates that a direct agent call is waiting in queue by flashing (fast flutter).
- If the agent has an analog set, the three-burst call waiting tone is provided regardless of how the ''Call Waiting Indication'' option is administered for this set.

After a Direct Agent call queues, the switch waits for the agent to become available or until the coverage answer time out limit expires for the destination agent. If this timeout value is reached, the call follows the agent's coverage path. (If the adjunct requests that the Direct Agent call be a priority call, the call does not go to coverage and remains queued until the call is delivered to the agent or the caller disconnects.)

In the case of a Direct Agent call joining the split queue, the destination agent's auto-in or manual-in button associated with the destination split will flash. Flashing will stop when no more Direct Agent calls are in the split queue waiting for this agent.

When an agent is logged into multiple splits, then all Direct Agent calls destined for the agent are serviced before all non-Direct Agent calls in all splits. When there is more than one split with Direct Agent calls waiting for the same agent, the Direct Agent call with the longest queue waiting time is serviced first.

Direct Agent calls are not included in any of the existing measurements affecting queue status displays and buttons. However, such calls are included in CMS measurements.

Supervisor Assist Calls

"Supervisor Assist Calls'' are set up between an ACD agent's extension and another voice terminal extension (typically a supervisor in the ACD split). This feature is used in either ICM or OCM applications when an agent, while on the phone with a client or when idle, wants to consult with the supervisor. If the agent is active on a call, the host application requests that the call first be placed on hold, then request Supervisor Assist Call. Typically, an agent submits this request from a data terminal keyboard. Supervisor Assist Calls are tracked by CMS.

Valid originators for Supervisor Assist Calls are voice terminal extensions logged into an ACD split or EAS skill; valid destinations are internal voice terminal extensions.

Third Party Domain (Station/ACD Split) Control Capabilities Group

Third Party Domain ACD Split Control capability allows an adjunct to receive agent login and logout event reports for a specified ACD split domain.

Third Party Domain Station Control capabilities allow the adjunct to control only the station extension associated with the domain-control association instead of allowing control of all parties (extensions) on a call. With these capabilities, the adjunct can:

- Control all calls originating and ending at a station extension
- Monitor all calls only at that specific station
- Initiate outbound calls from the station

NOTE:
The Third Party Domain (**Station/ACD Split**) Call Control Capability Group is a actually a subset of the Call Control Capability Group. For the purpose of this document, the Domain (Station/ACD Split) Call Group is treated as a separate capability group.

The following capabilities make up the Domain (Station/ACD Split) Call Control capabilities:

- **Initiating Capabilities**:
  - **Third Party Domain (Station) Control Request** — Monitors (receive event reports) and control all calls at a specified extension. After the switch accepts a Third Party Domain Control request from the adjunct, the extension is referred to as a controlled extension. (Only station extensions can be domain controlled by a CallVisor ASAI adjunct.)
  - **Third Party Domain Control Request for ACD Split Domain** — Receives event reports at the specified split domain. The Login and Logout Event Reports are available.
- **Control Capabilities (Station only)**:

— **Third Party Answer** — ‘‘answers’’ a ringing, bridged, or held call present at a station. Answering a ringing, bridged, or held call means to connect a call by forcing the station off hook, if the station is on hook, or cutting through the call to the headset or handset, if the station is off hook (listening to dial tone or being in the off hook idle state).

— **Third Party Selective Drop** — Drops a controlled extension from a call.

— **Third Party Selective Hold** — Places the controlled extension on hold.

— **Third Party Reconnect** — Reconnects a held call at the controlled extension to the controlled extension.

— **Third Party Merge** — Requests that a held call and an active call at the controlled extension be conferenced or transferred.

— **Third Party Autodial** — Originates a call from the controlled extension to an internal or external destination.

— **Redirect Call** — Reroutes a call alerting at a station.

— **Send DTMF Signals** — Sends a DTMT sequence on behalf of one party on an active call.

- **Terminating Capabilities (Station/Split)**:

— **Third Party Relinquish Control** — Stops monitoring and controlling the calls at the controlled extension.

— **Third Party Domain Control Ended** — Used by the switch to inform an application that domain control association has been terminated because the domain was removed or changed to become an invalid domain by administration.

- **Event Reporting Capability (Station/Split)**:

— **Event Report** — Used by the switch to convey call feedback (station) and event information in the form of login and logout events to the adjunct.

Notification Capabilities Group

Notification capabilities are used to request and terminate event reporting on certain calls. The Notification capabilities are as follows:

- Event Notification Request
- Event Report
- Event Notification Cancel
- Event Notification Ended
- Stop Call Notification

An adjunct uses the Event Notification Request function to request notification of calls to VDNs and ACD splits. Such splits cannot be vector-controlled or adjunct-controlled.

NOTE:
This precludes notification on EAS skills, since the EAS skills are equivalent to vector controlled splits.

In G3V3, up to three adjuncts can request notification on a vector directory number (VDN) or split. Once the adjunct requests event notification on a VDN or split, the VDN or split is referred to as a ‘‘monitored’’ VDN or split. From this point on, any call entering this monitored VDN or split causes the switch to start sending call events to the monitoring adjunct, and the call becomes a monitored call. The switch continues to send call events on the monitored call until the call is disconnected, dropped, abandoned, or enters another monitored VDN or split.

Call events are sent to the adjunct via the Event Report capability. Call events may be any of the following:

- Alerting
- Trunk Seized
- Cut-through/Progress
- Disconnect/Drop
- Queued
- Busy/Unavailable
- Answered
- Connected
- Reorder/Denial
- Call Offered to Domain
- Call Originated
- Call Transferred
- Call Conferenced
- Call Ended
- Call Redirected
- Hold
- Reconnected

NOTE:
See the next table in this section for a listing of the call events that apply to each capability group.

The Stop Call Notification capability stops an adjunct from receiving call events for a specific call. The switch no longer sends call events on that call to this adjunct. If a call enters another monitored VDN or split after a Stop Call Notification request, the switch provides event reports for the call over the newly entered monitored VDN or split.

An adjunct can cancel a notification request for all calls within a VDN or ACD with the Event Notification Cancel capability. Call events for a particular VDN or ACD split will no longer be provided.

If a switch administration change causes a VDN or ACD split to become invalid, the switch informs the adjunct using the Event Notification Ended capability.

Set Value Capabilities Group

Set Value allows the following items to be set to a specified value.

- **Message Waiting Lamp**

  The adjunct uses this capability to set the state of a message waiting lamp.

- **Billing Rate**

  The adjunct uses this capability to set the value of a billing change for 900-type calls.

Value Query Capabilities Group

The “Value Query’’ capability allows the adjunct to request information from the switch about the status or value of switch resources.

The following value queries are supported:

- **Time of Day Query** — The adjunct asks for the time. The switch response contains the year, month, day, hour, minute, and second.
- **ACD Split Query** — The adjunct sends a valid split (or EAS skill) extension in a value query message. The switch responds with the number of ACD agents available to receive calls in that split or EAS skill, the number of calls in queue, and the number of ACD agents logged in.
- **Trunk Group Query** — The adjunct sends a trunk access code in a value query message. The switch responds with the number of idle trunks in the trunk group and the number of in-use trunks.
- **Call Classifier Query** — The adjunct sends a classifier request in a value query message. The switch responds with number of idle and in-use call classifier ports.
- **ACD Agent Status Query** — The adjunct sends an ACD split (or EAS skill) and an agent extension in a value query message. The switch responds with the work mode and idle/busy state of the agent.

- **Extension Query** — The adjunct sends an extension number in the value query message and the switch responds with the type or class (for example, VDN, Hunt Group, EAS skill, Announcement, Voice).
- **ACD Agent Login** — The adjunct sends an ACD split extension and an ACD agent login audit flag in a value query message. The switch responds with a list that contains the extensions for each agent logged into the split.
- **Calls Query** — The adjunct passes an extension in the value query request. The switch responds with the call_id(s) for the calls present at the primary's extension, and also sends the party_id and party_state (for example, alerting, connected, or hold) of that station for each call. Additionally, information about a maximum of 10 calls present at the station is reported back to the adjunct. This query is valid for stations and logical agents only.
- **Station Status Query** — The adjunct uses this query for status information on the extension number of the endpoint. The switch responds with the talk (idle/busy) state of the voice terminal.
- **Party ID Query** — The adjunct sends a call_id in a value query message. The switch responds with the party_id and the extension number for all the local parties on the call.
- **Station Feature Query** — The application passes an extension number and the feature to the switch. The extension need not be domain-controlled, but it must support the particular feature. The switch responds with the state (active/not-active) of the specified feature at the specified extension.

  The following features may be queried:

  — Message Waiting Indication

  — Send All Calls

  — Call Forwarding
- **BCMS Data Value Query** — The adjunct uses this query to retrieve BDMS data that was collected for agent, agent extension, split/skills, trunk group, or VDN.
- **ASAI-Accessed Integrated Directory Database** — The application passes a target extension to the switch. The switch responds with the administered names associated with the specified extension.

Request Feature Capabilities Group

This capability allows the adjunct to request activation of the following switch features.

- **ACD Agent Login** — Log in agent to an ACD split or EAS skill. The following parameters are required for this function:

  — Logical agent login id (EAS skill only)

— Login Identifier (password) — any string of length equal to or greater than that administered on the switch.

— ACD Split Extension (or EAS skill)

— Agent Extension (Contains physical extension for EAS agent)

— Mode: After Call Work, Auto-In, Manual-In, Auxiliary Work (optional, corresponds to initial work mode; if not specified, the default is Auxiliary Work)

- **ACD Agent Logout** — Logout agent from an ACD split. The following parameters are required for this function:

  — ACD Split Extension (or EAS skill)

  — Agent Extension (physical or logical for EAS agent)

- **ACD Agent Change of Work Mode** — Change work mode of ACD or EAS agent to another mode. The following parameters are required for this function:

  — ACD Split Extension

  — Agent Extension

  — Mode: After Call Work, Auto-In, Manual-In, Auxiliary Work (corresponds to new work mode)

  NOTE:
  Work mode change ‘‘pending’’ states are not supported by ASAI. (For example, if the agent is active on an ACD call and is in Auto-In work mode, a request to place the agent in After Call Work mode is denied.)

- **Send All Calls** — Activate or cancel Send All Calls at a specific station. The adjunct provides the extension where Send All Calls (SAC) is activated.

- **Call Forwarding** — Activate or cancel Call Forwarding at a specific station. The adjunct provides the ‘‘forwarded from’’ and ‘‘forwarded to’’ extensions.

Adjunct-Controlled Splits

The switch may have adjunct-controlled splits. An adjunct-controlled split is an ACD split designated as adjunct-controlled via switch administration. An adjunct-controlled split has the following properties:

- An adjunct-controlled split must have an administered controlling adjunct (CallVisor ASAI link extension).
- Agent login may only be done by the adjunct via Request Feature capabilities.
- An agent may be logged into only one such split at any given time.

- Change of work modes may only be done by the adjunct via Request Feature capabilities.
- Work mode changes take effect as soon as they are requested and processed.
- Agent logout may be done by the adjunct via Request Feature capabilities or by the agent. When logout is caused by action at the voice terminal, a logout event report is sent to the adjunct associated with the split (provided domain control has been requested for that split).
- As long as the corresponding CallVisor ASAI link is up, Agents logged into an adjunct-controlled split have their voice terminal locked (all buttons and the touch tone pad) for the duration of the login. These agents must use Call Control and Request Feature capabilities to access voice terminal calling and ACD support features. (Adjunct-controlled splits cannot receive non-adjunct monitored and controlled calls. Non-adjunct monitored and controlled calls receive busy tones if they try to terminate at such splits.)
- Agents logged into adjunct-controlled splits can only receive calls that are monitored (via the Event Notification group) or controlled (via the third person Call Control group), primarily ASAI-OCM calls.
- An unmonitored or controlled call directed to an adjunct-controlled split receives a busy tone.
- A unmonitored or controlled call directed to an agent logged into an adjunct-controlled split receives a busy tone.
- Agents logged into an adjunct-controlled split may not make any calls from the voice terminal. All telephony functions must be done via third party call control.
- Adjunct-controlled splits may not have notification active. A request notification for an adjunct-controlled split will be denied.
- An adjunct-controlled split may be vector-controlled as well.
- When an agent is logged into an adjunct-controlled split, all display buttons are disabled. The display itself functions normally.

When the CallVisor ASAI link is down, adjunct-controlled splits behave like non-adjunct-controlled splits. Agents logged into such splits when the link is down have their voice terminals unlocked without being logged out. When the CallVisor ASAI link is restored, adjunct controlled splits return to being adjunct-controlled, and agent’s voice terminals become locked again.

Adjunct-controlled splits may also be vector-controlled. Splits that are both adjunct-controlled and vector-controlled have all the properties of both. Where there is a conflict, the more restrictive property applies. For example, agents logged into such a split cannot log into any other split and their voice terminal is locked.

An agent cannot be logged into multiple splits if that agent is logged into an adjunct-controlled split. Because of the numerous restrictions associated with adjunct-controlled splits, the use of such splits should be limited.

Routing Capabilities Group

The routing capabilities allow the switch to request (from the adjunct) routing instruction for a call. The adjunct may use incoming call information (such as CPN/BN and DNIS) to route the call to an internal number, external number, a split, a VDN, an announcement extension, or a particular agent. The adjunct might also determine the best route for the incoming call based on the called number, the dialed digits from a call prompter, the customer database, or agent availability. Using this information, the adjunct can also provide priority ringing, priority queuing, or personalized handling of incoming calls by rerouting the call to a VDN or a specific agent. The Routing Capabilities Group requires that Basic Call Vectoring be enabled. If digits are collected from the caller, Call Vectoring Prompting must also be enabled.

The following Routing Capabilities are provided:

- **Route**

  Adjunct routing is initiated with a **Route** request which is a message from the switch to the adjunct requesting the routing information. The **Route** capability is initiated by the switch when it encounters the ‘‘adjunct routing’’ command in a call vector. For details, see the *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide*, 555-230-520.

- **Multiple Outstanding Route Requests**

  Multiple Outstanding Route Requests allows multiple ASAI route requests for the same call to be active at the same time. The route requests can be over the same or different ASAI links. The requests are all made from the same vector. They must be specified back-to-back, without intermediate (wait, announcement, goto, or stop) steps. If the adjunct routing commands are not specified back-to-back, current adjunct routing functionality applies (that is, previous outstanding route requests are canceled when an adjunct routing vector step is executed).

  The first route select response received by the switch is used as the route for the call, and all other route requests for the call are canceled.

- **Route Select**

  The **Route Select** capability is sent from the adjunct to the switch in response to the **Route** capability. It provides the switch with the destination address where the call will be routed. In addition, the adjunct can request the switch to route the call as a Direct Agent call and/or a priority call. Other options include ASAI-provided digits, ASAI-requested digit collection, and user-to-user information (UUI).

- **ASAI-Provided Digits**

The ASAI-Provided Digits feature allows an adjunct to include digits in a Route Select capability. These digits are treated as dial-ahead digits for the call, which are stored in a dial-ahead digit buffer. They can be collected (one at a time or in groups) using the **collect digits** vector command(s).

Although the adjunct can send more than 24 digits in a Route Select, only the first 24 (or 24-x) digits are retained as dial-ahead digits. The rest are discarded. (The maximum number of dial-ahead digits that can be stored in the buffer is dependent on the number of digits already collected for the call by a previous **collect digits** command. If x-digits are collected by vector processing before executing an **adjunct vector** command, the x-digits collected reduces the maximum number of digits that can be stored as dial-ahead digits as a result of a Route Select.)

- **ASAI-Requested Digit Collection**

  ASAI-requested digit collection gives an adjunct the ability to request that a DTMF tone detector be connected for the purpose of detecting user-entered digits. The request is made via an option of the **Route Select** message. The digits collected as a result of this feature are passed to ASAI monitoring and/or controlling adjuncts for action. The switch handles these digits like dial-ahead digits.

  These digits are not necessarily collected while the call is in vector processing. They are sent to an ASAI adjunct, and/or they can be used by Call Prompting features.

- **Route End**

  The **Route End** capability is sent by the switch to terminate routing. A call that is already routed is not affected by this termination. It contains a success or failure indication to indicate if the call was successfully routed.

  User-to-User Information (UUI) Transport is also supported.

Maintenance Capabilities Group

The maintenance capabilities are used to disable and enable switch-administered alarms for periodic link maintenance and to obtain information about the condition of the CallVisor ASAI and the CallVisor ASAI link. The following Maintenance Capabilities are provided:

- **Heartbeat** — This capability enables the adjunct or the switch to send an application to application message and receive a response in order to determine the sanity of the application on the remote endpoint.
- **Suspend Alarms** — This capability enables the adjunct to disable switch alarms on an CallVisor ASAI link for maintenance functions.
- **Resume Alarms** — This capability enables the adjunct to resume switch alarms on an CallVisor ASAI link.

CallVisor ASAI Applications

The following examples are a few of the many CallVisor ASAI applications:

- **Data Screen Delivery**

  The system can pass network information such as CPN/BN, DNIS and switch information such as digits collected via the integrated call prompter and agent extensions to a host/adjunct if the host/adjunct requests notification. The host/adjunct can use this information to automatically display the proper data screen.

- **Data Screen Transfer**

  The system can notify the host/adjunct of the transferred-to party, if the adjunct is monitoring the call. This allows the host/adjunct to transfer the data screen associated with the call.

- **Adjunct Routing**

  The system can request routing from the host/adjunct. The host/adjunct may respond with any valid internal/external number.

- **Direct Agent Calling**

  The host/adjunct may wish to route or transfer (using third party make call) a call to a particular agent, yet have this call treated as an ACD call and tracked as an ACD call by CMS. The Direct Agent Calling function provides this treatment.

- **Supervisor Assist**

  The host/adjunct may initiate a call on behalf of an agent to the agent's supervisor (as defined by the host/adjunct). This call is treated as a supervisor assist call by CMS.

- **Voice Processor Integration**

  The system can send information such as CPN/BN, DNIS, or agent selected to the speech processing adjunct(s). The speech processor can integrate this information with information the speech processor collects and send it on to a host for screen delivery or call routing.

- **View First Dialing**

  The system allows a host/adjunct to place a call on behalf of an agent. The agent may have previewed a data screen and used the data keyboard to tell the host/adjunct to begin this call.

- **Auto Pace Dialing**

  The system allows a host/adjunct to place several calls on behalf of a group of agents. The system classifies the call, sending only answered calls to the agents. The system notifies the host/adjunct of each classification. Special Network Information Tones are detected and may be treated as answered or dropped. Answering Machine Detection is available with Auto Pace Dialing.

- **Agent Work Mode Control**

  The system allows the host/adjunct to log agents into and out of various splits. The system also allows the host/adjunct to change agents' work modes. A supervisor might initiate this type of activity via the host/adjunct or the host/adjunct might automatically make a change due to work loads.

Considerations

A maximum of eight (four for G3s PBP) CallVisor ASAI links may be administered.

NOTE:
The switch does not recognize or address CallVisor ASAI messages to/from specific applications on an CallVisor ASAI link. Multiple applications may exist on the adjunct, but the CallVisor ASAI interface does not address (for example) event reports to a specific application; the adjunct has to determine which of its applications receives the messages from the switch. The switch will, however, send multiple copies of the same event, one for each active monitor and controller for a given call or user extension.

— The maximum number of simultaneous outstanding route requests per call is 16.

— The maximum number of pending routing requests on a CallVisor ASAI link is 127.

— The maximum number of simultaneous notification requests is 50 for G3s PBP, 170 for G3i, 460 for G3rV1, and 3000 for G3rV2.

— The maximum number of active adjunct-controlled calls is 75 for G3s PBP, 300 for G3iV1 and G3iV2, and 3000 for G3r.

— The maximum number of active Third-Party Domain control stations is 250 for G3s PBP, 2000 for G3i and 6000 for G3r.

— The system-wide maximum number of monitors per call is 3.

— The system-wide maximum number of domain controllers per station domain is 2.

— The system-wide maximum number of domain controllers per ACD split domain is 3.

— The system-wide maximum number of call controllers per call is 1.

— The maximum number of simultaneous OCM Predictive Dialing/Auto Pace calls in the classification stage per system is 20 for G3s, 40 for G3i and PBP, and 400 for G3r. After the call is classified as answered, no-answer, busy or reorder, the call does not count toward this limit.

— The maximum number of simultaneous billing change requests is 25 for G3s, 100 for G3i, and 1,000 for G3r.

— The maximum number of simultaneous internally measured data queries is 25 for G3s/G3i and 50 for G3r.

Feature Initialization and Recovery

The effect of various levels of switch restarting on ASAI links, adjunct-monitored calls, adjunct associations and adjunct-controlled agents is detailed below:

- **Hot Start, Warm Start**
  - — Normally hot and warm starts do not affect ASAI, since the ASAI links will remain up during these re-starts. However, such a restart causes the PACCON to send a layer@2 inquiry over the ASAI links. Normally, the adjunct will reply and the link remains up. If the adjunct does not reply, the switch tears down the link and unanswered and queued switch classified calls are dropped. The PKTINT does not send a layer 2 inquiry over the ASAI links.

    During a Warm Start, any buffers that are awaiting transmission are discarded by the switch.
- **Coldstart Level 2**
  - — All calls are dropped
  - — All links (and therefore ASAI/BRI links) are dropped; all active associations are cleaned up
  - — All status data associated with links is initialized
  - — Agents are logged out and voice set locks removed
- **Coldstart Level 1, Reboot**

  In addition to the effect specified under Coldstart Level 2,
  - All translation data (link parameters) is initialized.

    NOTE:
    SAC or *CFW* may be lost if ‘‘save translations’’ was not done before Cold 1.
- **Extended Reboot**

  In addition to the effect specified under Coldstart Level 1,
  - All initialization will be run twice.

Interactions

The following features and functions interact with the CallVisor ASAI feature.

- **Abbreviated Dialing**

When an agent is logged into an adjunct-controlled ACD split, programmed Abbreviated Dialing buttons or feature activation buttons are disabled.

Abbreviated Dialing cannot be invoked through a Third Party Call Control request.

- **Announcements**

  ACD split forced first or second announcements and vector announcements do not generate event reports for the adjunct. However, nonsplit announcements generate events which are sent to other parties on the call.

  Extensions assigned to integrated announcements may not be domain controlled. The Third Party Auto Dial capability may specify integrated announcement extensions as destination endpoints.

- **Answer Detection**

  CallVisor ASAI switch-classified calls, Answer Detection, and Answering Machine Detection share the same set of call classifier ports.

- **Answer Supervision**

  The “Answer Supervision Time-Out” field determines how long the CO trunk circuit pack waits before sending the (simulated) answer message to the software. This is useful when true answer supervision is not available on a trunk. This message is used to send call information to CDR to generate the ASAI "connect" event report, and to trigger the bridging of a service observer onto an outgoing trunk call. This timer is ignored if the trunk receives true answer supervision from the network (the switch uses the true answer supervision whenever available). Monitored or controlled calls do not change this operation.

  With respect to switch-classified calls, when the “Answer Supervision yes/no (y/n)” field is set to ‘‘no,’’ the switch relies entirely on the call classifier to determine when the call was answered. When answer supervision on the trunk is set to ‘‘yes,’’ a switch-classified call is considered answered when the switch software receives the answer message from the trunk circuit pack. In reality, switch-classified calls may receive either an answer message from the trunk circuit pack or (if this never comes) an indication from the classifier that the far end answered. In this case, the switch acts on the first indication received and ignores any subsequent indications.

- **Attendant Auto-Manual Splitting**

  If an individual attendant receives a call with active domain-control associations and then activates the Attendant Auto-Manual Splitting feature, a Hold Event Report is returned to the associations controlling the extensions adjunct. The next event report sent depends on what button the attendant presses on the set (CANCEL = Reconnect, SPLIT = Conference, RELEASE = Transfer).

- **Attendant Call Waiting**

Calls that provide event reports over domain-controlled associations and are extended by an attendant to a local, busy, single-line voice terminal will generate the following event reports:

- — Hold - when the incoming call is **split away** by the attendant
- — Connect - when the attendant returns the call

The following events are generated, if the busy station does not accept the extended call and its returns.

- — Alerting - when the call is returned to the attendant
- — Connect - when the attendant returns to the call

- **Attendant Control of Trunk Group Access**

  Trunks seized for switch-classified Third Party Make Call attempts must not have attendant control activated. If they do, such calls are denied.

  If the adjunct attempts to route a call to a such a trunk, via adjunct routing, the route fails and vector processing for the call is continued.

- **Attendant and Attendant Groups**

  Individual attendants may be parties on adjunct-monitored calls and are supported like regular voice terminal users. The attendant group 0 is not supported in CallVisor ASAI Call Control requests and Event Reports are not provided for the attendant group.

- **Attendant Timers**

  Vectoring causes all other timers to be ignored.

- **AUDIX**

  Calls that cover to AUDIX do not maintain a simulated bridge appearance on the principal's station. The principal receives audible alerting followed by an interval of coverage response followed by the call dropping from the principal's set. When the principal receives alerting, the Alerting Event Report is sent. When the call is dropped from the principal's set, the Call Redirected Event Report is sent to domain controllers for the principal's station. Other monitors and controllers for the redirected call receive a Queue or Alerting Event Report for the AUDIX hunt group or port.

- **Authorization Codes**

  Calls that require authorization codes are not supported for switch-classified calls. If the check for authorization codes fails, the call request is denied.

- **Automatic Callback on Busy/Does Not Answer**

  Automatic Callback on Busy/Does Not Answer can be activated by a controlled station user. The callback appears as an incoming call to the domain controller(s) for the station.

  Automatic Callback on Busy/Does Not Answer cannot be activated by the adjunct over the CallVisor ASAI interface.

- **Automatic Call Distribution**

  ACD calls can be adjunct monitored and controlled; ACD agents/stations can be domain controlled. CallVisor ASAI does not affect how ACD calls are handled by the switch.

  An adjunct can monitor (Event Notification) hunt groups administered as ACD but not as vector controlled or adjunct controlled. The switch denies Event Notification requests for vector-controlled or adjunct-controlled ACD splits including EAS skills.

  For every call that enters the ACD split, a Call Offered to Domain Event report is provided to the ACD split monitor. If the call queues to the split, a Queue Event Report is provided. When the call is delivered to an ACD agent (zip tone or station alerting), an Alerting Event Report is provided. No event report are provided for first or second forced announcements.

  ACD is used by OCM predictive dialing applications to distribute answered outbound calls. When an ACD split or hunt group is specified as the originator of a switch-classified Third Party Make Call, the switch automatically distributes the answered calls to the available agents on the specified split. (It is recommended that agents handling OCM predictive dialing calls be administered as automatic answer with zip tone.)

- **Automatic Route Selection (ARS) and Automatic Alternate Routing (AAR)**

  The ARS and AAR features are accessible by CallVisor ASAI adjuncts through Third Party Make Call requests.

  When ARS/AAR is used, if the adjunct wants to obtain trunk availability information, it must query the switch about all trunk groups in the ARS partition dedicated for that application/adjunct. The adjunct may not use the ARS/AAR code in the query to obtain trunk availability information.

  When using ARS/AAR, the switch does not inform the adjunct which particular trunk group was selected for a given call.

  Care must be given to the proper administration of this feature, particularly the FRLs. If these are not properly assigned, it is possible that calls will be denied regardless of trunk availability.

  If required by switch administration, the adjunct must include the leading ‘‘1’’ in the destination number of Third Party Make Call, Third Party Auto Dial, and Route Select requests for long distance calls. (The switch deletes the leading ‘‘1’’ if switch administration indicates that it is not needed.)

- **Bridging**

  Direct Agent calls to an agent behave like regular ACD calls and do not alert bridged users of the agent’s extension. Monitored or controlled calls delivered to stations with bridged appearances provide an Alerting Event Report for the principal and for every alerted bridge appearance. A Connected Event Report is sent every time the principal or a bridged user connects to the call. Unless the complete bridge is dropped from the call, no Event Reports are provided when the principal or a bridge user disconnects from the call. The switch provides a Drop Event Report when the complete bridge is disconnected.

Third party make calls delivered as non-ACD calls can alert bridging users. Appropriate events are sent for each bridging user and principal. However, Third Party Selective Hold, Third Party Merge, Third Party Reconnect, and Third Party Selective Drop are not permitted on parties in the bridged state.

- **Busy Verification of Terminals**

  A Connected Event Report is provided when the verifying user is bridged onto a monitored or controlled call.

  A Dropped Event Report is provided for the verifying user when the user completes the busy verification.

- **Call Coverage**

  Monitored or controlled calls generate an Alerting Event Report for each coverage endpoint that is alerted.

  Switch-classified calls delivered to a local (on switch) destination are not allowed to go to coverage and remain alerting on the principal destination, even if the coverage criteria is met. Switch classified calls delivered to the originator of the call are allowed to go to coverage (either for the split or the voice terminal) provided the coverage criteria are met.

  Direct Agent Calls follow the agent's coverage path rather than the split's coverage path.

  If an adjunct-monitored call goes to coverage and is answered or picked up using the Call Pickup feature, the switch sends events to the adjunct.

- **Call Forwarding**

  Monitored or controlled calls (except for the destination of a switch classified call) may be forwarded, even when Priority Calling is specified, provided the forwarding criteria are met.

  Direct Agent calls will forward if the destination split has call forwarding activated. After a Direct Agent call successfully terminates to the destination split, the call is forwarded if the destination agent has call forwarding activated.

  If any monitored or controlled call is forwarded and is answered or picked up using the Call Pickup feature, then the switch sends an Alerting and Connected event to the adjunct with the actual party extension that was connected. If the call is forwarded off-premises via a non-PRI trunk, a ‘‘trunk seized’’ event is sent.

  Forwarded monitored or controlled calls (including Direct Agent calls) are treated like regular ACD calls when offered to a split. They are treated like non-ACD calls when offered directly to a voice terminal user. When an adjunct-monitored call gets forwarded to an internal destination, an alerting event is sent for the forwarded-to station. No alerting event is sent for the principal.

Switch-classified calls offered to destinations with Call Forwarding active remain at the destination. Switch-classified calls offered to originators with Call Forwarding active are forwarded.

Adjuncts may activate call forwarding for an EAS agent logical identifier.

- **Call Park**

  A call parked at a station using the transfer operation (switchhook flashes, Transfer button, Call Park button, or attendant or code calling access), causes the switch to report a Call Transferred Event Report for the call. If the call is parked using a conference operation (Conference button), the switch reports a Call Conferenced Event Report for the call. The ‘‘transferred to’’ extension provided in the Event Report is the extension where the call is parked. When a user or the attendant ‘‘unparks’’ the call, a Connected Event Report is generated for the call.

- **Call Pickup**

  A monitored or controlled call alerting a member of a Call Pickup group generates a single Alerting Event Report for the alerting member.

  When the call is answered using Call Pickup, the switch provides a Connected Event Report that includes the extension of the answering station in the ‘‘Connected Number” field.

  The domain controller(s) of principal’s station does not receive any subsequent events for the call until the principal connects to the call or the call is dropped from the station.

  An attempt to pick up a switch-classified call alerting a destination station is denied by the switch. Pick up of a switch classified call alerting an agent’s station (originator) is allowed.

- **Call Prompting**

  Up to 24 digits collected from the last collect digit vector command are passed to the adjunct in the Call Offered to Domain Event Report and the Route capability. CallVisor switch classified calls and Call Prompting use the same call classifier ports.

  A maximum of 80 for G3i, and 400 for G3r call classifier ports are available for Call Prompting, of which 40 for G3s PBP and G3i, and 400 for G3r, are available for switch-classified calls. The rest of the ports are available for Call Prompting.

- **Call Vectoring**

  An adjunct can only monitor (Event Notification) Vector Directory Numbers (VDNs) associated with Call Vectoring. Direct monitoring of call vectors is not allowed.

An event notification request for a vector-controlled ACD split extension is denied.

A Call Offered to Domain Event Report is provided to a VDN monitor for every call that enters a vector via the monitored VDN.

A Queue Event Report is provided every time a monitored call successfully queues to a split, via queue to main, check backup, or messaging split vector commands. Additional queue events are provided for the same split when queuing to the same split with a different priority.

The Alerting Event report provides an indication that the call has been delivered to an agent and dequeued from all splits to which calls have been queued. A specific cause value is provided if the call alerts a converse agent.

A Busy Event Report for the call is sent when a busy vector command is encountered.

Direct Agent calls are not included or counted as part of the conditional threshold checks (for example, oldest call waiting or number of calls in queue) of the call vectoring commands (for example, Check Backup Split and Goto Step).

Third Party Make Call requests cannot have a VDN as the originator unless EAS is enabled.

A vector-controlled split could be adjunct-controlled as well. If it is, the more restrictive properties of each apply.

A vector-controlled split is not mapped to an adjunct via administration, unless it is also adjunct-controlled.

A vector-controlled split (including an EAS skill) cannot be monitored. An event notification request for a vector-controlled split is denied.

VDN Return Destination does not take effect if the call has Flexible Billing. (If all parties but the calling party drop from a call with Flexible Billing and VDN Return Destination, the call is dropped.

- **Call Waiting**

  The Call Waiting tone is used to alert an analog user when a Direct Agent call is waiting.

- **CDR**

  Calls originated by the adjunct via the Third Party Auto Dial capability or Third Party Make Call capability are marked with the condition code ‘‘B.’’ Adjunct originated calls include calls originated by forcing the user off-hook after a Third Party Auto Dial request or Third Party Make Call request; calls originated by the user going off-hook and then requesting Third Party Auto Dial or Third Party Make Call.

  Calls originated manually from a domain-controlled station are not marked with condition code ‘‘B.’’

Starting with G3V4, for externally directed calls, the CDR record shows the redirected-to number as the called number and the redirected extension as the calling number.

- **Class of Restriction (COR)**

  Third Party Make Call attempts are placed using the originator's COR (voice terminal's or split's). The COR associated with the adjunct's link is not used at all.

  For switch-classified calls, if the destination's COR check fails, the call is dropped. COR checking is not done for the originator of a switch-classified call.

  A "Direct Agent Calling" field on the Class of Restriction form indicates whether the user can originate and receive Direct Agent calls. If either the originating or the destination party of a Direct Agent call does not have the proper COR, the call is denied.

  In the case of adjunct routing, the COR of the associated VDN is used for calling party restriction checks.

- **Conference/Transfer**

  When an agent is logged into an adjunct-controlled split, Conference and Transfer can only be done via the agent's data terminal, since the voice terminal is locked.

  A voice station user is allowed to transfer or conference monitored/controlled calls at their voice station using the switchhook flashes or the Conference and Transfer buttons. The switch reports the first switchhook flash or push of the Transfer or Conference button as a Hold Event Report. When the conference or transfer operation completes (second push of the Transfer or Conference button), the switch provides a Call Conferenced or Call Transferred Event Report containing a list of parties active on the call.

- **Consult**

  When the covering user presses the Conference or Transfer feature button and receives dial tone, a Hold Event Report is returned to all adjuncts monitoring or controlling the call. The adjunct controlling the covering user's station receives a Call Initiated event when the covering user listens to dial tone after pressing the Conference of Transfer button.

  A Call Initiated Event Report is then returned to the covering user's adjuncts. After the Consult button is pressed by the covering user, Alerting and Connected Event Reports are returned to the principal's and covering user's adjuncts. The covering user can then conference or transfer the call.

- **Coverage**

Coverage timer. When a call is redirected, the coverage time for the principal's party is restarted. Thus, a call may go to coverage (principal's) after being redirected.

Don't Answer Coverage (DAC) timer. If the DAC timer is assigned to the agent receiving a Direct Agent call, and the call is redirected, the DAC timer is cancelled. It is reset to the original value later.

Redirection and Coverage Calls. If a call is alerting at the coverage point and the alerting station is a station extension, the call may be redirected via the Redirect Alerting Call feature, provided the request specifies the alerting extension, not the principal.

Redirection and Caller Response Interval (CRI). If a call is redirected from the principal during the CRI, the CRI timer is cancelled. The call then won't go to coverage unless the new destination doesn't answer.

Coverage/Redirection Tone. When a call is redirected, a redirection tone is not played.

- **Distributed Communication Systems (DCS)**

  Direct Agent calls cannot be made over a DCS link. The destination on a Direct Agent call must be an internal ACD agent extension.

  Third Party Make Call, Third Party Auto Dial, and Route Select requests (excluding Direct Agent and switch-classified calls) can be placed over a DCS network. They are treated like off-switch calls.

  CPN/BN is usually not provided unless the call is originated from within the DCS network. When provided, it is the transparent DCS extension of the originator.

- **Do Not Disturb**

  Activation of this feature by an ACD agent only blocks personal calls from terminating at the agent's voice terminal. ACD calls (including Direct Agent and Switch-Classified) are still delivered to the ACD agent when this feature is activated.

- **Drop Button Operation**

  The operation of this button is not changed with CallVisor ASAI.

  When **Drop** is pushed by one party in a two-party call, the Disconnect/Drop Event Report is sent with the extension of the party that pushed button. The originating party receives dial tone and the Call Initiated Event Report is reported on its domain-controller.

  When **Drop** is pushed by the controlling party in a conference, the Disconnect/Drop Event Report is sent with the extension of the party who was dropped off the call. This might be a station extension or a group extension. A group extension is provided in situations when the last added party to a conference was a group (for example, TEG, split, and announcement) and **Drop** was used while the group extension was still alerting (or was busy). Since the controlling party does not receive dial tone (it's still connected to the conference), no Call Initiated Event Report is reported in this case.

- **Expert Agent Selection (EAS)**

  For G3V2 only, EAS allows incoming calls to be routed to specialized groups of agents within a larger pool of agents. With EAS, a set of skills are assigned to ACD agents based on their login identifiers (LoginIDs) and to incoming calls based on the vector directory number (VDN) associated with the calls. Incoming calls are delivered to the appropriate agent by matching the call's assigned skills and the agent's skills.

  When EAS is enabled all ACD hunt groups become Skill hunt groups and all ACD agents become logical agents.

  Also for G3V2, two types of direct agents are supported:

  — **Physical Direct Agent Calls**

  Physical direct agent call are always available independently of EAS. These calls can only be originated by an ASAI adjunct.

  — **Logical Direct Agent Calls**

  Logical Direct agent calls are only available when EAS is enabled. Logical direct agent calls can be adjunct initiated or voice terminal initiated and must be allowed by the originating and destination stations' class or restriction (COR). Otherwise the call is treated as a personal call to the specified agent.

- **Expansion Port Network (EPN)**

  The Expansion Interface (EI) board makes it possible for CallVisor ASAIs to terminate on an EPN as well as the Primary Port Network (PPN).

  It is recommended that any CallVisor ASAIs that are critical to your company's business terminate on the PPN to enable the CallVisor ASAI to remain operational in the event of a fiber link or EI failure. Further, resources that are used by a critical CallVisor ASAI adjunct such as classifiers, trunks, announcements, and agent ports should also home on the PPN for the following reasons:

  — To keep these resources in service in the event of a fiber link or EI failure; and

  — To minimize the amount of cross carrier traffic that could degrade CallVisor ASAI response time and system performance.

- **Forced Entry of Account Codes**

  Switch-Classified Call attempts to trunk groups with Forced Entry of Account Codes are denied. Agents logged into Adjunct-controlled splits cannot enter account codes from their voice station.

- **Hold**

When an agent is logged into an adjunct-controlled split, Hold can be invoked only from the agent's data terminal. The adjunct must be able to invoke Third Party Selective Hold on behalf of a party. This party cannot be an attendant, trunk, announcement, vector, or split.

When the voice terminal is not locked, and the user has a monitored or controlled call at the voice terminal, the user may place the call on hold and reconnect it to the voice terminal any number of times. A hold event is reported for calls placed on hold either via switch-hook flash, hold button, or conference/transfer button.

- **Hot Line**

  A Third Party Make Call request made on behalf of a voice terminal that has this feature administered is denied.

- **Hunt Groups**

  Event notification requests are not allowed for hunt groups (other than ACD splits). Therefore, the call offered event is never sent for hunt groups. However, the queued event is sent for hunt groups when an adjunct-monitored call queues at a hunt group.

- **Interflow**

  When a monitored or controlled call interflows to another switch, adjunct notification ceases except for trunk events (for example, Trunk Seized or Drop).

- **Intraflow**

  Direct Agent calls do not intraflow since they follow the agent's coverage path, rather than the split's.

- **ISDN**

  The Third Party Auto Dial calls follow ISDN rules for the originator's name and number. The Call Initiated Event Report is not sent for *en-bloc* BRI sets.

- **ISDN-PRI Facilities**

  ISDN-PRI Facilities may be used by either inbound or outbound adjunct monitored or controlled calls.

  An incoming call over an ISDN-PRI facility (if so provisioned by the network) provides the calling and called party information which is passed to the adjunct in the Event Report and Route capabilities.

  An outgoing call over an ISDN-PRI facility provides call feedback events from the network (Cut-through, Alerting, and Connected). Trunk Seized Event Report is not provided for outgoing calls using ISDN-PRI facilities.

  Switch-Classified calls always use a call classifier on ISDN-PRI facilities whether the call is interworked or not.

- **Last Number Dialed**

  The destination number of Third Party Make Call and Third Party Auto Dial requests (except switch-classified call requests) is recognized as the last number dialed for the originator of the adjunct controlled call. A user pushing the last number dialed button originates a call to the number last dialed on its behalf through a Third Party Make Call or Third Party Auto Dial attempt. A call originated in this fashion is not adjunct-monitored.

- **Lookahead Interflow**

  For the receiving PBX, the Lookahead interflow information element passed in the ISDN message is included in all subsequent call offered Event Report and Route capabilities for the call when the information exists. This information element includes ANI information, if available, the originally called VDN, but does not include any digits collected via call prompting at the original switch.

- **Multiple Split Queuing**

  When a call is queued in multiple splits/skill hunt groups, the "Party ID Query'' provides, in addition to the originator, only one of the split/skill hunt group extensions in the party list. When the call is dequeued, the Alerting Event Report provides the split/skill hunt group extension of the alerting agent. There are no other events provided for the splits from which the call was removed.

  A Queued Event Report is provided every time a monitored or controlled call is queued to a split, or when the call requeues to the same split at a different priority.

- **Music-On-Hold**

  Music-on-Hold (if administered and available) is provided to a call placed on hold via the Third Party Selective Hold capability.

- **Night Service**

  A controlled station can be a night service station. Direct Agent calls directed to an agent on a split with night service activated redirect to the night service extension, even if priority calling is requested.

  Switch-Classified calls delivered to an originating split with night service activated, redirect to the night service extension.

- **Personal Central Office Line (PCOL)**

  Members of a PCOL may be domain-controlled. PCOL behaves like bridging for the purpose of CallVisor ASAI event reporting. When a call is placed to a PCOL group, the Alerting Event Report is provided to each member's domain-controller(s). The called number information passed in the alerting message is the default station characters. When one of the members answers the incoming call, the Connected Event Report is provided with the station which answered the call. If another members connects to the call, another Connected Event Report is provided. When a member goes ''on hook'' but the PCOL itself does not drop from the call, no event is sent but the state of that party changes from the connected state to the bridged state. The Disconnect/Drop Event Report is not sent to each member's domain-controller(s) until the entire PCOL drops from the call as opposed to an individual member going on hook.

  Members that are not connected to the call while the call is connected to another PCOL member are in the bridged state. When the only connected member of the PCOL transitions to the held state, the state for all members of the PCOL changes to the held state even if they were previously in the bridged state. There is no event sent to the bridged user domain-controller(s) for this transition.

All members of the PCOL may be individually domain-controlled. Each receives appropriate events as applicable to the controlled station. Third Party Call Control requests are not recommended. However, Third Party Selective Hold, Third Party Merge, Third Party Reconnect, and Third Party Selective Drop are not permitted on parties in the bridged state and may also be more restrictive if the exclusion option is in effect from a station associated with the PCOL.

A Third Party Auto Dial or Third Party Make Call originate at the primary extension number of a user. For a call to originate at the PCOL call appearance of a primary extension, that user must be off hook at that call appearance when the request is received.

If a Party Query ID is requested while the PCOL is alerting or on hold, one party member is reported for the group with the extension number specified as the default extension.

If a Calls Query is requested on an extension while the PCOL call is active, only one call appearance is associated with the particular call identifier.

- **Priority Calling**

  Third Party Make Call, Third Party Auto Dial, and Route Select calls can be requested as priority calls. If they are, the priority Call Coverage and alerting rules apply.

  Direct Agent calls can also have priority calling. In this case, the call follows priority call rules for delivery to the destination. That is, calls are delivered with three bursts of distinctive ringing, and do not go to the covering point for Call Coverage or Send All Calls.

- **Privacy-Manual Exclusion**

  Event Reports are not provided to the adjunct when the station user activates Privacy-Manual Exclusion causing other bridged users to be dropped from the call (go to the bridged state). Bridged users that are not allowed to bridge into an existing call because the user has activate Privacy-Manual Exclusion cannot be bridged into the call using Third Party Control capabilities (like Third Party Answer or Third Party Reconnect).

- **Queue Status Indications**

  Direct Agent calls are not included or counted in any Queue Status Indications.

- **Queuing**

  Direct Agent calls have priority over all non-Direct Agent calls in the split queue. Each Direct Agent call occupies a queue slot from the maximum length queue administered on the Hunt Group form.

- **Ringback Queuing**

  Calls originated via Third Party Make Call or Third Party Auto Dial requests to a trunk supporting Ringback Queuing or routing are not allowed to queue and a busy/reorder tone is provided to the caller.

- **Redirection On No Answer (RONA)**

  An ACD call delivered to an agent is redirected when a ‘‘no answer’’ time-out limit is reached. In this case:

  — If the call requeues to the split a Queued Event Report is generated.

  — If the call is delivered to another agent, an Alerting Event Report is generated.

  — If the call cannot be requeued, the caller continues to listen to ringback and an Event Report is not generated.

  — Starting with G3V4, when a call is redirected from an agent with RONA active, prior to the RONA timer expiry, the RONA timer is cancelled.

- **Send All Calls**

  If the destination agent has Send All Calls activated, then Direct Agent calls go to the agent’s coverage path.

  If priority calling is used, then the Direct Agent call does not go to the agent’s coverage path and remains in the queue.

  Send All Calls can be activated/deactivated either manually or via the Request Feature capability.

- Service Observing

  Service Observing may only be initiated from the observer’s voice terminal. Any type of monitored or controlled call may be service observed as long as the service observing criteria are met.

  For a switch-classified call, the observer is bridged onto the connection when the call is given to the monitored agent. The observer receives the warning tone after the bridging is complete (provided the warning tone option is administered).

  For a user-classified call, the observer is bridged onto the connection when the destination answers. When the destination is a trunk with answer supervision, the observer is bridged onto the call when actual far-end answer occurs. When the destination is a trunk without answer supervision, the observer is bridged onto the call after the trunk answer supervision time-out event.

- Single-Digit Dialing And Mixed Station Numbering

  A call routed using the Route Select capability or initiated using the Third Party Make Call or Third Party Auto Dial capabilities is permitted to use single digit dialing.

- Single-Line Voice Terminals

  If a single-line voice terminal user (with Automatic Answer) is logged into an adjunct-controlled split and the user goes on-hook, the user is logged out (regardless of the work mode).

For regular stations or ACD agents, third party call control may be used in conjunction with switch-hook flash operations. Appropriate events are reported. For simplicity, it is recommended that third party operations should not be intermixed with manual ones.

If a user-classified call is placed on behalf of a single-line voice terminal user, the user must either be off-hook (and not busy on a call), or go off-hook within five seconds of the call setup request. Otherwise, the call origination fails.

- **Subnet Trunking**

  Third Party Auto Dial Adjunct monitored or controlled calls are allowed to use Subnet Trunking.

- **Supervisor Assistance**

  This feature can be accessed in the conventional way from the voice terminal if the voice terminal is not locked. In this case, the call is placed to the switch-administered split supervisor.

  If the voice terminal is locked (under adjunct control), this feature may only be accessed via the adjunct. This feature may also be accessed via the adjunct for voice terminals that are not locked.

- **Temporary Bridged Appearances**

  The operation of this feature has not changed with CallVisor ASAI. There is no event provided when a temporary bridged appearance is created at a multifunction set. If the user is connected to the call (becomes active on such an appearance), the Connected Event Report is provided. If a user goes on hook after having been connected on such an appearance, a Disconnect/Drop Event Report is generated for the disconnected extension (bridged appearance).

  If the call is dropped from the temporary bridged appearance by someone else, a Disconnect/Drop Event Report is also provided.

  Temporary bridged appearances are not supported with analog sets. Analog sets get the Call Redirected Event Report when such an appearance would normally be created for a multifunction set.

  The call state provided to queries on extensions with temporary bridged appearances are bridged if the extension is not active on the call or connected if the extension is active on the call.

  The Third Party Selective Drop request is denied for a temporary bridged appearance which is not connected on the call.

  Starting with G3V4, temporary bridged appearances are not maintained at a redirecting extension after the call has disconnected.

- **Terminating Extension Group**

  Members of a TEG may be domain controlled. A TEG behaves similarly to bridging for the purpose of CallVisor ASAI event reporting. If controlled stations are members of a terminating group, an incoming call to the group causes an Alerting Event Report to be sent to all domain-control associations for the terminating group. For the member of the group that answers the call, a Connected Event Report is returned to the answering member domain controller(s) with the station which answered the call. All the domain controllers for the other group members (non-answering members without TEG buttons) receive a Call Redirected Event Report. When a button TEG member goes on hook but the TEG itself does not drop from the call, no event is sent but the state of that party changes from the connected state to the bridged state. The Disconnect/Drop Event Report is not sent to each member's associations until the entire TEG drops from the call as opposed to an individual member going on hook.

Members that are not connected to the call while the call is connected to another TEG member are in the bridged state. When the only connected member of the TEG transitions to the held state, the state for all members of the TEG changes to the held state even if they were previously in the bridged state. There is not event report sent to the bridged user associations for this transition.

All members of the TEG may be individually ‘‘domain controlled.’’ Each receives appropriate events as applicable to the controlled station. Third Party Call Control requests work normally if invoked over the station domain. However, Third Party Selective Hold, Third Party Merge, Third Party Reconnect, and Third Party Selective Drop are not permitted on parties in the bridged state and may also be more restrictive if the exclusion option is in effect from a station associated with the TEG.

Third Party Auto Dial or Third Party Make Call requests cannot specify the TEG group extension and the originator. TEGs can only receive calls, not originate them.

If a Party Query ID is requested while the TEG is alerting or on hold, one party member is reported for the group with the extension number specified as the TEG group extension.

If a Calls Query is requested on an extension while the TEG call is active, only one call appearance is associated with the particular call identifier.

- **Timed Reminder**

  Monitored and controlled calls (including calls originated and routed with CallVisor ASAI) extended by an attendant and not answered are redirected back to the attendant when the timed reminder interval expires. See the "Attendant Call Waiting" feature for events returned to the adjunct. Vectoring causes all other timers to be ignored.

- **Transfer**

  Manual transfer from domain-controlled station is allowed subject to the feature's restrictions. The Hold Event Report is provided as a result of the first button push. The Transfer Event Report is provided as a result of the second button push, and only if the transfer is successfully completed.

The Transfer Event Report is sent to all active associations for the resultant call. For additional information on the Transfer interactions, refer to Conference/Transfer feature interaction bullet item in this section.

- **Trunk-to-Trunk Transfer**

  When this feature is enabled, adjunct-monitored or controlled calls (except switch classified calls) transferred from trunk-to-trunk are allowed.

- **Voice Message Retrieval**

  A station user hears “Please call message for more messages'' when the message waiting indicator is activated by an adjunct using the Set Value capability. If the voice terminal display is used to retrieve messages, and if the Message Waiting Indicator is activated by an adjunct using the Set Value Capability, then the voice terminal displays: ''You have adjunct messages.''

- **VuStats**

  Concurrent operation of VuStats and ASAI-Accessed BCMS reduces the operator's capabilities of both.

- **Wait-Answer Supervision Timer (WAST)**

  Starting with G3V4, if an alerting call is successfully redirected, the WAST is cancelled.

Administration

The CallVisor ASAI feature is administered on a per-system basis. The following items require administration:

- **System Parameters/Customer Options Form**

  On this form, the CallVisor ASAI Interface option must be enabled. In addition, the following CallVisor ASAI Capability Groups (previously described in this feature description) are also optioned on the form:

  — **Adjunct Call Control Group** — Allows the adjunct to invoke the Third Party Call Control capabilities.

  — **Adjunct Routing Group** — Allows the adjunct to provide adjunct routing information to the switch for incoming calls.

  — **Domain Control Group** — Allows an adjunct to control calls and receive event reports for station sets and receive logout event reports for adjuncts in a given ACD split.

  — **Event Notification Group** — Allows the adjunct to request incoming call notification and enables the switch to send event reports about such calls.

  — **Request Feature Group** — Allows the adjunct to request features such as change work modes, login/logout, Send All Calls, and Call Forwarding.

— **Set Value Group** — Allows the adjunct to request status changes for Message Waiting lamps (that is, control the on/off state of the lamps), and billing changes for 900-type calls. Note that billing changes require that the Flexible Billing option be enabled on page 1 of the System-Parameters Customer Options form.

AT&T-ASAI Links — Allows an AT&T ASAI enabled adjunct to activate a proprietary link.

Table 3-49 shows which capabilities are automatically available when CallVisor ASAI is enabled.

Table 3-49. Automatically Enabled CallVisor ASAI Capabilities

Capability	Call Control Group	Adjunct Routing Group	Domain Control Group	Event Notif. Group	Request Feature Group	Set Value Group
Event Reports	X		X	X		
- Agent Login						
- Agent Logout			X[1]			
- Alerting	X		X	X		
- Answered	X		X			
- Busy/Unavailable	X		X	X		
- Call Ended Event (FAC message)				X		
- Call Initiated			X			
- Call Offered to Domain				X		
- Call Originated						
- Call Redirected			X	X		
- Call Conferenced	X		X	X		
- Connected	X		X	X		
- Cut-Through	X		X	X		
- Denial/Reorder	X		X	X		
- Disconnect/Drop	X		X	X		
- Hold	X		X	X		
- Queued	X		X	X		
- Reconnected	X		X	X		
- Call Transferred	X		X	X		
- Trunk Seized	X		X	X		

Table 3-49. Automatically Enabled CallVisor ASAI Capabilities — *Continued*

Capability	Call Control Group	Adjunct Routing Group	Domain Control Group	Event Notif. Group	Request Feature Group	Set Value Group
Request Feature					X	
Notification Request				X		
Notification Cancel				X		
Notification Ended				X		
Route End		X				
Route Request		X				
Route Select		X				
Set Value						X
Stop Call Notification				X		
Third Party Answer			X			
Third Party Auto Dial			X			
Third Party Call Ended	X					
Third Party Clear Call	X					
Third Party Domain Control			X			
Third Party Domain Control Ended			X			
Third Party Make Call	X					
Third Party Merge	X		X			
Third Party Reconnect	X		X			
Third Party Relinquish Control	X		X			
Third Party Selective Hold	X		X			
Third Party Selective Drop	X		X			

Table 3-49. Automatically Enabled CallVisor ASAI Capabilities — *Continued*

Capability	Call Control Group	Adjunct Routing Group	Domain Control Group	Event Notif. Group	Request Feature Group	Set Value Group
Third Party, Redirect Call	X		X			
Third Party, Send DTMF Signals	X		X			
Third Party Take Control	X					

1. Login and logout are only provided for Domain Control of ACD Split; all other event reports are provided for Domain Control of stations.

NOTE:
By enabling the CallVisor ASAI option(s), the following capabilities are automatically enabled as well, regardless of the groups selected: Value Query (includes Response Continued Capability), Abort, Heart Beat, Restart Procedure, Suspend/Resume Alarm.

- **Hunt Group Form**

  If an ACD split is to be adjunct controlled, the “Controlling Adjunct” field of this form must be administered as only available with ASAI or ‘‘asai’’ (if the CallVisor ASAI Request Feature Group option has been enabled on the System Parameters Customer Options form).

  If the “Controlling Adjunct” field is administered as **asai,** an “ASAI Link Extension” field appears. This field should contain the extension of the administered CallVisor ASAI BRI link extension (see ‘Station" form administration).

- **Call Vector Form**

  The CallVisor ASAI Link Extension must be administered for the adjunct routing step on the ‘Call Vector’ form.

- **Class of Restriction Form**

  Direct Agent Calling is allowed only if both the originator and the destination agent have a Class of Restriction (COR) that allows Direct Agent Calling.

- **Trunk Group Form**

  Trunk Groups must be administered to pass or receive CPN/BN calling party information if the ASAI application expects or requires CPN/BN information.

For an OCM application using network answer supervision, those trunks which support network answer supervision should have the “Answer Supervision” field set to ‘‘yes.’’ For trunks that do not receive real answer, the “Answer Supervision Timeout” field determines when the ASAI event "connect" is sent.

- **Station Form**

  A CallVisor ASAI endpoint must be administered with station type ‘‘ASAI’’ or ‘‘asai,’’ or ‘‘AT&T-ASAI link.’’ A maximum of four/eight (depending on system) CallVisor ASAI endpoints can be assigned. An additional station type is AT&T-adjunct.

- **Special Information Tone**

  Each “Special Information Tone (SIT)” field [there are a total of six different SITs] and one Answering Machine Detection (AMD) Treatment must be administered as either dropped or answered. When a particular SIT is detected by the call classifier circuit pack, the switch acts in accordance with the choices indicated on this form.

For more detailed information on the administration of CallVisor ASAI, see *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653 or *DEFINITY Communications System Generic 3 CallVisor ASAI Technical Reference,* 555-230-220.

Hardware and Software Requirements

The TN744 Call Classifier or the TN2182 Tone Clock is required for switch call classification and is also used by on-switch call prompting. Each port on the circuit pack acts as a touch-tone receiver or call classifier and is capable of detecting tones, including Special Information Tones.

NOTE:
This call classification only works if the public network provides similar tones to those used in the United States.

Each ASAI-BRI Interface Link requires a port on a TN556 ISDN-BRI circuit pack. Up to eight ports may be assigned for G3v and four for G3v/s/i. Each ASAI-DEFINITY LAN Gateway Link requires the ED-IE546-70 (with TN2208 and TN2170 circuit packs, tape and disk drives, and software to support this transport mechanism). There is a limit of 4 links per LAN Gateway.

The G3s and G3i systems must be equipped with the TN778 packet control circuit pack and a TN771 Maintenance-Test circuit pack.

If EPNs are present, TN570B expansion interface boards are required.

ASAI Interface software RTU (right to use) with the DEFINITY G3 generic software is required.

CDR Account Code Dialing Forced Entry of Account Codes

Feature Availability

This feature is available with all Generic 3 releases except for G3vs/G3s ABP.

NOTE:
The "Call Detail Recording (CDR)" feature was previously called the ‘‘SMDR Account Code Dialing’’ feature.

Description

Allows certain calls to be associated with a particular project or account number. This is accomplished by dialing specified account codes before making outgoing calls. This information is recorded by the CDR feature and can be used later for accounting and/or billing purposes.

Requires users to dial an account code when making certain types of outgoing calls. The conditions under which dialing of account codes is required depends on system administration. To associate an account code with a particular call, a user first dials a CDR access code. The user then dials the desired account code, which can contain up to 15 digits. The user then dials the desired trunk access code, or ARS access code.

CDR Account Code Dialing can be optional or mandatory (forced). Forced entry of account codes can be assigned for any of the following:

- Designated Toll Calls

  ‘‘Toll calls’’ are defined by the administered toll analysis table. Each ‘‘Dialed String’’ entry in the toll analysis table that is designated as a toll call can also be administered to require forced entry of account codes. If the system is administered to require forced entry of account codes, and a specific number or ‘‘Dialed String’’ is administered to require forced entry of account codes, any system user must dial an account code before dialing that number.

  This includes all calls made by ARS, or TAC.

- Toll Calls Made By Users With a Specific COR

  If forced entry of account codes is assigned to a specific COR, any voice terminal assigned that COR must dial an account code before making toll calls.

- Designated Toll Calls Made By Users With a Specific COR

  If forced entry of account codes is assigned to a specific COR, any voice terminal assigned that COR must dial an account code before making toll calls that are administered to require forced entry of account codes.

- All Calls Made on a Trunk Group With a Specific COR

  Any trunk group that is assigned a COR with forced entry of account codes cannot be accessed until an account code is dialed. If a call is being routed via ARS, account code checking is not done on the trunk group's COR.

Any time an account code is required and the user does not enter an account code, intercept tone is heard. An account code is never required for the following:

- Attendant originated call
- Busy verification of a trunk by an attendant or voice terminal user
- Distributed Communications System (unless required by the trunk group's COR)
- PCOL
- Remote Access Without Barrier Codes
- Trunk-to-Trunk Connections

Considerations

CDR Account Code Dialing provides an easy method of allocating the costs of specific calls to the correct project, department, and so on. Call information is recorded by the CDR feature for this purpose.

Account Code length can be up to 15 digits. However, not all CDR formats support 15 digits. The maximum decreases if an authorization code is dialed.

The validity of the entered account codes cannot be checked by the system.

Interactions

The following features interact with the CDR Account Code Dialing feature.

- Authorization Codes

  Authorization codes are recorded on all CDR printouts except for the 59-character, CDRU, and ISDN CDRU formats, without regard to account code length. For the non-ISDN and the ISDN LSU formats, authorization codes are recorded on CDR printouts if the account code length does not exceed six digits. For Enhanced LSU, the account code length must not exceed six digits.

- ARS

  If a trunk group is accessed via ARS, the trunk group's COR is not used to determine if an account code needs to be entered.

- Busy Verification of Terminals and Trunks

  An attendant or voice terminal user is never required to enter an account code when making a busy verification.

- Call Forwarding

  Calls cannot be forwarded to a destination where a user is required to enter an account code.

- Last Number Dialed

  The CDR access code and account code dialed are stored as part of the Last Number Dialed. However, some digits may be lost due to the limit on the number of digits stored for this feature.

- CDR

  CDR does not record the correct account code if the length of the account code is changed during an active call. For example, if the account code length is 5, a user dials 12345, and the account code length is changed during the call to 2, the CDR record shows only the first 2 digits (12) of the account code.

Administration

CDR Account Code Dialing is administered by the System Manager. The following items require administration for forced entry of account codes:

- Whether or not all toll calls require account code entry (per system)
- Whether or not each individual COR requires account code entry
- Whether or not each Dialed String in the Toll Analysis table requires account code entry

Hardware and Software Requirements

No additional hardware is required. Optional CDR Account Code Dialing software is required.

Centralized Attendant Service (CAS)

Feature Availability

This optional feature is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows services performed by attendants in a private network of switching systems to be concentrated at a central, or main, location. Each branch in a CAS has its own LDN or other type of access from the public network. Incoming trunk calls to the branch, as well as attendant-seeking voice terminal calls, are routed to the centralized attendants over RLT.

The CAS attendants are located at the main location. The main location can be a DEFINITY system Generic 1 or 3, a DEFINITY system Generic 2.1, System 85, a DIMENSION PBX, or a System 75 (V3).

The CAS main PBX operates independently of the CAS branch PBXs. The operation for CAS main PBX traffic is identical to a stand-alone PBX.

Each branch in a network with CAS is connected to the main by way of RLTs. These trunks serve three basic functions:

- Paths for sending incoming attendant seeking trunk calls at the branch to the centralized attendant to be processed and extended back to their destinations at the branch (both parts of a call use the same trunk)
- Paths for returning timed-out waiting and held calls from the branch to the main
- Paths for routing calls from voice terminals in the branch to the centralized attendant at the main

RLTs can be seized only from the branch switch and are used only for CAS calls and CAS signaling. After processing by a centralized attendant, CAS calls are extended back over the same RLT to, for example, the requested extension number or outgoing trunk. The RLT is then dropped and becomes available for other calls toward the centralized attendants.

Two queues are associated with CAS calls, one at the main and one at the branch. When idle RLTs are available from the branch to the main, RLTs are seized and CAS calls are placed in the attendant queue at the main along with other attendant-seeking calls. If all RLTs are in use, the branch switch puts calls to the attendant in a RLT trunk queue at the branch. The length of RLT trunk queue can vary from 1 to 100 and is set during administration of the RLT group. Backup service provides for all CAS calls to be sent to a backup extension in the local branch if all RLTs are maintenance busy or out of service, or if a Backup button is pressed while not lighted. The backup extension can be assigned a Backup button and associated status lamp to activate the feature and provide notification that backup service is in effect. The status lamp remains lighted as long as backup service is in effect. If the Backup button is pressed while the status lamp is lighted, calls are not be sent to the backup extension unless all RLTs are maintenance busy or out of service.

A CAS call from a branch can be put on Remote Hold by the CAS attendant. The branch holds the call and drops the RLT. After a time-out (same as the timed reminder for an attendant-held call), the branch automatically attempts to route the call back to the CAS attendant. It is possible for the returning call to be queued for the RLT. It is recommended that CAS attendants use Remote Hold when they have to put a call on hold. This keeps RLTs from being tied up unnecessarily.

The branch in a CAS network generates call identification tones and transmits them to the CAS attendant by way of the RLT. These tones indicate to the attendant the type of call coming from the branch or the status of a call extended to or held at the branch. The attendant hears these tones in the console handset prior to actually being connected to the caller. The tones may vary by country.

- Incoming trunk call: 480 Hz (100 ms), 440 Hz (100 ms), 480 Hz (100 ms) in sequence; heard immediately after attendant lifts handset
- Call from branch terminal to attendant or transferred by branch terminal to attendant: 440 Hz (100 ms), silence (100 ms), 440 Hz (100 ms) in sequence; heard immediately after attendant lifts handset
- Call extended to idle station or recall on does not answer: ringback tone for 300 ms followed by connection to normal ringing cycle
- Call extended to busy terminal — automatically waiting or recall on attendant call waiting: 440 Hz (100 ms)
- Call extended to busy terminal — waiting denied or not provided: busy tone
- Remote hold or remote hold recall: a series of four to six cycles of 440 Hz (50 ms), silence (50 ms)
- Recall on does not answer: 300 ms burst of ringback, then connection to normal ringback at any point in its cycle
- Recall from a call on remote hold: a series of four to six cycles of 440 Hz for 50 ms, silence for 50 ms
- Recall from a call waiting at a single-line terminal: 100 ms burst of 440 Hz

The centralized attendant at the main has access, through RLTs, to all outgoing trunk facilities at the branches in a CAS network. The attendant can extend an incoming LDN call to an outgoing trunk at a branch by dialing the access code and allowing the caller to dial the rest of the number or by dialing the complete outgoing number. ARS is available to the attendant in establishing outgoing calls.

Calls extended to busy single-line voice terminals at the branch wait automatically. When any waiting extended call is not answered within an administered interval, the branch switch attempts to return the call to the centralized attendant. The Call Waiting feature does not apply to multiappearance terminal; if no appearances are available, busy tone is sent to the attendant, who tells the caller that the line is busy.

Calls from voice terminals at the branch to a centralized attendant are also routed over RLTs seized by the branch switch. A branch caller reaches the attendant by dialing the attendant group access code. For G3vs/G3sV1 and G3iV1, the access code is **0**; for G3i-Global, G3rV1, G3V2, and later releases, the access code is administrable and the default is **0**. The conversation between the branch caller and the attendant ties up the seized RLT, but calls of this type are usually short.

Considerations

CAS reduces the number of attendants required at each branch location. More efficient call handling is provided by letting one group of centralized attendants handle calls for the individual branches. For example, a chain of department stores can have a centralized attendant location at the main store. The centralized attendant can then handle calls for the individual stores.

In a CAS network, DEFINITY Generic 3 switches can function as branches or as the main; the main location, where the centralized attendants reside, must be a system capable of providing attendant concentration.

A system can be a branch to only one main location.

A network with CAS can also be a DCS, but this association is not required.

A branch can have a local attendant. Access to the local attendant must be by way of an individual attendant extension. Incoming trunk calls in a CAS network may bypass local attendants but can be routed back to them by the centralized attendant.

The CAS branch calls are terminated on the CAS main PBX based on the incoming RLT trunk group day-destination or the night-service destination. A CAS call may also be answered by the Trunk Answer Any Station feature. An attendant console is not always answering/extending the incoming CAS calls. If a non-attendant answers a CAS call, the call may be extended back to the branch through use of the FLASH button on a multiappearance voice terminal or a switchhook flash on a single-line voice terminal. The branch reaction to Flash Signals and the branch application of tones is the same whether an attendant or non-attendant answers/extends the call.

If an extended call returns to the CAS main attendant because it was unanswered, the called party at the branch is not dropped but continues to be alerted until the caller is released. This allows the attendant to talk to the caller, then extend the call again, if the caller wishes, without redialing the number.

If the recall time-out occurs for an extended CAS call which has gone to Coverage and no one has answered, then the branch leaves the extended-to party ringing and drops coverage from the call.

When an analog station's call goes to coverage, the analog station is dropped from that call. This is the exception to the branch leaving the extended-to party ringing. If the CAS main attendant extends a call to an analog station and that call goes to coverage and later returns to the CAS main attendant, this call is treated as an incoming LDN call and the attendant must re-extend the call, if requested by the user.

On an incoming CAS call to the main attendant, the "Name" field from the trunk group form for that RLT is displayed to the attendant. It is recommended that the "Name" field in the trunk form provide the attendant with CAS branch identification information.

If the Music-on-Hold feature is provided in a CAS branch, it is applied to two stages of LDN calls. During the brief period in which the attendant is extending a call, the caller (who is on ''soft hold'' at the branch) receives music. Music-on-Hold is also connected to callers on Remote Hold.

Interactions

The following features interact with the Centralized Attendant Service feature.

- Attendant Control of Trunk Group Access

  If a local attendant has control of the outgoing RLT trunk group, when CAS is in effect, new attendant seeking calls are routed to the local attendant.

- Abbreviated Dialing

  The main attendant may use an Abbreviated Dialing button to extend CAS calls after obtaining branch dial tone.

- Attendant Auto-Manual Splitting

  The SPLIT lamp and button do not function on CAS main calls extended via the RLT trunk. Attendant conference does not function on CAS calls.

- Busy Indicator Buttons

  Busy Indicators can identify incoming calls over an RLT. Busy Indicators may also be used to dial after the attendant has started to extend the call.

- Call Coverage

  Calls can be redirected to a centralized attendant by Call Coverage. Calls to a CAS backup extension for backup service should not be redirected via Send All Calls to the backup extension's coverage path.

- Call Forwarding

  Calls to a CAS extension should not be forwarded.

- Call Park

  If a CAS Attendant parks a call and the call returns to the CAS attendant after the Call Park expiration interval, the CAS attendant hears incoming trunk call notification.

- DXS and DTGS Buttons

  DXS and DTGS buttons at the main attendant console can be used with CAS operation. However, when a DXS button is used to make a CAS call, it takes a few seconds before the attendant hears ringback tone.

- DCS Operation

  If the RLT trunk group is administered as a DCS trunk, the following interaction applies: On an incoming CAS call to the attendant the DCS message is displayed instead of the name of the incoming RLT trunk group. Upon answering the call, the attendant hears the call identification tones. Receipt of these tones indicates to the attendant that the call is a CAS call. In this situation, a "TRUNK-NAME" button may be used to obtain the name of the RLT trunk group.

- Emergency Access to the Attendant

  CAS Branch Emergency Access calls generated by Feature Access Code or Off-hook Alert are routed to the branch's local attendant group. If there is no attendant in the branch PBX, the emergency call is routed to the branch's administered Emergency Access Redirection Extension. When the branch PBX is in CAS Backup Service, the emergency calls are routed to the backup station and the call is treated as a normal call.

- Hunt Groups

  If an incoming CAS call is directed to a hunt group, the call does not redirect to the hunt group's coverage path. Depending on the circumstances, the attendant could get a busy tone, or it would ring until answered.

- Leave Word Calling

  If a message is left for a user on a branch switch and the attendant on the main switch tries to retrieve the message by using LWC message retrieval, permission is denied.

- Night Service — Night Console Service

  When the CAS main enters night service, CAS calls terminate at the CAS main night service destination. Calls do not go to CAS attendants when Night Service has been activated at the branch. Calls are routed to the local attendant night console, the LDN night station, or to the TAAS.

- Night Service — Trunk Answer From Any Station

  In a multiswitch DCS environment with CAS, the result of transferring incoming trunk calls via the Night Service Extension or the Trunk Answer From Any Station feature varies depending on the home switch of the transferred-to station, the home switch of the connected trunk, and the type of night service function chosen (Night Service Extension, Trunk Answer From Any Station, or both).

- Non-Attendant Console Handling of CAS Calls

  The CAS branch calls are terminated on the CAS main PBX based on the incoming RLT trunk group day-destination or the night-service destination. A CAS call may also be answered by the Trunk Answer Any Station feature.

  Normally, a non-attendant extends a CAS call by using the Flash button. However, if the non-attendant does not have a Flash button, the call can be extended as follows:

  — Multiappearance voice terminal users can extend a CAS call by pressing the Conference or Transfer button and then dialing the extension of the party the call is being extended to. To complete the call, the user must then drop the call. To drop the extended-to party, the user must press the Conference or Transfer button again.

  — Single-line voice terminal users can extend a CAS call by flashing the switchhook and then dialing the extension of the party the call is being extended to. To complete the call, the user must then drop the call. To drop the extended-to party, the user must flash the switchhook again.

- Non-Attendant Console Releases Call

  The non-attendant can drop the RLT by going on-hook, using the DISCONNECT or DROP button, or by selecting another call appearance.

- Non-Attendant Console Holds Call

  A multifunction nonattendant may hold a CAS call by pressing the hold button.

- Non-Attendant — Display Trunk Name

  If the nonattendant with a display presses the ‘‘TRUNK-NAME’’ button while active on a trunk call, then the PBX displays the “Name” field from the trunk group form.

- CDR

  If the CAS main RLT trunk has the CDR option selected, CDR records are generated for the incoming CAS calls.

- Timed Reminder

  The timer value used for recalling held calls at the attendant console is a parameter that can be set on the console form.

If an attendant at the main location transfers a call from a branch location to an extension at the main location, the timed reminder does not apply and the call does not return to the attendant if unanswered. If a branch call is unanswered, the branch timed reminder times out and the call is routed to a new RLT trunk and back to a CAS main attendant.

- Trunk-Name Button

  The trunk-name button can be used when an outgoing call has been made over a trunk which has been administered to have no outgoing display.

Administration

CAS is administered by the System Manager. The following items require administration:

- Access to CAS (branch, main, or none)
- Branch attendant individual extension number
- RLT group (outgoing for branch) (incoming for main)
- RLT group queue length (this must be greater than 0)
- CAS backup extension
- CAS Backup buttons (used to activate/deactivate CAS Backup)
- Extension permitted to put system into night service if the system has no local attendant
- Recall time-out values: Held Call or Return Call Timers
- Remote hold access code
- Trunk-Name Button, if DCS is provided
- Flash Button for a multiappearance voice terminal if a nonattendant answers CAS calls

Hardware and Software Requirements

Requires a TN760B Tie Trunk circuit pack (TN760D supports A-law). The TN760B serves all other tie trunk applications in addition to CAS. As an alternative, the TN722 DS1 Tie Trunk or TN767 DS1 Interface circuit packs (TN464B/C/D support A-law) can be used for the release link trunks of the CAS network.

NOTE:
The TN464C and later revision DS1 Interface circuit packs provide a DSX1 interface.

CAS software is required.

Class of Restriction (COR)

Feature Availability

This feature is available with all Generic 3 releases.

Description

Defines different classes of call origination and termination privileges. Systems may have only a single COR, one with no restrictions, or may have as many CORs as necessary to effect the desired restrictions.

A COR is assigned to each of the following:

- Attendant consoles (as a group)
- Authorization Code
- ACD split
- Code Calling Access zone
- Data module
- DDC group
- Individual Attendant Consoles
- Loudspeaker Paging Access zone
- Remote Access barrier code
- Terminating Extension Group
- Trunk group
- UCD group
- Voice terminal
- VDN

Use of CORs can be categorized as follows:

- Calling party restrictions
- Called party restrictions
- VDN of Origin Announcement restrictions
- Forced entry of account codes
- Partitioned Group Number
- Service Observing
- Priority Queuing
- Time of Day Plan Number

- Direct Agent Calling
- Facility Access Trunk Test
- Fully Restricted Service (G3i-Global, G3rV1, and G3vsV1/G3sV2)
- Restriction Override (G3i-Global, G3rV1, and G3V2)
- Restricted Call List
- Unrestricted Call List
- Miscellaneous restriction groups
- Selective denial of public network calling through a CCSA or EPSCS network
- An ARS or AAR FRL for control of call routing

Features assignable as calling party restrictions are as follows:

- Origination Restriction
- Outward Restriction
- All-Toll Restriction
- TAC-Toll Restriction

Features assignable as called party restrictions are as follows:

- Inward Restriction
- Manual Terminating Line Restriction
- Termination Restriction
- Public Restricted (G3i-Global and G3V2)

Use of CORs

CORs can be used to assign a variety of restrictions to a variety of facilities. The types of restrictions which can be assigned are discussed in the following paragraphs.

Calling Party and Called Party Restrictions

Calling party restrictions prevent specified users from placing certain calls or accessing certain features. Features assignable as calling party restrictions are Code Restriction, Origination Restriction, Outward Restriction, and Toll Restriction. These individual features are fully described elsewhere in this chapter. A brief description is given here:

- Outward Restriction — Prevents callers at specified voice terminals from accessing the Public Network Access feature. Calls can be placed to other voice terminal users, to an attendant, and to tie trunks.
- Origination Restriction — Prevents callers at specified voice terminals from originating calls. Voice terminal users can, however, receive calls.

- TAC-Toll Restriction — Prevents callers at specified voice terminals from making trunk access calls to certain toll areas as defined on the system's administered Toll Analysis form, unless the number is on an Unrestricted Call List associated with the caller's COR. This restriction applies to calls made using trunk access codes of CO or FX trunk groups. See the "Restriction — Toll" feature for more details.
- All-Toll Restriction — This restriction is identical to the TAC-Toll Restriction described above, except this restriction also applies to ARS calls. See the "Restriction — Toll" feature for more details.

Called party restrictions prevent specified users from receiving certain calls. Features assignable as called party restrictions are Inward Restriction, Manual Terminating Line Restriction, Termination Restriction, and Public Restriction. These individual features are fully described elsewhere in this chapter. A brief description is given here:

- Inward Restriction — Restricts users at specified voice terminals from receiving public network, attendant-originated, and attendant-extended calls.
- Manual Terminating Line Restriction — Restricts users at specified voice terminals from receiving calls other than those from an attendant.
- Termination Restriction — Restricts users of specified voice terminals from receiving any calls.
- Public Restriction — Restricts users of specified voice terminals from receiving direct public network calls.

Looking at the screen form used to administer CORs, the "Calling Party Restriction" and the "Called Party Restriction" fields are both administered as none. However, the "Calling Party Restriction" field could be administered as any of the other previously described calling party restrictions. Likewise, the "Called Party Restriction" field could be administered as any of the other previously described called party restrictions. Including ''none'' as a choice of restrictions, as many as 20 combinations of calling and called party restrictions are possible. However, it is unlikely that all 20 combinations are needed in any one situation. Therefore, only the required ones should be established.

Calling and called party restrictions are the basis for all CORs. In cases where no restrictions are needed, a single COR could be assigned with calling and called party restrictions of ''none.'' This same COR could be used for unrestricted voice terminals, trunk groups, terminating extension groups, UCD groups, DDC groups, data modules, the attendant group, and individual attendant extensions.

The following are typical examples of calling and called party restrictions which may be assigned to a COR:

- Long-distance calling is limited by All-Toll Restriction, but no restrictions are placed on incoming calls.
  - Calling party restriction=All-Toll

- — Called party restriction=None

- A voice terminal in a storeroom should not be used for outside calling. Also, all incoming calls should be from internal callers.
  - — Calling party restriction=Outward
  - — Called party restriction=Inward

- A voice terminal in a certain department cannot be used for outside calling. Incoming calls must be from the attendant (assuming that department cannot be dialed directly from the outside).
  - — Calling party restriction=Outward
  - — Inward party restriction=Manual Terminating Line

- Certain voice terminals are to be included in a UCD group for answering business calls only. These terminals are not to be used individually.
  - — Calling party restriction=Origination
  - — Called party restriction=Termination

The called party restriction is checked only at the called terminal, module, attendant console, zone, or group. For example, if a call redirects from one voice terminal to another, as through the Call Coverage feature, the called party restriction of the called (redirected from) voice terminal is the only one checked.

Each COR is established as needed and is arbitrarily identified by a number. For example, if the COR for the storeroom is 12, the storeroom voice terminal(s) is assigned COR 12.

Forced Entry of Account Codes

Account Codes are used to associate calling information with specific projects or account numbers. This is accomplished by dialing a specific account code before making an outgoing call. Account code dialing can be optional or mandatory (forced) on a per-COR or system-wide basis.

Looking at the screen used to administer CORs (the screen above) the “Forced Entry of Account Codes” field is preset as **n**. This means that account code dialing is optional. A **y** in the field would indicate that account code dialing is mandatory for users placing designated calls that have the CDR Forced Entry of Account Codes activated on the “Toll Analysis” form and have a COR with a **y** in the field assigned to that user.

If this field is **y** and the COR is assigned to a trunk group, all calls made using the TAC of that trunk group require an account code, no matter what number is dialed.

If this field is **n**, account dialing is still required if the system-wide parameter is set to **y** on the ‘‘CDR System Parameters’’ form.

Partitioned Group Number (PGN)

When AAR and ARS services are to be partitioned among different groups of users within a single system, all users in a specific group must share the same PGN. A PGN is assigned to each COR. The PGN is not a restriction, but a means used to indicate the choice of route tables to be used on a particular call.

If the Time of Day Routing feature is assigned, this field is replaced with a “Time of Day Plan Number” field. This field is used to assign each COR one of eight Time of Day Routing Plans to be used when routing AAR and ARS calls. Time of Day Routing provides the most economical routing of AAR and ARS calls based on the time of day or week that each call is made. For more information on Time of Day Routing, see the "Time of Day Routing" feature elsewhere in this chapter.

Service Observing

Service Observing allows a specified user, such as a split supervisor, to observe a call that involves other users while the call is in progress. For a user to observe calls, they must be assigned a COR that allows them to be a service observer. For someone to have their calls observed, they must be assigned a COR that allows them to be service observed. If the user wishes to observe VDNs or use vector initiated service observing, VDNs must also be assigned appropriate CORS.

When activation of Service Observing occurs through a sequence of multiple entities, for example several VDNs, Remote Access Barrier Codes, or Authorization Codes, the COR of the last entity is used to determine observer permissions. When Service Observing is activated by Remote Access Barrier Code or Authorization Code, the code must have a COR that allows the user to be a service observer.

For more information, see the "Service Observing" feature description elsewhere in this chapter.

VDN of Origin Announcement

VDN of Origin Announcement (VOA) is a feature that provides a brief message for agents to know a caller’s city of origin or requested service, based on the vector processing that directed the call to the agent. VOA is COR-dependent. Users who have a COR that allows VOA can receive the VDN of Origin messages. Users who have a COR that denies VOA are not able to receive the VDN of Origin messages.

Priority Queuing

Priority Queuing allows calls with increased priority to be queued (at hunt groups) ahead of calls with normal priority. If a COR is administered as having Priority Queuing, calls made by a user with that COR are queued ahead of non-priority calls and are answered sooner. VDN of Origin Announcement is only available with G3V3 and later releases.

For intraflowed calls from one hunt group to another to be given priority queuing treatment, ACD software must be activated.

DAC

The “Direct Agent Calling” field on the COR form indicates whether the user can originate and receive Direct Agent calls through an adjunct. If either the originating or the destination party of a Direct Agent call does not have the proper COR, the call is denied.

Direct Agent Calling allows an adjunct to transfer a call to a particular agent and have the call treated as a call to a split. For more information on DAC, see the "Automatic Call Distribution (ACD)" and "Inbound Call Management (ICM)" features elsewhere in this chapter.

Facility Access Trunk Test

This field on the COR form is used to grant a user permission to make Facility Test Calls to access trunks. A **y** in this field allows users with this COR to make these calls. An **n** in this field causes users with this COR to receive intercept treatment when they attempt to make these calls. For more information on Facility Test Calls, see the "Facility Test Calls (with Security Measures)" feature elsewhere in this chapter.

Fully Restricted Service (G3i-Global, G3rV1, and G3V2 and later)

Denies the specified voice terminal access to public network trunks for either incoming or outgoing completion. Note that Restriction Override should be set to “none.” See below.

Restriction Override (3-way COR calling)

The Restriction Override feature (available only with G3i-Global and G3V2 and later releases) determines whether or not there is a 3-way COR check made on Conference and Transfer calls. If a Conference or Transfer is denied because of the 3-way COR check, a further check is made to see if Restriction Override is possible. The COR causing the denial is checked to see if Restriction Override is assigned to be other than “none.” If the override type is “attendant” or “all,” the third (or controlling) party is checked to see if it belongs to the restricition override class of *user.* If so, the call is allowed. If the controlling party is not of the matching type or the restriction override is set to “none,” the call is denied.

Restricted Call List

A Restricted Call List is assigned to the system by the System Manager. This call list is made up of specific digit strings which cannot be dialed from facilities that are restricted by the call list. The “Restricted Call List” field on the COR form is used to determine which facilities are restricted by this call list. If a COR has this field administered as **y**, facilities with that COR cannot be used to dial a number that matches one of the digit strings in the Restricted Call List.

Unrestricted Call List

Ten Unrestricted Call Lists are assigned to the system by the System Manager. Each Unrestricted Call List is made up of specific digit strings which can be dialed from facilities associated with the call list. The “Unrestricted Call List” fields on the “COR” form are used to determine which facilities have access to each of the ten call lists. If a COR has any of these ten fields administered with an Unrestricted Call List number, facilities with that COR can call any numbers contained in the assigned list(s) (unless the number is included in the Restricted Call List, which overrides the Unrestricted Call List).

Selective Denial of Public Network Calling Through a CCSA or EPSCS Network (APLT)

Public network calling via the private CCSA or EPSCS network (commonly referred to as off-network calling) is optional on a per-private network basis. If off-network calling is not provided, then the “APLT” field can be ignored. If off-network calling is provided, then permission or denial to access the off-network capability is set via the “APLT” field. Users assigned a COR that has APLT set to **n** (no) can use off-network calling. Users assigned a COR that has APLT set to **y** (yes) cannot. If there is a need for both yes and no choices in a system, separate CORs must be assigned to reflect this.

Looking at the screen used to administer CORs, the “APLT” field is preset as **y**. This means that a facility with this COR is not allowed to access CCSA or EPSCS off-network capabilities for public network calling. An **n** in this field would indicate that the facility can access CCSA or EPSCS off-network capabilities.

ARS/AAR FRL for Control of Call Routing

If the system does not use AAR or ARS to determine the most preferred routing of calls, then the “FRL” field can be ignored. If AAR or ARS is used, then an FRL is used to either allow or deny access to certain routes. The FRL for the outgoing (trunk) side of the call is provided in the AAR/ARS Routing Pattern. Although each outgoing trunk group has a COR and each COR has an FRL, this FRL is not used unless the trunk group is the originator of the call. Call routing is determined by a comparison of the FRLs in the AAR/ARS Routing Pattern and the FRL in the COR of the call originator (typically, a voice terminal user).

The “FRL” field is preset to **7**. However, this field can have a value of 0 through 7. An originating FRL of 0 has the least calling privileges, whereas an originating FRL of 7 has the most calling privileges. Each route (For number of routes, see, Appendix A, "System Parameters") in each of the ARS/AAR Routing Patterns also has an FRL. These route FRLs can also have a value of 0 through 7. A route FRL of 0 is the least restrictive, whereas a route FRL of 7 is the most restrictive. An FRL of 0 is checked before the other routes in a given ARS routing pattern. To access a route, the originating FRL must be greater than or equal to the route FRL. Determination of appropriate FRL values must be made with respect to the outgoing routes from a specific system and the desired levels of calling privileges. This is part of AAR/ARS customization. The FRL of the call originator is contained in the COR assigned. The “FRL” field in a COR assigned to an outgoing trunk group is never checked and should be ignored.

Assuming AAR and/or ARS has been customized for a system, the System Manager must establish unique CORs for each of the up to eight levels of ARS calling privileges that is used in the system. However, these CORs must maintain the desired restrictions dictated by the other fields on the screen form. The simplest case is a COR specifying no restriction. Ordinarily, this COR can be assigned to all unrestricted users. However, if some subset(s) of these users requires different FRLs, separate CORs must be established for each different FRL required.

For a detailed description of "Automatic Alternate Routing (AAR)", "Automatic Route Selection (ARS)", and "Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs)", refer to the individual feature descriptions given elsewhere in this chapter.

Miscellaneous Restriction Groups

Miscellaneous Trunk and Miscellaneous Terminal Restriction groups restrict access to a terminal, module, zone, attendant console, or group. This is accomplished via the COR assigned to the calling and the called facilities. When a COR is administered, access by that COR to each of the CORs is either allowed or denied. Since a given COR can be assigned to both calling and called facilities, calling to one’s own COR can be restricted. This is fully explained in the following paragraphs.

The simplest way to understand miscellaneous restrictions is to look at the screen used during implementation. When a COR is established, the assigned number is entered in the “COR Number” field. If this COR is assigned to a facility that originates a call, such as a voice terminal, the calls to CORs associated with terminating facilities can be prohibited. The Miscellaneous Restriction group information is found in the “Calling Permission” field. A “y” entry in this field indicates that the COR specified at the top of the form can call the COR numbers that contain a “y.” If an “n” is entered, the specified COR cannot be called by the COR number at the top of the form. On the screen form no restrictions apply because all CORs are specified as “y.”

Miscellaneous restrictions are checked on the initial call termination and on redirection.

Miscellaneous Restriction groups apply on a per-COR basis. However, the same COR can be assigned to more than one facility. Facilities with the same COR may be like facilities (such as two voice terminals) or different facilities (such as a voice terminal and a trunk group). In either case, the same restrictions apply to both facilities.

Certain facilities, such as voice terminals, can originate and receive calls. Call origination and termination restrictions are specified via a single COR. Miscellaneous Restrictions can prevent calling any COR, including one's own COR.

When a COR is administered, the allowance or denial of access from that COR to each of the CORs applies only to the COR being administered. For example, if COR 6 is administered with access denied to COR 3, this only specifies that COR 3 cannot receive a call from COR 6. Whether or not COR 3 can be accessed by any other COR (for example, COR 7) is determined when that COR (COR 7) is administered. From this, it follows that a single COR cannot be used to provide both unrestricted service and miscellaneous restrictions.

COR Examples

The examples given here are designed to help in the understanding of CORs and to illustrate some of the practical aspects of CORs. These are, however, only examples. In reality, each system must be administered to meet its individual needs.

Example Using Miscellaneous Restrictions

As an illustration of miscellaneous restrictions, assume a system installation provides the following:

- Central office trunks
- WATS
- FX trunks
- Data modules
- Attendant service
- Voice terminals
- DID trunks
- Remote Access

In an unrestricted environment, each of the above facilities could have the same COR. However, suppose the following requirements exist:

- Attendants cannot make data calls.
- Remote Access can be used for data calls only.
- DID cannot be used for data calls except through Remote Access

  (A dedicated Remote Access trunk group is not required, although one or more could be provided. This example assumes all Remote Access is via DID.)
- There are three classes of voice terminals:
  - Those that can call anywhere, any time.

— Those that can place local central office and in-house calls only.

— Those that can place local central office, FX, and in-house calls only.

To implement the above requirements, a COR must be assigned to each facility or group of facilities. For simplicity, each can have a unique COR. The CORs are arbitrarily assigned as follows:

- COR 30 — Local central office trunks
- COR 31 — WATS trunks
- COR 32 — FX trunks
- COR 33 — Data modules
- COR 34 — Attendant group
- COR 35 — Unrestricted voice terminals
- COR 36 — Voice terminals that can place in-house and local central office calls only (no FX or WATS calls)
- COR 37 — Voice terminals that can place in-house, local central office, and FX calls only (no WATS calls)
- COR 38 — DID trunk group
- COR 39 — One of the remote access barrier codes (can be up to 10)

With the CORs defined, it should be individually determined which CORs cannot call other CORs. This is done as follows:

- COR 30 (local CO trunks) — No restrictions were specified for these trunks. The default values on the screen form are sufficient. No action is required, except to specify a COR number of 30.
- COR 31 (WATS) — CORs that cannot use WATS are specified as they are encountered. WATS itself is an outgoing service without any calling capabilities. Thus, Miscellaneous Restrictions are not specified on this form. The Calling Party Restriction should be ‘‘none’’ (although this restriction does not really have any meaning for an outgoing facility). Similarly, the Called Party Restriction applies to facilities capable of answering a call. Since this is not the case with WATS, ‘‘none’’ should be specified. Again, the default values are sufficient, so only the COR number needs to be specified.
- COR 32 (FX) — According to the requirements for this example, no restrictions apply. Reasons are the same as for WATS. Only the COR number needs to be specified.
- COR 33 (data modules) — No restrictions apply for reasons similar to the reasons why no restrictions were assigned for WATS. Only the COR number needs to be specified.

- COR 34 (attendant group) — The attendant group cannot call COR 33 (data modules). Specify an n beside COR 33 in the “CALLING PERMISSION” field. Specify 34 in the “COR Number” field.
- COR 35 (unrestricted voice terminals) — Since no restrictions were specified, only the COR number needs to be entered.
- COR 36 (no FX or WATS calls) — This COR cannot call COR 32 (FX) or COR 31 (WATS). Specify an n beside CORs 32 and 31 in the “CALLING PERMISSION” field. Specify 36 in the “COR Number” field.
- COR 37 (no WATS calls) — This COR cannot call COR 31 (WATS). Specify an n beside COR 31 in the “CALLING PERMISSION” field. Specify 37 in the “COR Number” field.
- COR 38 (DID) — This COR cannot call COR 33 (data modules). Specify n beside COR 33 in the “CALLING PERMISSION” field. Enter 38 in the “COR Number” field.
- COR 39 (Remote Access barrier code) — This COR can be used for data calls only. Thus, this COR can call COR 33, but not CORs 30 (local central office), 31 (WATS), 32 (FX), 34 (attendant group), 35, 36, or 37 (voice terminals). Specify an n beside CORs 30, 31, 32, 34, 35, 36, and 37 in the “CALLING PERMISSION” field. Enter 39 in the “COR Number” field. (The CORs listed in the “CALLING PERMISSION” field can be viewed as terminating or screening CORs that can or cannot be called by the originating COR. Since COR 38 [DID] is neither a terminating nor a screening COR, it does not have to be considered when assigning the barrier code COR.)

Example Using Calling Party Restrictions, Called Party Restrictions, and Miscellaneous Restrictions

To illustrate the use of both Calling and Called Party restrictions, and Miscellaneous restrictions, assume a system installation provides the following:

- Central office trunks (outgoing)
- WATS
- FX trunks (outgoing)
- Voice terminals
- Data modules
- Terminating Extension Groups
- Loudspeaker Paging

Suppose that the following requirements exist:

- Only the attendant can access loudspeaker paging.
- Terminating Extension Groups can only accept calls from internal voice terminals.
- There are six classes of voice terminals:

— Those that are toll restricted

— Those that cannot call outside to a public network (outward restricted)

— Those that can receive calls only from an attendant

— Those that can call anywhere, any time

— Those that cannot place FX or WATS calls

— Those that cannot place WATS calls

To implement the above requirements, a COR must be assigned to each facility or group of facilities. For simplicity, each can have a unique COR. The CORs are arbitrarily assigned as follows:

- COR 40 — Local central office trunks
- COR 41 — WATS trunks
- COR 42 — FX trunks
- COR 43 — Attendant group
- COR 44 — Data modules
- COR 45 — Terminating Extension Groups
- COR 46 — Loudspeaker Paging Access Zones
- COR 47 — Unrestricted voice terminals
- COR 48 — Voice terminals that are toll restricted
- COR 49 — Voice terminals that are outward restricted
- COR 50 — Voice terminals that can only receive calls from an attendant
- COR 51 — Voice terminals that cannot place FX or WATS calls
- COR 52 — Voice terminals that cannot place WATS calls

With the CORs defined, it should be determined individually which CORs cannot call other CORs. This is done as follows:

- COR 40 (local CO trunks) — Restrictions that prohibit access to this COR are assigned when the originating CORs are considered. Only the COR number has to be specified on this form.
- COR 41 (WATS) — This is the same case as described in the previous configuration example. Only the COR number needs to be specified.
- COR 42 (FX) — Again, only the COR number needs to be specified.
- COR 43 (attendant group) — No restrictions were stated, so only the COR number needs to be specified.
- COR 44 (data modules) — No restrictions were stated, so only the COR number needs to be specified.

- COR 45 (TEG) — This COR can receive internal voice terminal-originated calls only. Since no tie trunks are specified for this example, the Inward Restriction feature can provide the desired restriction. Specify ‘‘inward’’ as the Called Party Restriction. If dial repeating tie trunks are provided, Miscellaneous Restrictions could be used to deny trunk access to the group. Also, specify 45 as the COR number.
- COR 46 (Loudspeaker Paging Access zones) — Since this COR can be accessed by an attendant only, the Manual Terminating Line feature can provide the restriction. Specify ‘‘manual’’ as the Called Party Restriction. Specify 46 as the COR number.
- COR 47 (unrestricted voice terminals) — No restrictions were stated, so only the COR number needs to be specified.
- COR 48 (toll restricted voice terminals) — Specify ‘‘tac-toll’’ as the Calling Party Restriction. Specify 48 as the COR number.
- COR 49 (outward restricted voice terminals) — Specify ‘‘outward’’ as the Calling Party Restriction. Specify 49 as the COR number.
- COR 50 (voice terminals that can only receive calls from an attendant) — Specify ‘‘manual’’ as the Called Party Restriction. Specify 50 as the COR number.
- COR 51 (voice terminals that cannot place WATS or FX calls) — None of the Calling Party Restrictions uniquely prohibit WATS and FX calls, so Miscellaneous Restrictions are used. Enter an n beside COR 41 (WATS) and COR 42 (FX) in the “CALLING PERMISSION” field. Leave the “Calling Party Restriction” field as none and specify 51 as the COR number.
- COR 52 (voice terminals that cannot place WATS calls) — Enter an n beside COR 41 (WATS) in the “CALLING PERMISSION” field. Leave the “Calling Party Restriction” field as none and specify 52 as the COR number.

Another method to determine COR assignment is to consider the restrictions to be assigned. The requirements given for this example were as follows:

- Only the attendant can access loudspeaker paging.
- Terminating Extension Groups can only accept calls from internal voice terminals.
- The six classes of voice terminals are:
  - Those that are toll restricted
  - Those that cannot call outside to a public network (outward restricted)
  - Those that can receive calls only from an attendant
  - Those that can call anywhere, any time
  - Those that cannot place FX or WATS calls
  - Those that cannot place WATS calls

Assignments for these requirements could be made as follows:

- COR 20 — Manual Terminating Line Restriction
- COR 21 — Inward Restriction
- COR 22 — Toll Restriction
- COR 23 — Outward Restriction

  NOTE:
  A new Manual Terminating Line Restriction for voice terminals was not established. COR 20, above, can be assigned.

- COR 24 — Unrestricted
- COR 25 — COR for WATS
- COR 26 — COR for FX
- COR 27 — Provides Miscellaneous Restrictions for WATS and FX. Enter an n beside COR 25 and COR 26 on the form for COR 27.
- COR 28 — Provides Miscellaneous Restriction for WATS. Enter an n beside COR 25 on the form for COR 28.

Now assign the appropriate COR to each physical or screening facility:

- Central office trunks — COR 24 (unrestricted)
- WATS — COR 25 (WATS COR)
- FX — COR 26 (FX COR)
- Attendant group — COR 24 (unrestricted)
- Voice terminals — COR 22 (toll), COR 23 (outward), COR 20 (manual), COR 24 (unrestricted), COR 27 (WATS and FX miscellaneous), or COR 28 (WATS miscellaneous), as required
- Data Modules — COR 24 (unrestricted)
- Terminating Extension Group — COR 21 (inward)
- Loudspeaker Paging trunks — COR 20 (manual)

This latter method is probably more difficult to use, but it minimizes the number of CORs established. This method required 9 CORs to effect the same restrictions as 13 CORs with the previous method.

Considerations

COR provides the means to consolidate assignment and administration of the various restriction features available with the system.

All items associated with a COR are distinct and separate. A unique COR must exist for each needed combination of FRLs, CCSA/EPSCS off-network restrictions, calling party restrictions, called party restrictions, and miscellaneous restrictions. CORs can be established, as required, to provide the needed combinations.

Interactions

The following features interact with the Class of Restriction feature.

- AAR/ARS Partitioning

  Partition Group Numbers are assigned via a COR.

- AAR/ARS

  Originating FRLs are assigned via a COR. Termination and Miscellaneous Restrictions do not apply to ARS/AAR calls.

- Bridged Call Appearance

  The COR assigned to a voice terminal's primary extension also applies to calls originated from a bridged call appearance of that extension on another terminal.

- Call Coverage

  Users who may normally be restricted from calls can still receive calls directed to them via Call Coverage.

  When a call goes to coverage, it is the called party's (not the covering party's) restrictions that are used.

- Call Forwarding All Calls

  If a call would normally be restricted between the forwarding and forwarded-to extensions, Call Forwarding activation is denied. Restrictions are always checked when Call Forwarding is activated, but not when a call is actually forwarded.

- Controlled Restriction

  Restrictions assigned via the Controlled Restriction feature override the calling and called party restrictions via a COR. Activate and deactivate controlled restrictions for another extension or group of extensions.

- Emergency Access to Attendant

  Emergency Access to Attendant calls are not restricted by COR.

- Forced Entry of Account Code

  This feature can be assigned via a COR.

- Inward Restriction

  This feature is assigned via a COR.

- Manual Terminating Line Restriction

  This feature is assigned via a COR.

- Origination Restriction

  This feature is assigned via a COR.

- Outward Restriction

  This feature is assigned via a COR.

- Private Network Access and Public Network Access

  Access to the public network via the private network is allowed or denied via a COR (assuming the private network provides the capability to access the public network).

- Termination Restriction

  This feature is assigned via a COR.

- All-Toll Restriction, TAC-Toll Restriction

  This feature is assigned to an originating facility via a COR. (Toll Restriction is assigned to an outgoing trunk group on the trunk group form.) TAC-toll restriction can be disabled for specific outgoing trunk groups on the trunk group form.

Administration

COR is administered by the System Manager. For each COR which is assigned, the following items must be administered:

- Access to Malicious Call Trace
- COR Number
- FRL
- Permission to access EPSCS or CCSA off-net facilities
- Calling Party Restriction
- Called Party Restriction
- Permission to call other CORs
- Forced Entry of account codes for CDR (yes or no)
- Partitioned Group Number
- Priority Queuing (yes or no)
- Can Be Service Observed (yes or no)
- Can Be A Service Observer (yes or no)
- Time of Day Plan Number
- Direct Agent Calling
- Facility Access Trunk Test
- Fully Restricted Service
- Restricted Call List
- Unrestricted Call List

Assignment of Restrictions

A COR is assigned to each of the following:

Voice Terminals

All voice terminals must be assigned a COR. The same COR may be assigned to all voice terminals or a unique COR may be assigned to a particular voice terminal or group of voice terminals. This COR applies individually to each voice terminal and is independent of all other COR applications, such as Miscellaneous Restriction groups or UCD groups.

The main items of concern for individual voice terminals are calling party restrictions and called party restrictions (discussed previously under ‘‘Use of CORs’’). If no restrictions are needed for a certain group of voice terminals, ‘‘none’’ can be specified for both calling party and called party restrictions. If it is desired to restrict a group of voice terminals from making outside calls, a COR specifying a calling party restriction of ‘‘outward’’ should be established.

Additionally, miscellaneous restrictions, restrictions to CCSA and EPSCS off-network calling capabilities, and FRLs also apply. A separate COR must be established for each unique set of restrictions.

Trunk Groups

Each trunk group is assigned a COR. Trunk groups are assigned CORs mainly for the use of miscellaneous restrictions.

Calling party and called party restrictions should be ‘‘none.’’ Whether or not a CO or FX trunk group is restricted is specified on the trunk group form used during implementation.

CO and FX trunk groups default to being toll restricted for TAC calls. Toll Restriction for TAC calls can be disabled for certain CO/FX trunk groups on the trunk form.

Attendant Consoles (as a group) and Individual Attendant Extensions

Attendants are normally allowed full access to the system’s capabilities. Therefore, calling and called party restrictions are usually set to ***none***. Also, access to the attendant is normally allowed to all CORs. This is accomplished via a ***y*** (yes) for the attendant’s COR in the “CALLING PERMISSION” field on the screen form for each assigned COR.

Tenant restriction are usually assigned to permit:

— Attendant and served tenants to call each other

— Attendant to call trunks, etc. that served tenants can access.

Data Module, Loudspeaker Paging Access Zone, Code Calling Access Zone, and Remote Access Barrier Code

Each data module, Loudspeaker Paging Access zone, Code Calling Access zone, and Remote Access barrier code is assigned a COR. Through Miscellaneous Restriction groups certain users are allowed access to certain facilities, while other users are denied access. For example, if a Loudspeaker Paging Access zone has a COR of 3, then a voice terminal with a COR marked Calling Restriction = N for COR 3 cannot access that Loudspeaker Paging Access zone.

Terminating Extension Group, Automatic Call Distribution Split, Uniform Call Distribution Group, and Direct Department Calling Group

These groups are set up to receive calls. A COR is assigned to each group. This COR is distinct and separate from CORs assigned to the individual group members. The group COR allows or denies calls to the group. Since Miscellaneous Restriction groups are normally used to restrict calling, called party restrictions should be specified as ‘‘none.’’ Since a group cannot originate a call, calling party restrictions do not apply. However, for simplicity, ‘‘none’’ is normally specified. For calls by group members or calls to individual group members, the COR assigned to the voice terminal applies. The group COR has no effect on calls directly to or from a group member.

The important aspect of these CORs is that they allow the called party restrictions of the group (normally none) to be different from the called party restrictions of the individual group members (Inward, Manual Terminating Line, or Termination).

Hardware and Software Requirements

No additional hardware or software is required.

Class of Service (COS)

Feature Availability

Class of Service is available with all Generic 3 releases.

Description

Defines whether or not voice terminal users may access the following features and functions:

- Automatic Callback
- Call Forwarding
- Data Privacy
- Priority Calling
- Off-Hook Alert
- Console Permission
- Client Room

There are only two choices for each feature; a voice terminal user or individual attendant *can* or *cannot* access the feature.

There are 16 possible COSs. Each COS is used to allow or deny access to seven features and functions. The parameters can be changed to meet individual COS needs. To assign a COS, administer the desired allowed/denied combination of features and functions for one of the 16 COSs, and indicate that COS number when implementing voice terminals. Which COS numbers represent which combination of allowed/denied features are given in the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

In addition to Automatic Callback, Call Forwarding, Data Privacy, and Priority Calling, the DEFINITY system Generic 3 offers the following functions:

- Off-Hook Alert

  Can be administered only if the optional Emergency Access to the Attendant feature is provided. The Off-Hook Alert function lets the customer administer yes/no to each of the 16 established COS parameters according to the allowed/denied capability to access this feature.

- Console Permission

  Allows multi-appearance voice terminal users to control the same features the attendant controls. This feature is usually available to front desk personnel in a hotel/motel. With console permission, you can do the following:

  — Activate Automatic Wakeup for another extension.

  — Activate and deactivate controlled restrictions for another extension or group of extensions.

  — Activate and deactivate Do Not Disturb for another extension or group of extensions.

  — Activate Call Forwarding for another extension.

- Client Room

  This function can be administered when Hospitality Services are provided. This function allows the Check-in, Check-out, Room Change/Swap, and Maid Status features. In addition, it is required at consoles or terminals that are to receive Message Waiting Notification.

Other than to allow/deny access to the described features, COS has no other use in the system. Restriction groups and call origination/reception privileges are defined and assigned by a COR, not a COS.

Considerations

COS is used to assign as many as seven features. Each voice terminal and individual attendant is assigned one of 16 COSs to determine whether or not it has any or all of these features. COS serves no other purpose than to assign these features.

Interactions

None.

Administration

A COS is assigned to each voice terminal extension by the System Manager. The parameters for each COS can be changed. The only other administration required is the assignment of a COS to each individual attendant and voice terminal.

A COS should be assigned on ‘Data Module’ and ‘Access Endpoint’ forms. A separate COS should be used for data applications.

A COS can also be assigned to a remote access barrier code.

Hardware and Software Requirements

No additional hardware or software is required.

Code Calling Access

Feature Availability

Code Calling Access is available with all Generic 3 releases.

Description

Allows attendants, voice terminal users, and tie trunk users to page with coded chime signals.

As many as nine individual paging zones can be provided. (A zone is the location of the loudspeakers, for example, conference rooms, warehouses, etc.) In addition, one zone can be provided to activate all zones simultaneously. Each paging zone requires a separate Code Calling Access code.

A paging party dials the Code Calling Access code and the extension number assigned to the person to be paged. The paging party is automatically parked (through the Call Park feature) on the paged party's extension number. The system translates the number to a chime code and then plays the code over loudspeakers. The paged party, recognizing the chime code, can answer the call from any voice terminal within the system by dialing the Call Park Answer Back access code and his or her own extension number.

Considerations

With Code Calling Access, users do not have to be at their own voice terminal in order to answer calls. Users who are frequently away from their voice terminal or at a location where a ringing voice terminal might be disturbing can be assigned a chime code. When a user's chime code is heard, that user can answer the parked call from a nearby voice terminal.

The system can have up to nine individual zones plus one zone to activate all zones simultaneously.

As many as 125 three-digit chime codes can be provided. Only one extension number can be assigned to each chime code.

Interactions

The following features interact with the Code Calling Access feature.

- Abbreviated Dialing

  If Abbreviated Dialing is used for Code Calling Access, special characters should not be used. If they are used, the call is denied.

- Call Park

  This feature is automatically provided with Code Calling Access.

- Conference — Attendant

  A call cannot be conferenced while accessing paging equipment. The attendant can, however, release the call after paging the called party.

- Conference — Terminal

  A call cannot be conferenced while accessing paging equipment.

- Controlled Restriction

  Controlled Total restriction prohibits use of Code Calling Access.

- Loudspeaker Paging Access

  It is not possible to use a PagePac. paging system for Code Calling Access when multi-zone paging is desired. The PagePac paging systems expect a two-digit code to access a particular zone. The system, however, immediately plays the chime code once a connection is established.

- Miscellaneous Trunk Restriction

  Voice terminals and tie trunks with this restriction cannot use Code Calling Access.

- Origination Restriction

  This restriction prohibits use of Code Calling Access.

- Transfer

  A call cannot be transferred while accessing paging equipment.

Administration

Code Calling Access is administered by the System Manager. The following items can be administered:

- Trunk access code and COR for each of the nine individual paging zones and for the zone used to activate all zones simultaneously.
- Number of times (one to three) the chime code plays. If the chime code is set to play more than once, the paging party must remain on the call until the chime code is repeated the desired number of times.
- Loudspeaker locations (name of zone).
- Three-Digit chime codes for extensions. The codes are combinations of the digits one through five.

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law and Mu-law) for each individual zone. (These hardware requirements can be shared with the Loudspeaker Paging Access feature. Activation of each feature is by the assigned trunk access code.)

No additional software is required.

Conference — Attendant

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the attendant to set up a conference call for as many as six conferees, including the attendant. Conferees from inside and outside the system can be added to a conference call. To set up a conference, the attendant dials a number and uses the split key to add each party to the rest of the conference.

Considerations

Whenever an attendant needs to talk with more than one party at the same time, the Attendant Conference feature can be used. An attendant can also establish a conference call for other voice terminal users or parties outside the system.

The attendant can set up only one conference call at a time. The attendant can hold a conference call on the console or release from it.

The attendant cannot handle any other calls while setting up a conference call.

Once an attendant adds a party to a conference call (whether the call was established by an attendant or other voice terminal user), only the attendant can add another party to the call.

Interactions

The following features interact with the Conference — Attendant feature.

- Bridged Call Appearance

  A Bridged Appearance button can be used to make conference calls.

- Call Vectoring

  A call to a VDN can be included as a party in a conference call only after vector processing terminates for that call (for example, after a successful ***route to*** command).

- Trunk-to-Trunk Transfer

  If Trunk-to-Trunk Transfer is disabled and the attendant releases from a conference call involving only trunk conferees, the trunks are also disconnected.

- When a multi-function station (BRI/Digital/Hybrid) dials sufficient digits to route a call, but could route differently if additional digits were dialed, the station will not recognize the Conference or Transfer buttons. The user must delay dialing for ten seconds or dial a # to indicate the call can be routed based on the digits already dialed. The Conference or Transfer buttons are then recognized and the operation is completed by the switch.
- Attendant Conferencing may not operate properly if the CO does not provide answer supervision and the “Answer Supervision Timeout” field is set to “0” and the “Receive Answer Supervision” is set to “y.” In this case, the use of any non-zero number can be used to represent the number of seconds of delay required for answer supervision.

  If the CO *does* provide answer supervision, the “Answer Supervision Timeout” can be set to “0” and the “Receive Answer Supervision” can be set to “y.”

Administration

For G3i-Global, G3rV1, G3V2, and later releases the administrator may specify on the System Parameters form lower limits on the number of parties that can be on a conference call.

Hardware and Software Requirements

No additional hardware or software is required.

Conference — Terminal

Feature Availability

Conference — Terminal is available with all Generic 3 releases.

Description

Allows multi-appearance voice terminal users to set up six-party conference calls without attendant assistance.

Considerations

With the Conference — Terminal feature, voice terminal users can set up their own conference calls without assistance from an attendant.

With assistance from other users, a single-line voice terminal can have more than three parties on a conference call. For example, one user can add a party, who can add another party, etc.

If a voice terminal user releases from a conference call involving only trunk conferees, the trunks are also disconnected if trunk-to-trunk connections are disallowed through administration.

Single-line voice terminal users can set up three-party conference calls without attendant assistance, only if the “Allow Conference via Flash” field is set to y (yes) on the “Feature Related System Parameters” form.

Up to six parties may be conferenced together on an active call. If the conference tone feature is turned on, all six conferees hears the conference tone.

If an analog single line set has the Call Wait feature enabled and has created a conference call, the Call Wait feature is denied. For example, when caller A on an analog set is talking to caller B, then flashes and is talking to caller C, and flashes to conference B and C, then, if caller D calls caller A, the Call Wait feature is denied.

Interactions

- Bridged Call Appearance

  A Bridged Appearance button can be used to make conference calls.

- Call Vectoring

  A call to a VDN can be included as a party in a conference call only after vector processing terminates for that call (for example, after a successful ***route to*** command).

- Class of Restriction (COR)

  The COR of the party being added is always checked against the COR of the party controlling the add-on, but the new party's COR is not checked against any other conferee's CORs if Restriction Override is set to ''all.'' It is checked against ''attendants'' or ''none.''

- Internal Automatic Answer (IAA)

  Internal conference calls are eligible to be automatically answered via the IAA feature. If more than one conference party has joined a conference call through automatic answer, such parties remain connected until either they or the controlling party drops the call.

- When a multi-function station (BRI/Digital/Hybrid) dials sufficient digits to route a call, but could route differently if additional digits were dialed, the station will not recognize the Conference or Transfer buttons. The user must delay dialing for ten seconds or dial a # to indicate the call can be routed based on the digits already dialed. The Conference or Transfer buttons are then recognized and the operation is completed by the switch.

Administration

On the 'System Parameters" form there is a field where the administrator may specify lower limits on the number of parties that can be on a conference call. A conference tone is available through administration.

Hardware and Software Requirements

No additional hardware or software is required.

Constellation Voice/Data Terminal Support

Feature Availability

This feature is available with the Generic 3 Version 3 and later release.

Description

Provides support for integrating voice terminals with the Constellation Voice/Data Terminal. The Constellation is a voice/data workstation that provides an integrated environment for applications using DCP-based (digital communications protocol) signaling.

The Constellation operates in conjunction with an adjunct voice terminal.

Feature History and Development

The Constellation was available on the DEFINITY system Generic 2 and on a custom version of G3i. It is supported in its entirety in Generic 3 V3 and later releases. The Constellation terminal and its associated Cluster Controller are no longer manufactured or sold by AT&T. These are Memorex/Telex products that are natively supported on the DEFINITY system Generic 3 V3 and later releases. Further details on Constellation terminal applications and installation should be obtained from Memorex/Telex.

Applications and Benefits

The Constellation Voice/Data Terminal provides enhanced capability for a telemarketing environment. It is meant to assist a telemarketing environment by interacting with the switch and a host computer to provide agents with information about the calling and called party. The Constellation interprets information about a call, passes the information to a host computer, and receives from the host information about the either party. The information sent by the host computer appears on the workstation monitor in a user-supplied application and format.

The Constellation operates by ‘‘listening’’ to the DCP connection between the switch and the voice terminal. When a new call originates or terminates on the switch, the switch sends the Constellation a start-of-call message in the form of a DCP S-channel message (Transparent Call Control). This message contains the best available information about the calling party and called party. If either piece of information is not available, it sends a ‘‘NULL’’ message. Likewise, when the call has terminated, the switch sends a similar end-of-call message to the Constellation.

Configuration

The following diagram illustrates the hardware/signaling arrangement for the Constellation.

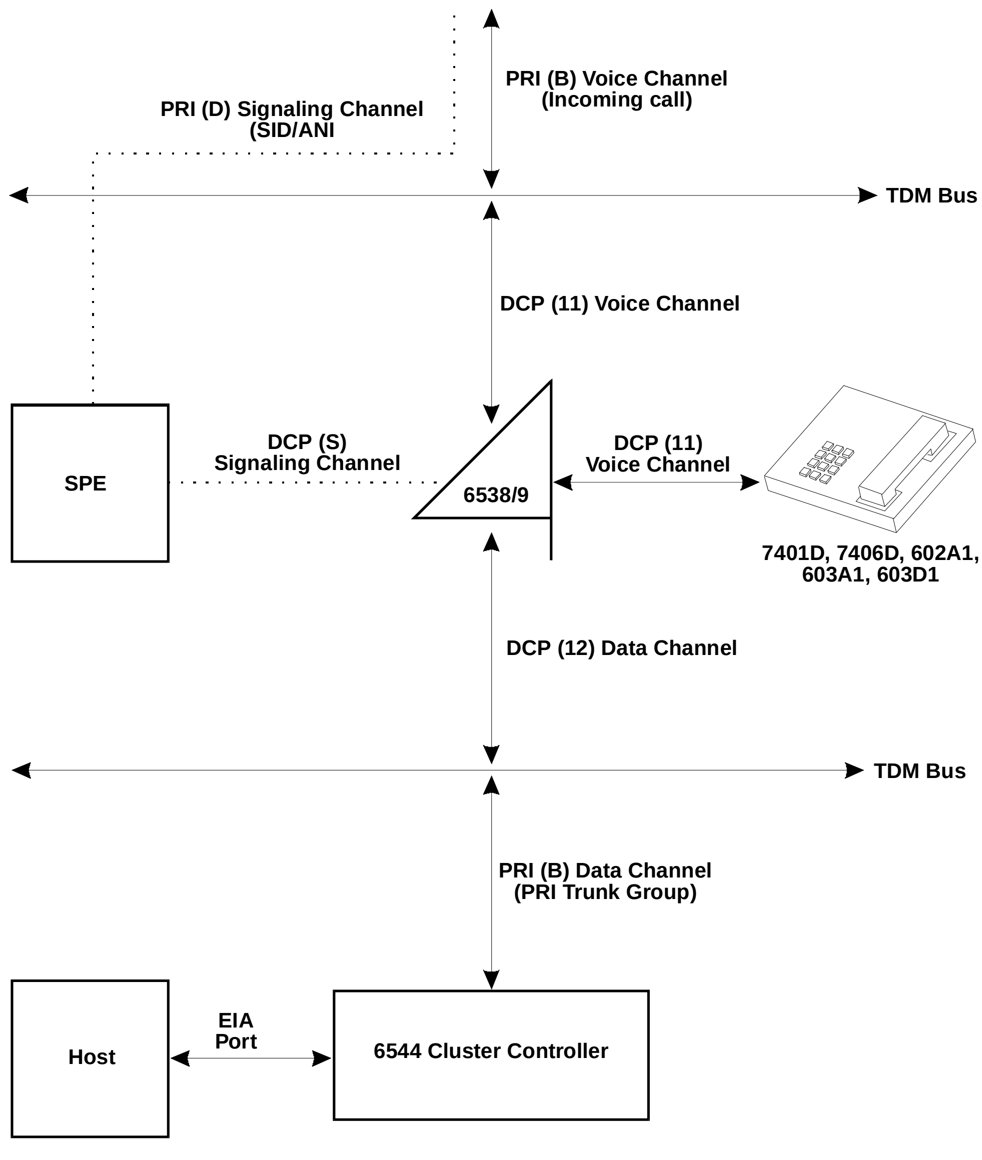


Figure 3-3. Constellation Hardware/Signaling Arrangement

Considerations

G3V3 and later releases provide native support for the Constellation. ‘‘Native support’’ means the switch stores a predefined terminal type and fully supports the terminal operation. The constellation terminal is a Memorex/Telex product. Memorex/Telex is responsible for manufacturing and installing this terminal. automatically downloads the terminal type information to the terminal.

Administration

The Constellation is administered on the station form as a type ‘‘6538.’’ The form for this type includes a “Set” field, which requires one of the following values, based on the adjunct voice terminal: ‘‘7401D,’’ ‘‘7406D,’’ ‘‘602A1,’’ ‘‘603A1,’’ or ‘‘603D1.’’ The value in the “Set” field determines the remaining fields that appear on the form.

Hardware/Software Requirements

The Constellation requires the use of one of the following adjunct voice terminals: 7401D, 7406D, 602A1, 603A1, and 603D1.

The adjunct voice terminal can function as a standalone station even when the Constellation is powered down.

The Constellation also requires a 6544 cluster controller to provide a data connection to the host.

Consult

Feature Availability

Consult is available with all Generic 3 releases.

Description

Allows a covering user, after answering a coverage call, to call the principal (called party) for private consultation.

Consult is activated by first pressing the Conference or Transfer button followed by the Consult button to call the principal. This places the calling party on hold and establishes a connection between the principal and the covering user. The covering user can then add the calling party to the conversation, transfer the call to the principal, or return to the calling party.

Details of how Consult is used in conjunction with Call Coverage are given in the Call Coverage feature description elsewhere in this section.

Considerations

Consult can be used to let a covering user consult with the principal, to determine whether he or she wishes to speak with the called party (for example, an executive's secretary may wish to consult the executive on an established call).

Interactions

Consult calls use the Temporary Bridged Appearance maintained on the call. At the conclusion of a Consult call, the bridged appearance is no longer maintained.

Bridged Call Appearances of the principal's extension are not alerted on a Consult call to the principal extension.

Consult is only used in conjunction with the Call Coverage feature.

A Consult call acts as a priority call and waits at a single-line voice terminal, even if the single-line voice terminal does not have Call Waiting Termination assigned.

Administration

Consult is administered by the System Manager on a per-voice terminal basis. The only administration required is the assignment of a Consult button.

Hardware and Software Requirements

No additional hardware or software is required.

Coverage Callback

Feature Availability

Coverage Callback is available with all Generic 3 releases.

Description

Allows a covering user to leave a message for the principal (called party) to call the calling party.

Coverage Callback is activated by pressing the Cover Callback button after answering a coverage call.

Details of how Coverage Callback is used in conjunction with Call Coverage are given in the Call Coverage feature description elsewhere in this section.

Considerations

Coverage Callback is useful whenever it is necessary to let the principal know that a call has been received from a certain party.

Interactions

Coverage Callback is only used in conjunction with the Call Coverage feature.

Administration

Coverage Callback is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Cover Callback button.

Hardware and Software Requirements

No additional hardware or software is required.

Coverage Incoming Call Identification (ICI)

Feature Availability

Coverage Incoming Call Identification is available with all Generic 3 releases.

Description

Allows multiappearance voice terminal users without a display in a Coverage Answer Group to identify an incoming call to that group.

When an incoming call is directed to a Coverage Answer Group, the status lamp associated with the Coverage Answer Group button lights at group member's voice terminal.

Details of how Coverage ICI is used in conjunction with Call Coverage are given in the Call Coverage feature description, elsewhere in this section.

Considerations

With Coverage ICI, members of Coverage Answer Groups do not have to have a display in order to identify incoming calls to the group.

A second coverage call takes control of the Coverage ICI lamp and does not return control to the previous call when the second call is released.

Interactions

Coverage ICI is used only in conjunction with the Call Coverage feature.

Administration

Coverage ICI is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Coverage Answer Group button.

Hardware and Software Requirements

No additional hardware or software is required.

Customer-Provided Equipment (CPE) Alarm

Feature Availability

Customer-Provided Equipment (CPE) Alarm is available with all Generic 3 releases.

Description

Provides the customer with an indication that a system alarm has occurred and that the system has attempted to contact a preassigned service organization about the problem. A customer-provided device, such as a lamp or a bell, is used to indicate the alarm situation.

The system can be administered so that the CPE Alarm will be activated during certain alarm levels. Only one of these levels may be administered. The CPE Alarm will be activated when an alarm occurs which corresponds to, or is more severe than, the administered alarm activation level. The levels for which the CPE Alarm can be activated are listed below in descending order, beginning with the most severe.

- Major Alarm — This alarm is the most severe system alarm. A major alarm indicates that a vital system hardware component, which will seriously affect overall service, has failed.
- Minor Alarm — This alarm indicates that a hardware component, which may affect service on a limited scale, has failed.
- Warning Alarm — This alarm indicates that a problem may exist with a hardware component, but the problem does not affect service.

The system can also be administered so that the CPE Alarm is not activated under any of the previously listed alarm levels.

The CPE Alarm is also activated during a Power Failure Transfer (see the Power Failure Transfer feature elsewhere in this manual) regardless of the administered alarm activation level. Even if the system is administered so that the CPE Alarm is not activated at any alarm level, it will be activated during a Power Failure Transfer.

The CPE Alarm is deactivated when the problem that caused the alarm is resolved. If there are multiple problems, the CPE Alarm will not be deactivated until all problems, at or above the administered alarm activation level, are resolved.

For more information, see *DEFINITY Communications System Generic 1 and Generic 3 Installation and Test*, 555-230-104.

Considerations

The CPE Alarm feature lets customers use their own equipment to indicate an alarm condition. This indication lets the customer know when there is a problem with the system and when the problem has been resolved.

Interactions

The following features interact with the CPE Alarm feature.

- Power Failure Transfer

  The CPE Alarm is always activated during a Power Failure Transfer regardless of the administered alarm activation level.

Administration

The CPE Alarm feature is administered on a per-system basis by the System Manager.

Hardware and Software Requirements

The only hardware required is the actual CPE Alarm device (lamp, bell, and so on). This device must be customer-provided and customer-installed.

No additional software is required.

Data Call Setup

Feature Availability

Data Call Setup is available with all Generic 3 releases.

Description

Provides multiple methods to set up a data call:

- Keyboard dialing
- Voice terminal dialing
- Hayes "AT" command dialing
- Permanent switched connections
- Administered connections
- Automatic calling unit interface (MPD and HSC)
- Hotline dialing

Typically, when a data terminal is available, keyboard dialing is more convenient and requires less steps; therefore, it should be used whenever possible.

In addition to data terminal dialing and voice terminal dialing, the system accepts calls from other devices, such as a MPDM equipped with an ACU interface module. An analog modem interfaced with an ACU can also be used to provide dialing capability for a host computer.

The "Administered Connections" feature, described elsewhere in this chapter, may also be used to establish a data call.

This section describes the data call setup features for both DCP sets and ISDN BRI sets.

Data Call Setup for DCP Modules

Voice Terminal Dialing for DCP Data Modules

Allows voice terminal users to originate and control data calls from the voice terminal. DCP voice terminal dialing must be used when the Data Terminal is not accessible for dialing.

The Transfer feature functions the same for data calls as it does for voice calls. The feature permits a user to set up a call using any unrestricted voice terminal and then transferring the call to a data endpoint. However, the primary way to establish data calls is with the multiappearance voice terminal Data Extension button(s). Any administrable feature button can be assigned as a Data Extension button in system administration. The Data Extension button provides one-touch access to a data module.

The voice terminal Data Extension buttons control the One-Button Transfer to Data, Return-to-Voice, and Data Call Preindication operations for the associated data module. These operations are discussed below. Multiple Data Extension buttons can be assigned to a multiappearance voice terminal, and that voice terminal can set up data calls for other data terminals. Also, a single data module can be accessed by a Data Extension button on a voice terminal. Only one Data Extension button can be administered for a single data module.

Voice terminal dialing has the advantage that the user may hear the different types of network tones.

For off-premises dialing, particularly for toll calls, the user may opt for voice terminal dialing, instead of keyboard dialing.

The following options, either alone or combined, permit flexible procedures for establishing data calls:

- One-Button Transfer to Data

  Allows a user to transfer the call to the associated data module simply by pressing the Data Extension button after the called data endpoint answers. This method is recommended for voice terminal data call setup.

- Return-to-Voice

  Allows a user to return the data connection to the voice terminal. The user simply presses the Data Extension button associated with the busy data module. If the user hangs up following the return, the call is disconnected. If Return-to-Voice is affected by two voice terminal users, each through use of the Data Extension button associated with the two data endpoints of the call, then a voice call is established. Return of a data call to the voice terminal implies that the same (data) call will be continued in the voice mode, or transferred to another data endpoint.

- Data Call Preindication

  Allows the user, before dialing the distant data endpoint, to reserve the associated data module by pressing the Data Extension button. This ensures that a conversion resource, if needed, and the data module are reserved for the call. Use of Data Call Preindication before one button transfer to data is recommended when establishing data calls that use toll network facilities. Needed conversion resources are reserved before any toll charges are incurred.

Data Terminal (Keyboard) Dialing for DCP Data Modules

Allows a user to set up and disconnect data calls directly from a data terminal. A voice terminal is not needed. The voice terminal functions of switchhook and the audible call progress tones are replaced with keyboard dialing and text known as call progress messages. The message **DIAL:** prompts the user to enter the called data number manually from the keyboard, and RINGING informs the user the called data number is being rung. If the data call is placed in queue, the message WAIT, xx IN QUEUE is received (xx represents queue position). This queue number is updated by the system as the call moves up in the queue. The next table lists the call progress messages.

To originate and disconnect a call using Data Terminal Dialing, the user presses BREAK on the terminal. [This is equivalent to a voice terminal user lifting the handset (call origination) or hanging up (call disconnect).] If the terminal being used does not generate a 2-second continuous break signal, the user can press Originated/Disconnect on the data module. Then, the data terminal allows the user to enter digits from the data terminal keyboard, after the message **DIAL:** (which is the equivalent of dial tone on a voice terminal).

In addition to the numeral, #, and * characters found on a touch-tone pad, the dialing information may contain the following special characters:

- SPACE, —, (, and) may be used to improve legibility. These characters are ignored by the system during dialing.
- + character (wait) may be used to interrupt or suspend dialing until dial tone is received from the distant switch.
- , (pause) character may be used to place a 1.5-second pause in dialing (multiple, can be used).
- % (mark) character may be used to indicate the following digits are for end-to-end signaling (touch-tone). This is required when the trunk is rotary. It is not required when the trunk is touch-tone.
- UNDERLINE or BACKSPACE may be used to correct previously typed characters on the same line.
- @may be used to delete the entire line and start over with a new `DIAL`: prompt.

Each line of dialing information may contain up to 42 characters (the + and % characters count as two each).

Examples of dialing are as follows:

- DIAL: 3478
- DIAL: 9+(201) 555-1212
- DIAL: 8, 555-2368
- DIAL: 9+555-2368+%9999+123 (remote access)

Single-Line Dialing

All of the dialing information, including pauses and ignored characters, are typed on a single line. The line with the `DIAL`: prompt must be complete; that is, the dialing information must specify a complete call before the carriage return or line feed.

Single-line dialing is recommended if all dialing information can be entered on one line.

Multiple-Line Dialing

Automatically invoked when a single line of dialing information is incomplete. Multiple-line dialing is only used with off-premises calling.

In multiple-line dialing, the `DIAL`: prompt follows on the next line when all of the dialing information of the previous line has been sent and dial tone has occurred; additional dialing information is requested.

This is a typical off-premises dialing sequence:

- DIAL: 9
- DIAL: (201) 555-2368
- RINGING
- ANSWERED

Alphanumeric Dialing

Alphanumeric Dialing enhances Data Terminal Dialing by allowing a data terminal user to place a data call by entering an alphanumeric name. This capability makes Data Terminal Dialing both convenient and user-friendly. Instead of dialing a long string of numbers, the user can enter a simple alphanumeric name. For more detailed information, see the "Alphanumeric Dialing" feature description, elsewhere in this chapter.

Call Forwarding All Calls

Call Forwarding All Calls allows incoming data calls to be redirected to another extension that is designated by the user. Activation/deactivation of the feature is done either by the attendant or by the forwarding party itself through the dialing of a feature access code.

Default Dialing

Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination by simply entering a carriage return at the `DIAL`: prompt. The data terminal user can still place calls to other destinations by entering the complete address after the `DIAL`: prompt (normal Data Terminal Dialing or Alphanumeric Dialing). For more detailed information, see the "Default Dialing" feature description elsewhere in this chapter.

Administered Connections

An Administered Connection is an end-to-end connection between two access endpoints or data endpoints that is automatically established by the system whenever the system is restarted or the Administered Connection is administered and due to be active. The attributes of these connections are user-defined. To administer Administered Connections, use the “Administered Connection” form via the SAT.

Once the ADM has been administered as one endpoint of an administered connection, the system waits for the scheduled time to set up the connection. At the scheduled time, the system establishes the connection and maintains it for the length of time specified. Once the call is accepted, the set will enter into the continuous mode for the length of time specified. If the switch is rebooted during the continuous connection, the connection will reinitiate the call setup. At any time that the connection drops (for example, disconnected cabling), the switch will reinitiate the call setup.

Hotline Dialing

Hotline Dialing is discussed in detail in the Data Hot Line feature description elsewhere in this manual. Table 3-50 shows call progress messages for DCP.

Table 3-50. Call Progress Messages for Keyboard Dialing for DCP

Displayed Message	Application	Meaning
DIAL:	Placing a call	Equivalent to dial tone. Enter the desired number or feature access code followed by a carriage return or a line feed.
RINGING	Placing a call	Equivalent to ringing tone. Called terminal (far-end) is ringing.
BUSY	Placing a call	Equivalent to busy tone. Called number is in use or out of service.
ANSWERED	Placing or receiving a call	Notifies calling and called users that call has been answered.
ANSWERED - NOT DATA	Placing a call	Notifies calling and called users that call has been answered and a modem answer tone has not been detected.
TRY AGAIN	Placing a call	Equivalent to reorder tone. System facilities are currently not available.
DENIED	Placing a call	Equivalent to intercept tone. Call cannot be placed as dialed.
ABANDONED	Receiving a call	Notifies called user that the calling user abandoned the call.
NO TONE	Placing a call	Notifies user that tone was not detected.

Continued on next page

Table 3-50. Call Progress Messages for Keyboard Dialing for DCP — *Continued*

Displayed Message	Application	Meaning
CHECK OPTIONS	Placing a call	Notifies calling terminal that data module options are incompatible.
XX IN QUEUE	Call in queue	Current position of the user in queue.
		XX-indicates position.
PROCESSING*	Call in queue	Notifies user when out of queue. Facility is available.
TIMEOUT*	Call in queue	Notifies user when time has been exceeded. Call will be terminated.
FORWARDED*	Receiving a call	Equivalent to redirection notification signal. Called terminal has activated Call Forwarding and received a call, and call has been forwarded.
INCOMING CALL-*	Receiving a call	Equivalent to ringing.
INVALID ADDRESS	Placing a call	The entered name is not in the Alphanumeric Dialing Table.
PLEASE ANS-	Receiving a call	Originating voice terminal user has transferred call to data module using One-Button Transfer to Data.
-TRANSFER	Call is transferred to voice	Notifies calling terminal when Data Call Return-to-Voice occurs.
CONFIRMED	Activating or deactivating a feature	Equivalent to confirmation tone. Feature request is accepted, or call has gone to a local coverage point.
-OTHER END	During a call	Notifies user that the other end
		terminated the call.
DISCONNECTED*	Call is terminated	Call or call attempt is disconnected from system
WAIT	Placing a call	Notifies user that normal processing is continuing.
WAIT, XX IN QUEUE	Placing a call	Notifies user that call entered a local hunt group queue. XX indicates position.

* Bell sounds when message is displayed.

Data Call Setup for ISDN-BRI Modules

Voice Terminal Dialing for ISDN-BRI Data Modules

Allows ISDN-BRI voice terminal users to directly originate a data call. To set up a data call, the user just presses the Data button on the ISDN-BRI voice terminal, enters the desired number on the dial pad, and then presses the Data button again.

The following data functions are not supported by BRI terminals:

- One button transfer to data
- Return-to-voice
- Data call pre-indication
- Voice call transfer to data and data call transfer to voice

Data Terminal (Keyboard) Dialing for ISDN-BRI Data Modules

Allows a user to set up and disconnect data calls directly from a data terminal without using a voice terminal. The voice terminal functions of switchhook and the audible call progress tones are replaced with keyboard dialing and text known as call progress messages. Unlike DCP, BRI is terminal dependent, meaning that the BRI data module, not the switch, prompts the user to enter information.

NOTE:
The 7500B Data Module also allows users to set up calls via its front panel. For more information on this feature, consult the *7500B Data Module User's Manual*, 555-021-717.

Before the user can make a data call using Data Terminal Dialing, CMD: must appear on the screen of the terminal. To access the CMD: prompt before placing a call, the user must press Enter on the keyboard a few times. If the CMD: prompt does not appear, the user must press Break A and T at the same time, and then press Enter. To make a data call, the user types dial, enters a space, types the desired telephone number, and presses Enter at the CMD: prompt (For example, dial 1234567).

To disconnect a data call using Data Terminal Dialing, the user must first enter **+++** to access the CMD: prompt. At the CMD: prompt, the user types end and presses ENTER.

In addition to the numeral, #, and * characters found on a touch-tone pad, the dialing information may contain the following special characters:

- SPACE,—, (, and) may be used to improve legibility. These characters are ignored by the system during dialing.
- + character (wait) may be used to interrupt or suspend dialing until dial tone is received from the distant switch.

- , (pause) character may be used to place a 1.5-second pause in dialing. (multiple, can be used).
- % (mark) character may be used to indicate the following digits are for end-to-end signaling (touch-tone). This is required when the trunk is rotary. It is not required when the trunk is ‘‘touch-tone.’’
- UNDERLINE or BACKSPACE characters may be used to correct previously typed characters on the same line.
- @ may be used to delete the entire line and start over with a new

  CMD: prompt.

Each line of dialing information may contain up to 42 characters (the + and % characters count as two each).

Examples of dialing are as follows:

- CMD: d 3478
- CMD: d 9+(201) 555-1212
- CMD: d 8, 555-2368
- CMD: d 9+555-2368+%9999+123 (remote access)

Basic Digit Dialing

Regular digit dialing is provided through the Asynchronous Data Module (ADM) or 7500B Data Module. Digits from 0 to 9, ‘‘*’’, and ‘‘#’’ can be entered. This feature can be used by the user either from the associated 7500 Series voice terminal keypad or from the EIA terminal interface.

Alphanumeric Dialing

Alphanumeric Dialing enhances Data Terminal Dialing by allowing a data terminal user to place a data call by entering an alphanumeric name. This capability makes Data Terminal Dialing both convenient and user-friendly. Instead of dialing a long string of numbers, the user can enter a simple alphanumeric name. For more detailed information, consult the "Alphanumeric Dialing" feature description elsewhere in this chapter.

Default Dialing

Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination by either typing a d and entering Return at the CMD: prompt, or by pressing the data button twice. If no default dialing has been administered, the call will be disconnected in less than one second. The data terminal user can still place calls to other destinations by typing a d and the complete address at the CMD: prompt, and then entering Return (normal Data Terminal Dialing or Alphanumeric Dialing). This feature is mutually exclusive with the Data Hotline feature. For more detailed information, see the "Default Dialing" feature description elsewhere in this chapter.

Call Forwarding All Calls

Call Forwarding All Calls allows incoming data calls to be redirected to another extension that is designated by the user. Activation/deactivation of the feature is done either by the attendant or by the forwarding party itself through the dialing of a feature access code.

Data Hotline

Data Hotline is a security feature. The switch will terminate the call to a preadministered hotline destination. If a user enters an address either intentionally or unintentionally, the call processing will discard the address string received for the hotline endpoint. The call processing will automatically route the call just as if the hotline destination address had been entered by the user. This service does not impose any restriction on incoming calls received at the endpoint. This feature is mutually exclusive with the Default Dialing feature.

Administered Connections

An Administered Connection is an end-to-end connection between two access endpoints or data endpoints that is automatically established by the system whenever the system is restarted or the Administered Connection is administered and due to be active. The attributes of these connections are user-defined. To administer Administered Connections, use the “Administered Connection” form via the SAT.

Once the ADM has been administered as one endpoint of an administered connection, the system waits for the scheduled time to set up the connection. At the scheduled time, the system establishes the connection and maintains it for the length of time specified. Once the call is accepted, the set will enter into the continuous mode for the length of time specified. If the switch is rebooted during the continuous connection, the connection will reinitiate the call setup. At any time that the connection drops (for example, disconnected cabling), the switch will reinitiate the call setup.

Call Request

DEFINITY system Generic 3 call processing will handle all various BRI Bearer data call requests that are presently defined. Some capabilities that are not supported by AT&T terminals may be provided by a non-AT&T terminal. The switch will complete most call requests. For those capabilities that the switch does not support, a proper cause value will return back to the terminal.

Cause Value

BRI stations will receive a cause or reason code that identifies why the call is being cleared. The BRI data modules will convert certain cause values to text messages and display them for the user.

Endpoint Initialization

BRI endpoints have to successfully complete endpoint initialization procedures in order to be fully operative. It is usually carried out at the time of installation, or as part of reconfiguration.

Multipoint Configurations on BRI ports

In a passive bus multipoint configuration, the system supports two BRI endpoints per port, thus doubling the capacity of the BRI circuit pack. When changing the configuration of a BRI from point-to-point to multipoint, the original endpoint need not be reinitialized. However, only endpoints that support SPID initialization can be administered in a multipoint configuration.

Exchange of User Information

The BRI protocol provides the users the capability of exchanging up to 128 octets of user information end-to-end. Displayed messages are shown in Table 3-51. The information is passed in the User Information IEs to the receiving endpoint without being interpreted by the switch. However, there are some limitations to the exchange of User Information IEs.

Table 3-51. Call Progress Messages for Keyboard Dialing for BRI

Displayed Message	Application	Meaning
CMD:	Placing a call	Equivalent to dial tone. Enter the desired number or feature access code followed by a carriage return or a line feed.
RINGING	Placing a call	Equivalent to ringing tone. Called terminal (far-end) is ringing.
BUSY	Placing a call	Equivalent to busy tone. Called number is in use or out of service.
ANSWERED	Placing or receiving a call	Notifies calling and called users that call has been answered.
TRY AGAIN	Placing a call	Equivalent to reorder tone. System facilities are currently not available.
DENIED	Placing a call	Equivalent to intercept tone. Call cannot be placed as dialed.
ABANDONED	Receiving a call	Notifies called user that the calling user abandoned the call.
NO TONE	Placing a call	Notifies user that tone was not detected.

Continued on next page

Table 3-51. Call Progress Messages for Keyboard Dialing for BRI — *Continued*

Displayed Message	Application	Meaning
CHECK OPTIONS	Placing a call	Notifies calling terminal that data module options are incompatible.
XX IN QUEUE	Call in queue	Current position of the user in queue.
		XX-indicates position.
PROCESSING*	Call in queue	Notifies user when out of queue.
		Facility is available.
TIMEOUT*	Call in queue	Notifies user when time has been exceeded. Call will be terminated.
FORWARDED*	Receiving a call	Equivalent to redirection notification signal. Called terminal has activated Call Forwarding and received a call, and call has been forwarded.
INCOMING CALL-*	Receiving a call	Equivalent to ringing.
WRONG ADDRESS	Placing a call	The entered name is not in the Alphanumeric Dialing Table.
PLEASE ANS-	Receiving a call	Originating voice terminal user has transferred call to data module using One-Button Transfer to Data.
CONFIRMED	Activating or deactivating a feature	Equivalent to confirmation tone. Feature request is accepted, or call has gone to a local coverage point.
-OTHER END	During a call	Notifies user that the other end terminated the call.
DISCONNECTED*	Call is terminated	Call or call attempt is disconnected from system.
WAIT	Placing a call	Notifies user that normal processing is continuing.

* Bell sounds when message is displayed.

Considerations

All systems have Data Call Setup capability. This facilitates data calling by eliminating the need to dedicate a voice terminal for data calls. DEFINITY system Generic 3 offers the enhancement of off-premises Multiple Line Dialing.

BRI has a voice to data restriction. A voice terminal cannot call a data terminal, and a data terminal cannot call a voice terminal.

BRI voice terminals cannot have Data Extension buttons. Although DCP sets have Data Extension buttons, these sets cannot have Data Extension buttons for BRI data extensions.

When a voice terminal user places a data call to a digital data endpoint, and does not transfer the call to another digital data endpoint but uses a modem or acoustically coupled modem, the user must dial the Data Origination access code assigned in the system before dialing the distant endpoint.

Data Call Preindication is activated by pressing a Data Extension button before dialing the distant data endpoint. Preindication is in effect until the associated Data Extension button is pressed again for a one-button transfer; there is no time-out.

The number of assigned Data Extension buttons per voice terminal is not limited. However, only one voice terminal can be assigned buttons that access the same data module.

When placing outgoing or off-premises calls via keyboard dialing, the call progress message WAIT indicates recognition of the nature of the call and acceptance of the call. The ANSWERED text indicates completion of outpulsing over the selected trunk, or if answer supervision or an answer tone is received.

Interactions

The following features interact with the Data Call Setup feature.

- Abbreviated Dialing

  This feature can be used by voice terminal or Data Terminal (Keyboard) Dialing users on calls to data endpoints. Only 22 of the 24 available digits in an abbreviated dialing number can be used for keyboard dialing. The remaining two digits must contain the ‘‘wait’’ indicator for tone detection.

- Call Coverage

  A hunt group made up of digital data endpoints should not be assigned a coverage path.

- Call Forwarding All Calls

  Calls incoming to a data module can be forwarded. That is, calls can be redirected to another endpoint. This feature is activated using Data Terminal (Keyboard) Dialing. If the forwarded-to endpoint is an analog data endpoint, and the calling user is a digital endpoint, modem pooling is activated automatically.

- Data Call Hot Line

  Upon going off-hook for origination, the system automatically places a call to a predesignated local or off-premises destination.

- Default Dialing

  Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination.

- Internal Automatic Answer (IAA)

  Data calls are not eligible for IAA.

- Modem Pooling

  When modem pools are provided, this feature is automatically available on data calls when the system ascertains the need for a conversion resource. The system automatically inserts the conversion resource. Data Call Preindication or Data Origination can also be used to indicate that a conversion resource is needed.

- CDR

  Data Call CDR records the use of modem pools on trunk calls.

- UCD

  UCD can provide a group of data modules or analog modems for answering calls to facilities, such as computer ports, connected to the data modules or modems.

- ISN Interface

  ISN consists of packet data switches which support data calls between data endpoints and the system. The physical connection to the system is via the DLC board. The DLC provides eight ports for connection with asynchronous EIA RS-232C compatible Data Terminal Equipment to the ISN interface.

  Data Terminal (Keyboard) Dialing is used to access ISN endpoints.

- World Class Tone Detection

  Multiple-Line Data Terminal Dialing is only supported if the administered Level of Tone Detection is ‘‘precise.’’ G3i-Global and G3V2 and later releases provide administration of tone detection options. The message the data call setup feature sends to users vary according to which tone detection option is selected.

If the tone detection option is not set to ‘‘precise,’’ and a data call is set up over an analog trunk, the data call setup messages describing the status of the called endpoint (that is, RINGING, BUSY, TRY AGAIN, etc.), changes according to which tone detection option is selected.

Tone Detection applies to non-ISDN trunk lines as well.

Administration

Data Call Setup does not require assignment as such; however, the following related items require administration by the System Manager:

- Data Origination Access Code — Allow users to indicate a need for a conversion resource on an analog to digital data call origination.
- Port Assignments — Assign the data modules, BCTs, DLCs, 7404D, analog modems.
- Modem Pooling — Assign Circuit Packs or ports.
- Data Extension buttons — Assign Data Extension buttons to multi-appearance voice terminals.
- Default Dialing — See the "Default Dialing" feature.
- Alphanumeric Dialing — See the "Alphanumeric Dialing" feature.

Hardware and Software Requirements

Data Call Setup is a means of using data equipment to establish data calls. Requirements for data modules, voice terminals, and modems are as follows:

- Data Modules: Each DCP data module requires one port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law). A DTDM shares the port with its associated voice terminal.

  Each BRI data module requires one port on a TN556 BRI circuit port pack. Each BRI port may be shared by two endpoints, with each endpoint providing either voice or data capability. To support ISDN-BRI, the switch requires the TN778 Packet Control circuit pack. An ADM shares the port with its associated BRI voice terminal.
- 7400A Data Module: The 7400A Data Module may be used instead of an MTDM when supporting the combined Modem Pooling feature. The 7400A Data Module supports asynchronous operation and provides a DCP interface to the switch and a RS-232c interface to the associated modem. The 7400A can operate in the stand-alone mode as a data module.
- 7500B Data Module: The 7500B Data Module is a stand-alone unit that supports asynchronous or synchronous DCE and asynchronous DTE on the Basic Rate ISDN (BRI) switch interface. In asynchronous mode, the 7500B supports packet or circuit-switched data communications, and can be controlled via the front panel or the keyboard of a connected terminal. The following optional enhancements are available for the 7500B in an

asynchronous DCE configuration: an RS-366 ACU interface and a second asynchronous EIA-232D interface. In synchronous mode, the 7500B supports circuit-switched or nailed-up data communications, requires either the Multi-purpose Enhancement Board or the High-Speed Synchronous Enhancement Board, and can only be controlled via the front panel. In order to be configured as a synchronous DCE, the 7500B must have either the Multi-purpose Enhancement Board or the High-Speed Synchronous Enhancement Board.

When configured as an asynchronous DTE, the 7500B provides an EIA-232D interface and supports full-duplex data transmission at rates of up to 19200 bps. This configuration is most commonly used for modem pooling applications. Regardless of the configuration, the 7500B provides no voice functions and is not used with voice terminals.

- 7400C HSL: 7400C High Speed Link is a data service unit that allows data equipment to access the DCP data services. It provides synchronous data transmission at speeds of 56 and 64 kbps and provides a link to high speed data networks. It can be used for Group 4 Fax applications that will include electronic mail and messaging and electronic storage of printed documents and graphics. It can also be used for video teleconferencing and LAN interconnect applications.
- 7400D Series or CALLMASTER Terminals: For shared use of voice and data, each Voice Terminal requires one port on the following Digital Line circuit packs: the 4-wire TN754, TN413, or the TN754B. The TN754B is A-law or Mu-law selectable. 603EI CALLMASTER uses the TN754, TN754B, or TN2181 circuit pack. The TN2181 is a 2-wire, 16-port, A-law/Mu-law selectable circuit pack. The 7403D and 7405D voice terminals require an optional digital terminal data module. The 7404D requires an optional messaging cartridge, the 7406D requires an optional 703A Data Stand, and the 7407D requires an optional 702A DSU for connection to associated data terminals.
- 7500 Series ISDN Voice Terminals: Each 7500 Series ISDN Voice Terminal requires one port on the TN556 BRI port circuit pack. Each voice terminal requires an optional ISDN ADM to support asynchronous DTE. Consisting of a board located inside the BRI voice terminal, the ISDN ADM allows the transmission of integrated voice and data through one voice terminal. The ISDN ADM shares the port with its associated voice terminal and supports the Hayes command set for compatibility with PC communications packages.
- Modems: Each modem requires one port on a TN742 or TN746B (A-law) Analog Line circuit pack. (Administration designates the modem as a 2500-series voice terminal and assigns an extension number. A modem is connected to the port instead of a voice terminal. Access is through the assigned extension number.)

- Modem Pooling: DEFINITY system Generic 3 requires either a TN758 Modem Pool circuit pack, or one digital port associated with a Trunk Data Module (either TDM or MTDM) and one analog port with analog modem for each conversion resource. A 7400A Data Module may be used in place of the TDM or MTDM.
- Data Line Data Module: Each port is connected to an ADU that converts to an RS-232 interface for connection to a data terminal.

Keyboard Dialing to off-premises data endpoints requires the use of a tone detector circuit pack such as TN748C-Tone Detector, TN420C-Tone Detector, TN744C-Tone Detector/Call Classifier, or TN2182-Tone Clock/Detector/Generator.

NOTE:
The TN748C (Mu-law) and TN420 (A-law) support precise tone detection. However, the TN744C or TN2182 only support precise tone detection in countries that strictly adhere to U.S. standards. Therefore, if the detectors are used in countries that do not adhere to U.S. standards, the users may not experience precise feedback during the data call setup.

T748C and TN2182 support Mu-law. TN420C, TN744C, and TN2182 support A-law. Extensive use of features and services using tone detection may necessitate adding additional tone detector circuit packs.

A TN726 Data Line circuit pack can be used to provide direct access for data terminal users.

No additional software is required.

Data Hot Line

Feature Availability

Data Hot Line is available with all Generic 3 releases.

Description

Data Hot Line provides for automatic nondial placement of a data call to an endpoint when the originator goes off-hook. It may be used for security purposes.

Data Hot Line calls are automatically placed, by the system, from specified digital data endpoints to preassigned extension numbers or off-premises numbers. Hot Line originating endpoints are destinations connected to the system by a data module. The destination number is stored in the Abbreviated Dialing List.

Considerations

Data Hot Line offers fast and accurate call placement to commonly called data endpoints. Data terminal users that constantly call the same destination number can use Data Hot Line to automatically place the call by going off hook.

The number of terminals that can be assigned Data Hot Line is not limited, and the number of terminals that can be assigned the same destination number is not limited. The only limit, if any, would be on the number of entries stored in the Abbreviated Dialing List.

Interactions

The following features interact with the Data Hot Line feature.

- Call Forwarding — All Calls

  A Hot Line originator cannot activate Call Forwarding, since an off-hook intended to dial the Call Forwarding Feature Access Code will cause activation of the Data Call Hot Line feature instead.

- Data Terminal (Keyboard) Dialing

  Any Terminal Dialing text may occur when a Data Hot Line Call is being established, with the exception of the inhibition of the initial dial text message prompt normally given on off-hook for origination.

- System/ISN Access

  A data call to an ISN data endpoint from a system digital data endpoint requires a two-stage dialing. A Hot Line Destination may be an extension number for an outgoing ISN group only, the Hot Line originator must then interact with ISN and manually enter the second address (data endpoint).

Administration

Data Hot Line is administered on per-data terminal basis by the System Manager. The following item requires administration:

- Hot Line Destination Number — The preadministered Hot Line Destination must be stored in an abbreviated dialing list. This list can be a system list, group list, personal list, or enhanced list. Therefore, before this feature can be activated, users' terminals need to be administered with an abbreviated dialing list that stores the Hot Line Destination. Then, each data module needs to specify the index number of the entry which stores the destination. The following fields from the "Data Module" form are needed to activate the Data Hotline feature.

  — Special Dialing Option (default/hotline) — For Data Hotline, ''hotline'' must be selected. Default Dialing and the Data Hotline feature cannot both be assigned to an extension.

  — An Abbreviated Dialing List — One entry in the data module's abbreviated dialing list needs to store the default destination.

  — Dial Code (0-999) — An index to the entry of the abbreviated dialing list that stores the default destination.

Hardware and Software Requirements

No additional hardware or software is required.

Data Privacy

Feature Availability

Data Privacy is available with all Generic 3 releases.

Description

NOTE:
This feature is know as ‘‘Data Protection’’ in G2.

Protects analog data calls from being disturbed by any of the system’s overriding or ringing features. Data Privacy, when activated by a user, denies the system the ability to gain access to, or to superimpose tones onto, the protected call.

To activate this feature, the user dials the activation code at the beginning of the call.

Considerations

All systems have the capability for Data Privacy to provide interruption protection by denying the system the ability to gain access to the protected, analog data call.

Connections involving one or more digital data endpoints (data module) are automatically protected from receiving system-generated tones. In this case, the Data Privacy feature is not needed.

Data Privacy, when activated, applies to both voice and data calls. The feature can be activated on Remote Access calls, but not on other incoming trunk calls. Data Privacy is canceled if the call is transferred, added to a conference call, bridged onto, or disconnected from by the activating user. Data Privacy can be activated on calls originated from attendant consoles.

Interactions

The following features interact with the Data Privacy feature.

- Attendant Call Waiting and Call Waiting Termination

  If Data Privacy is activated, Call Waiting is denied.

- Bridged Call Appearance — Single-Line Voice Terminal

  When Data Privacy is activated or Data Restriction is assigned to a station involved in a bridged call and the primary terminal and/or bridging user attempts to bridge onto the call, Data Privacy and Data Restriction are automatically overridden.

- Intercom — Automatic and Dial

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. Intercept tone is received when the ICOM button is pressed under this condition.

- Music-on-Hold Access

  If a call with Data Privacy activated is placed on hold, Music-on-Hold access is withheld to prevent the transmission of some musical tone which a connected data service might falsely interpret as a data transmission.

- Priority Calls

  If Data Privacy is activated, Priority Calls to the activating extension number are denied on analog voice terminals. However, Priority Calls appear on the next available line appearance on multi-appearance voice terminals.

- Busy Verification cannot be done when data privacy is active.

Administration

Data Privacy is activated by dialing a Data Privacy Feature Access Code (FAC) before originating the call. Permission to use the feature is assigned in COS.

Hardware and Software Requirements

No additional hardware or software is required.

Data Restriction

Feature Availability

Data Restriction is available with all Generic 3 releases.

Description

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data Restriction, when administered to an extension number or trunk group, denies the system the ability to gain access to, or to superimpose tones onto, the protected call.

This feature is administered at the system level to selected analog and multi-appearance voice terminals and trunk groups. Once administered, the feature is active on all calls to or from the associated terminal or trunk group.

Considerations

All systems have the capability for Data Restriction to prevent overriding or ringing features from interrupting the voice or data call.

Connections involving one or more digital data endpoints (data modules) are automatically protected from receiving system-generated tones. In this case, the Data Restriction feature is not needed.

Data Restriction applies to both voice and data calls. Also, Data Restriction cannot be assigned to attendant consoles. Data Restriction is removed from the current call if it is transferred, added to a conference call, bridged onto, or disconnected from by the restricted extension.

Interactions

The following features interact with the Data Restriction feature.

- Attendant Call Waiting and Call Waiting Termination

  If Data Restriction is assigned, Call Waiting is denied.

- Intercom — Automatic and Dial

  An extension with Data Privacy when activated cannot originate an intercom call. An extension with Data Restriction assigned cannot originate an intercom call. Intercept tone is received when the ICOM button is pressed under this condition.

- Music-on-Hold Access

  If a call with Data Restriction assigned is placed on hold, Music-on-Hold access is withheld to prevent the transmission of some musical tone which a connected data service might falsely interpret as a data transmission.

- Priority Calls

  Priority Calls are not denied if the analog station is idle; Call Waiting (including Priority Call Waiting) is denied if the station is busy. However, Priority Calls appear on the next available line appearance on multi-appearance voice terminals.

- Busy Verification cannot be done when Data Restriction is assigned.

Administration

Data Restriction is assigned on a per-line or trunk basis by the System Manager.

Hardware and Software Requirements

No additional hardware or software is required.

Data-Only Off-Premises Extensions

Feature Availability

Data-Only Off-Premises Extensions is available with all Generic 3 releases.

Description

Allows users to establish data calls involving DCE or DTE that is located remotely from the System site using DATAPHONE digital service or other private line data facilities. A Data-Only Off-Premises Extension uses an MTDM located on-premises. Communication with the remote data equipment is accomplished through the private line facility linking the on-premises MTDM and the remote data equipment.

The Trunk Data Module and DCE or DTE constitute a digital data endpoint. Data calls to this type of data endpoint can be placed using Voice Terminal Dialing or Data Terminal (Keyboard) Dialing. Since there is no voice terminal at the remote site, data calls can be originated from the remote data terminal using Keyboard Dialing only. If computer-generated dialing is used on calls, it must follow the Keyboard Dialing protocol.

Considerations

The systems have the capability for Data-Only Off-Premises Extensions to allow for data calls to remote DCE using DATAPHONE digital service or other private line data facilities.

Data-Only Off-Premises Extensions provides digital data endpoints located off-premises through a Trunk Data Module located on-premises. Communications to or from this Trunk Data Module (and the associated off-premises equipment) must be through an on-premises Processor Data Module or Digital Terminal Data Module. Communications between two Trunk Data Module are not supported. Likewise, Modem Pooling, which is conceptually similar to a Trunk Data Module, cannot be used on calls to or from a Data-Only Off-Premises Extension.

Interactions

The following features interact with the Data-Only Off-Premises Extensions feature.

- Voice Terminal Dialing

  An on-premises multiappearance voice terminal may have a Data Extension button associated with the Trunk Data Module used for a Data-Only Off-Premises Extension. The voice terminal user and the remote data equipment user share control of the data module. Actions of the user at the voice terminal may affect the remote user.

  — One-Button Transfer to Data

    The on-premises voice terminal user can transfer a call to the Data-Only Off-Premises Extension. The Data Extension button on the voice terminal lights and the Call in Progress lamp on the data module lights during an established data call.

  — Return-to-Voice

    If a data call has already been established, the voice terminal user may press the associated busy Data Extension button to transfer the call to the voice terminal. The data module associated with the Data Extension button is disconnected from the call. The Call in Progress lamp on the data module goes dark.

  — Data Call Preindication

    The multiappearance voice terminal user presses the idle associated Data Extension button to reserve the data module. The data module is then busy to all users except the Preindicating user, including the remote user. When the data module is reserved, the lamp associated with the Data Extension button winks at the preindicator's voice terminal and lights at any other associated voice terminals. A remote user receives the BUSY message when attempting to originate a call.

Administration

Data-Only Off-Premises Extensions is assigned on a per-line basis by the System Manager. The following item requires administration:

- Digital Line Circuit Pack — Assign the associated data module to a vacant port.

Hardware and Software Requirements

Requires a Trunk Data Module and one port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law). No additional software is required.

DCS Alphanumeric Display for Terminals

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows calls to or from terminals equipped with alphanumeric displays to have transparency with respect to the display of call-related information.

Calling Name Display is the presentation, on the *called* terminal's alphanumeric display, of the name of the party who originated the call. Called Name Display is the presentation on the *originating* terminal's display of the name of the party to whom the call is directed. Both displays provide more useful and precise information than such general identifiers as a trunk group name or an extension number.

The transparency allows calling and called name information, plus miscellaneous identifiers (IDs) to be sent from a terminal on one node to a terminal on another node. Transparency in this area is limited by the type of systems at the endpoint nodes and at the intermediate node, if any.

Considerations

DCS Alphanumeric Display for Terminals gives the user considerable call handling capabilities by displaying call related information on calls to and from other DCS nodes.

Calls to and from a DEFINITY system Generic 3 in a DCS network have Calling/Called Name Display transparency under the following conditions:

- The other party is at another DEFINITY system Generic 3 and the tandem node is a System 75 Version 3 or later, DEFINITY system Generic 1, DEFINITY system Generic 3, System 85 Release 2 Version 2 or later, or a DEFINITY system Generic 2.1.
- The call is not routed through a tandem System 85 Release 2 Version 1 or Enhanced DIMENSION PBX node. (Such calls will display only the extension number of the calling or called party.)

On outgoing DCS calls, display of the called name may be delayed for a few seconds until the required information arrives from the distant node. The called name display only works between DEFINITY systems Generic 1, Generic 3 systems, and System 75s.

Within the same DEFINITY system Generic 1 or Generic 3 node in a DCS, complete transparency of Calling and Called Name Display exists.

Interactions

The following DCS configurations provide transparency of alphanumeric display information:

- Networks of two or more DEFINITY switches (Generic 1 or Generic 3i) with a System 75 Version 3 or later, Generic 2, System 85 Release 2 Version 2 or later, or a Generic 2.1 as an intermediate node
- A DEFINITY system Generic 1 connected to a System 85 Release 2 Version 2 or later, or a Generic 2.1

Configurations in which DEFINITY system Generic 1s are connected to or through a System 85 Release 2 Version 1 or an Enhanced DIMENSION PBX are not covered because these nodes do not provide display transparency.

If both DCS and ISDN-PRI features are provided with a system, the ISDN-PRI display information is displayed in DCS format.

The following features have transparency with respect to Calling and Called Name Display and miscellaneous ID. If the display for a DCS call differs at all from the display for a call between terminals at the same system, the difference is noted. Refer to the *DEFINITY Communications System Generic 1 and Generic 3 Voice Terminal Operations*, 555-230-701, for detailed descriptions of call information displays.

- Automatic Callback

  Complete display transparency.

- Call Coverage

  At the calling terminal, the miscellaneous id ‘‘cover’’ is not displayed.

- Call Forwarding

  When a system user calls a party on a different node in the DCS and the call is forwarded, the miscellaneous ID ‘‘forward’’ is not displayed. At the covering (forwarded-to) user’s terminal, only the calling party’s name is shown; the called party’s name is not displayed.

- Call Park

  When a DCS call between a local system user and a user on another node is parked by the remote user, the miscellaneous ID ‘‘park’’ is not displayed at the local terminal.

- Call Pickup

  When a DCS call from a system user to another node is answered by way of Call Pickup, the miscellaneous ID ‘‘cover’’ is not displayed at the caller’s terminal.

- Call Waiting

  When a DCS call from a system user to another node is waiting at the called terminal, the miscellaneous ID ‘‘wait’’ is not displayed at the caller’s terminal.

- CAS

  When a user dials the extension for CAS, a RLT is seized or the caller is queued for an RLT. The caller’s terminal will display the trunk group identifier, such as OPERATOR.

- Conference

  When a DCS call is conferenced either at a remote node or at the local system, all DCS Calling and Called Name Display transparency is lost to local system users. If all parties drop out except for a local user and another DCS user, the local user’s terminal will display the trunk group identifier.

- DDC/UCD

  Complete display transparency.

- Internal Terminal-to-Terminal Calling

  Complete display transparency.

- Transfer

  When a DCS call is transferred at a remote node to a user on any node, all DCS Calling and Called Name Display transparency is lost to users on the local system.

Administration

DCS tie trunk groups between nodes must be administered by the System Manager with the Outgoing Display disabled. This enables the called party’s name to be displayed at the calling terminal.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Attendant Control of Trunk Group Access

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows an attendant at any node in the DCS to exercise control over an outgoing trunk group at an adjacent node in the cluster.

Each attendant console has 12 Trunk Hundreds Select buttons to be used with the Attendant Direct Trunk Group selection feature. Each button allows the attendant direct access to an outgoing trunk group by merely pressing the button assigned to that trunk group. Each of the 12 buttons has a Busy lamp which lights when all trunks in the associated trunk group are busy. On a basic console, six of these buttons have two additional lamps that are used for Attendant Control of Trunk Group Access. On an enhanced console, all 12 buttons have the additional lamps. The two additional lamps are as follows:

- Warn (warning) lamp

  Lights when a preset number of trunks are busy in the associated trunk group.

- Cont (control) lamp

  Lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group.

Attendant control of a remote trunk group in the DCS network is activated by pressing the Cont Act button followed by the desired Remote Trunk Hundreds Select button. Then the initiating node sends a message to the remote node where the trunk group to be controlled resides. The message indicates that control of that trunk group has been initiated.

When the remote node receives the control activation message from the initiating node, it has four seconds to send a reply message back to the initiating node if control of the remote trunk group can be activated. A confirmation message will be sent to the initiating node and the Cont lamp at the corresponding Trunk Hundreds Select button is lighted at the remote node if control of the remote trunk group can be activated. An error message is sent to the attendant at the initiating node if the trunk access code is invalid, if the trunk group is already controlled, or if the remote node is a System 85 or Enhanced DIMENSION PBX and the attendant does not have a Trunk Hundreds Select button with Cont lamp for that trunk group.

When a trunk group is controlled in a DCS environment, calls to the trunk group by anyone other than an attendant are routed to the local attendant at the node where the trunk group resides. If that node does not have an attendant, the call is routed to a CAS main attendant or an attendant at a location arranged for Inter-PBX Attendant Calls. However, if CAS or the Inter-PBX Attendant Calls feature is not provided, the party attempting to call on the controlled trunk receives intercept tone.

A detailed description of "Centralized Attendant Service (CAS)" and "Inter-PBX Attendant Calls" is given elsewhere in this chapter.

Considerations

DCS Attendant Control of Trunk Group Access allows attendants to obtain control of access to specific trunk groups at any node in the DCS network. This allows the attendant to monitor the use of the controlled trunk group.

There must be direct DCS tie trunk connections between the initiating node and the remote node where the trunk group to be controlled originates. Otherwise, control of remote trunk groups is denied.

If the remote node (where the trunk group to be controlled resides) is a

System 75, Generic 1, or Generic 3, it is not necessary for that node to have an attendant console with corresponding three-lamp Trunk Hundreds Select button. However, if the remote node is a System 85, Generic 2.1, or Enhanced DIMENSION PBX, control of the trunk group is not allowed unless an attendant at that node has a corresponding three-lamp Trunk Group Select button.

The attendant must use the Remote Trunk Hundreds Select button to directly access the controlled remote trunk group. If an attendant controls a remote trunk group, and that attendant dials the trunk access codes of the DCS tie trunk and the controlled remote trunk group, the call is routed to the attendant at the node where the trunk group resides.

If Attendant Control of Trunk Group Access is activated, and no attendant is assigned, or the attendant is later removed, calls to a controlled trunk group route to the attendant queue.

Interactions

The following features interact with the DCS Attendant Control of Trunk Group Access feature.

- DCS Attendant Display

  When a user attempts to access a controlled trunk group and is routed to the local attendant, the display shows the reason the call was redirected. If the call is routed via CAS or the Inter-PBX Attendant Calls feature, the display does not show the reason the call was redirected.

- UDP

  DCS tie trunks should not be attendant controlled. This would result in all UDP calls on the controlled tie trunk being routed to the controlling attendant instead of to the desired destination.

Administration

The ability of an attendant to control access to a remote trunk group is dependent on the administration by the System Manager of Trunk Hundreds Select buttons for remote trunk groups in the DCS.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Attendant Direct Trunk Group Selection

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows attendants at one node to have direct access to an idle outgoing trunk at a different node in the DCS.

A Trunk Hundreds Select button can be assigned to access a trunk group at the local node or a trunk group at a remote node. A Trunk Group Select button assigned to access a remote node is referred to as a remote Trunk Hundreds Select button. Pressing a remote Trunk Group Select button has the same affect as dialing the tie trunk group access code for the remote node and the trunk access code of the selected trunk.

DCS Attendant Direct Trunk Group Selection functions the same as the regular "DCS Attendant Direct Trunk Group Selection" feature (fully described elsewhere in this chapter). The only difference is an attendant can access a trunk group at a remote node.

Considerations

With "DCS Attendant Direct Trunk Group Selection", an attendant can have faster access to trunk groups at remote nodes. There is no need to look up trunk access codes, because the press of a button connects the attendant to the desired trunk group.

There must be a direct DCS tie trunk connection between the initializing node and the remote node where the trunk group to be accessed originates. Otherwise, access to the remote trunk group is denied.

Interactions

None.

Administration

The system manager must assign:

— A remote Trunk Hundreds Select button.

— The tie trunk access code to the remote node.

— The trunk access code of the remote trunk group.

The trunk access codes (TACs) must be three characters or less.

In addition to the 12 fixed Trunk Hundreds Select buttons on each attendant console, feature buttons may be assigned remote Trunk Hundreds Select buttons,

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Attendant Display

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Provides some transparency with respect to the display of call-related information.

Calls to and from a DEFINITY switch in a DCS environment have Calling/Called Party Identification transparency under the following conditions:

- The other party is at another DEFINITY system Generic 1, Generic 3, or System 75, and the intermediate node is a DEFINITY system Generic 1, Generic 2.1, Generic 3, System 75 Version 3 or later, or a System 85 Release 2 Version 2 or later.
- The other party is at a System 85 Release 2 Version 2 or later, or a DEFINITY system Generic 2.1.
- The call is not routed through an intermediate System 85 Release 2 Version 1 or Enhanced DIMENSION PBX node. (Such calls will display only the extension number of the calling or called party.)

A detailed description of the "Attendant Display" feature is given elsewhere in this chapter.

Considerations

DCS Attendant Display gives the attendant considerable call handling capabilities by displaying call related information on calls to and from both local and remote nodes. This detailed information can be very useful in processing calls.

CORs for a DEFINITY switch may not correspond to those used by an Enhanced DIMENSION PBX, System 85, or DEFINITY system Generic 2.1. Therefore, if the DCS network contains nodes other than DEFINITY system Generic 1 or Generic 3, the display CORs may be misinterpreted. If it is important that certain CORs between various systems correspond with each other, those CORs should be administered accordingly.

On outgoing calls, the display of called party information may be delayed a few seconds until the required information arrives from the remote node. The called party information is displayed only if both nodes are DEFINITY system Generic 1 or System 75.

DCS tie trunks between nodes must be administered with the Outgoing Display enabled. This enables the called party's name to be displayed at the calling attendant's display.

Interactions

When both ISDN and DCS display information, or only DCS display information, are received, the switch will display the DCS display information in the DCS format. If ISDN display information is received, and no DCS display information is received, then the ISDN display information is displayed in the ISDN formats.

Administration

The administration required for DCS Attendant Display is the same as that required for the "Attendant Display" feature. This information is given elsewhere in this chapter.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Automatic Callback

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows a user at one node to make an automatic callback call to a user at another node in the DCS.

A DCS Automatic Callback call can be activated from a voice terminal at one node to a voice terminal at another node in the same way as if at a local node under the following conditions.

- If the called party is at a System 85, DEFINITY system Generic 2, or Enhanced DIMENSION PBX node, the callback call can only be activated if the called node is returning busy tone or special audible ringback.
- If the called party is at a DEFINITY system Generic 3, Generic 1 or System 75 node, the callback call can be activated if the called node is returning busy tone, Call Waiting ringback tone, or ringback tone.
- The calling party must disconnect within six seconds after hearing the confirmation tone for Automatic Callback activation.

The callback of the calling or called parties is as follows when a callback call has been made to a user at another node:

- When the calling party answers the callback call, and no tie trunk to the called party's node is available, Automatic Callback is reactivated toward the called party. The calling party hears confirmation tone instead of ringback when this occurs.
- If the calling party is on a System 85, DEFINITY system Generic 2, or Enhanced DIMENSION PBX node and is unable to receive the callback call (for example, a busy single-line voice terminal without Call Waiting), Automatic Callback is reactivated by the calling party's node. If the calling party is on a DEFINITY system Generic 3, Generic 1, or System 75 node and is unable to receive the callback call, the callback call is canceled.
- If the called party is unable to receive the callback call because they are busy again (for example, the called party goes on hook and then off hook immediately to place another call), the calling party hears busy tone again and can choose to reactivate Auto Callback, if desired.

A detailed description of the Automatic Callback feature is found on page 3-192.

Considerations

DCS Automatic Callback eliminates the need for voice terminal users to continuously redial busy or unanswered calls to voice terminals within the DCS network.

An Automatic Callback request is canceled automatically if the called party does not become available within 40 minutes, or if the calling party does not hang up within six seconds after activating Automatic Callback.

DCS Automatic Callback does not work on the last trunk between nodes. Thus, if *''n''* trunks are provided, there can be up to *''n - 1''* Automatic Callback calls.

Interactions

The following features interact with the DCS Automatic Callback feature.

- Attendant Control of Trunk Group Access and DCS Attendant Control of Trunk Group Access

  Automatic Callback cannot be activated if the call uses a controlled trunk group.

- Call Forwarding All Calls and DCS Call Forwarding All Calls

  Automatic Callback calls cannot be activated on a voice terminal at a DEFINITY switch or System 75 node that has Call Forwarding activated.

Administration

The administration required for DCS Automatic Callback is the same as that required for the "Automatic Callback" feature. This information is given elsewhere in this chapter.

Hardware and Software Requirements

DCS interface and DCS software are required.

DCS Automatic Circuit Assurance (ACA)

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows a voice terminal user or attendant at a node to activate or deactivate ACA referral calls for the entire DCS network. This transparency also allows the referral calls to be generated at a node other than the node that detects the problem.

If referral calls are generated at a node for one or more remote nodes, the remote nodes are notified when ACA referral is activated or deactivated.

If referral calls are generated at a remote node for a DEFINITY system Generic 3 node, the DEFINITY system Generic 3 node is notified when ACA referral is activated or deactivated at the remote node. This notification is accomplished via the ACA button located on the attendant console or voice terminal at the DEFINITY system Generic 3 node. The lamp associated with the ACA button lights when ACA referral is activated and goes dark when ACA referral is deactivated. The ACA button serves no other purpose when a remote node generates the DEFINITY system Generic 3 referral calls.

A detailed description of the "Automatic Circuit Assurance (ACA)" feature is given elsewhere in this chapter.

Considerations

The DCS Automatic Circuit Assurance feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time.

Interactions

None.

Administration

DCS Automatic Circuit Assurance requires that the System Manager administer whether ACA referral calls are to be local, remote, or primary:

- If administered as local, referral calls are generated at the Generic 3 node for the Generic 3 node.

- If administered as remote, referral calls are generated at a remote node for the Generic 3 node. In this case, the remote node identification must also be entered.
- If administered as primary, referral calls are made at the Generic 3 node for a remote node.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Busy Verification of Terminals and Trunks

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows attendants and multiappearance voice terminal users to make test calls to voice terminals and trunk groups that are located at other nodes within the DCS.

Attendants and multiappearance voice terminal users can busy verify voice terminals at a remote location. This is done by first pressing the Verify button and then entering the desired UDP extension number. The verification then continues the same as if the voice terminal being verified is on the same node.

Multiappearance voice terminal users can busy verify an adjacent at a remote location. This is done by first pressing the Verify button, then dialing the trunk access code of the tie trunk group to the remote node, pressing the Verify button a second time, and then entering the desired trunk access code and the trunk group member number to be verified. The verification of the trunk then continues as if the trunk being verified is on the same node.

Attendant operation is the same except a Trunk Hundreds Select button can be used to access the tie trunk to the remote node. A detailed description of the "Busy Verification of Terminals and Trunks" feature is given elsewhere in this chapter.

Considerations

DCS Busy Verification of Terminals and Trunks provides an easy method of checking the working condition of extensions and trunks at remote nodes.

Interactions

If the Trunk Identification by Attendant feature is used during busy verification of a trunk (Trunk ID button is pressed), the trunk access code and trunk group member number of the DCS tie trunk being used is displayed.

DCS Busy Verification of Terminals and Trunks transparency is lost if the routing pattern is administered to not delete the RNX and the AAR prefix is inserted on the terminating switch trunk group. The voice terminal display at the terminating switch displays only **a=station name**. The “Extension” field is left blank.

Administration

The administration for DCS Busy Verification of Terminals and Trunks is the same as that for the "Busy Verification of Terminals and Trunks" feature, which is fully described elsewhere in this chapter.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Call Coverage

Feature Availability

This feature is available in G3V4. It is not available in a mixed system of G3V4 and pre-G3V4 systems.

Description

DCS Call Coverage provides the additional DCS messaging required to allow calls to be covered by coverage points on remote systems providing there is a DCS signaling link (BX.25 or ISDN PRI) for the trunk groups to which the covered calls are routed. Thus, with DCS Call Coverage, calls to an extension on one system can be covered by extensions administered as coverage points on remote systems in a similar way to how Call Coverage functions within a single system. It provides transparency across systems for the Call Coverage feature.

NOTE:
DCS Call Coverage should not be confused with coverage to a remote AUDIX.

DCS Call Coverage appears to the user to operate the same as Call Coverage with the following exceptions:

- Coverage Answer Groups across nodes are not supported.
- Under some conditions, a call will follow the coverage point's coverage path. See "Operation Under Error Conditions" on page 3-618 for specific information.
- Displays will differ in some cases. See "Displays" on page 3-619 for more details.
- DCS Call Coverage does not support Coverage Call Back from a remote node. See "Feature Interactions" for more information.
- Under some conditions consult will not work properly. See "Operation Under Error Conditions" later in this description for more information.

In releases prior to G3V4, transparent call coverage to remote systems with normal call coverage functionality was limited to a single point of coverage on the remote system. Furthermore, that coverage point had to be an AUDIX, DEFINITY AUDIX, or Intuity. Note that you could cover to a remote point with the Call Coverage Off Premises feature, but this does not have the normal call coverage functionality.

Feature Applications

Consider the following figure

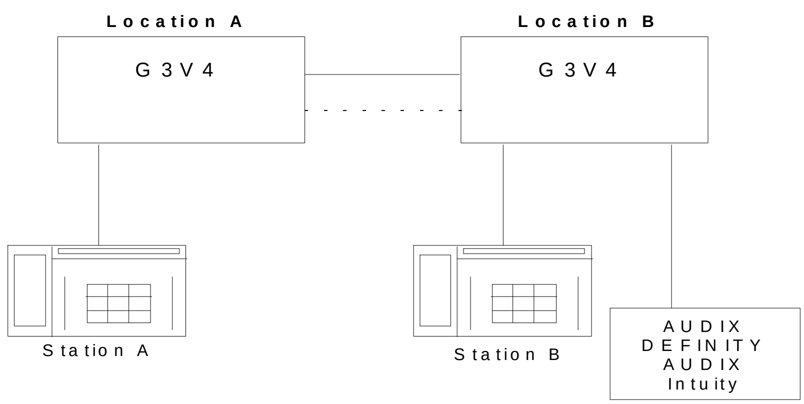


Figure 3-4. Applications for DCS Call Coverage

The following list describes some of the applications for DCS Call Coverage not otherwise possible without this feature:

- A user spends time in multiple locations. An account executive, for example, might spend a lot of time at locations A and B. He or she might want calls to station B, for example, to be covered first by station A and then by AUDIX, DEFINITY AUDIX, or Intuity.
- A user moves from location A to location B. In this case, the user might want calls to station A to be covered first at station B then at AUDIX.
- A secretary is located in different locations. A user at station B might want calls to first be covered by a secretary at station A and then by the AUDIX at location B.

Previous Feature Operation

The following figure illustrates the extent of call coverage over DCS tie trunks in pre G3V4 systems. Previous to G3V4 there was only one case when a call would cover to a point on a remote switch, with normal call coverage functionality, which was when covering to a remote AUDIX as described below.

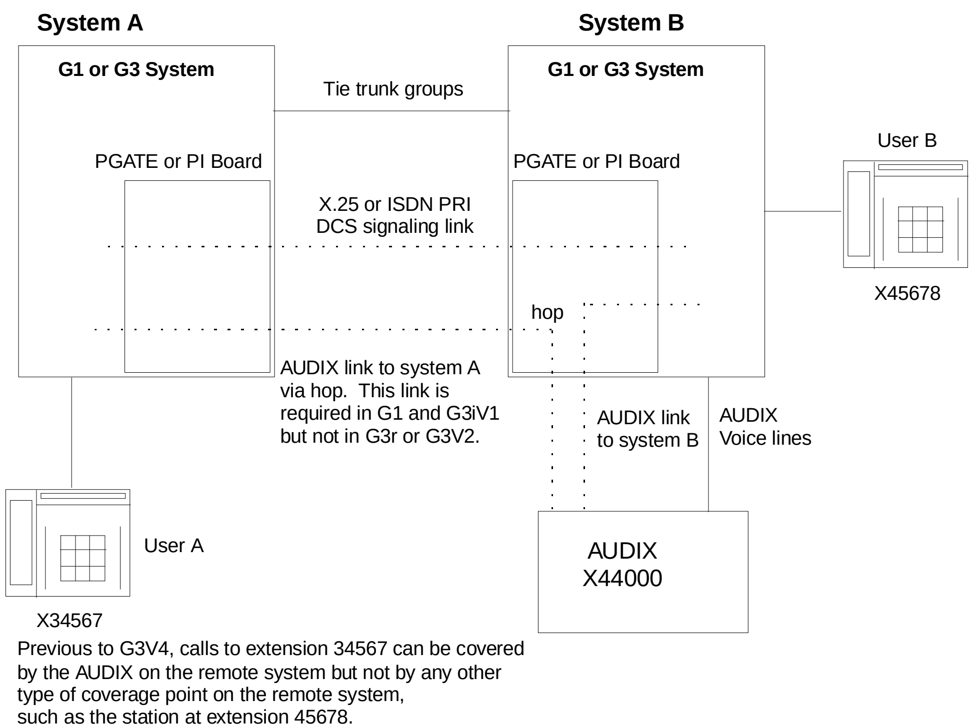


Figure 3-5. DCS AUDIX Coverage to Remote System Prior to G3V4

In this scenario, calls to extensions on system A can be covered by the AUDIX on system B as follows:

- A call comes into the station at extension 34567. After the administered number of rings, the call goes to coverage.
- If the coverage point is an AUDIX hunt group with **rem-audix** in the "Message Center" field and **44000** in the "AUDIX Extension" field on page 2 of the "Hunt Group" form, the AUDIX, if it is available, will cover this call with user A's personal greeting. If the AUDIX is not available, no further coverage points will be attempted. If the trunk between the two systems is busy, the next coverage point, which must be on system A, will be tried.
- The connect message sent to AUDIX by System B gets the principal's extension (34567) from a DCS message sent on the DCS link between system A and system B.

- After AUDIX answers the covered call, it sends a message to system A via an AUDIX signaling link to system A lighting user A's message lamp.
- Since a physical link does not exist between the AUDIX and system A, the signaling link must hop from the system B-to-AUDIX BX.25 link to the system B-to-system A BX.25 link. In G1 and G3iV1, this link between AUDIX and system A must be present for the call to be covered. In G3rV1, G3V2, and G3V3 the call will be covered without this link present but the message lamp will not be updated.

New Feature Operation

The DCS Call Coverage feature for G3V4 adds the capability to cover to points other than AUDIX on a remote system. It does this by providing DCS messaging that provides transparency for the G3V4 Call Coverage feature. The following figure illustrates some of the possibilities associated with DCS Call Coverage

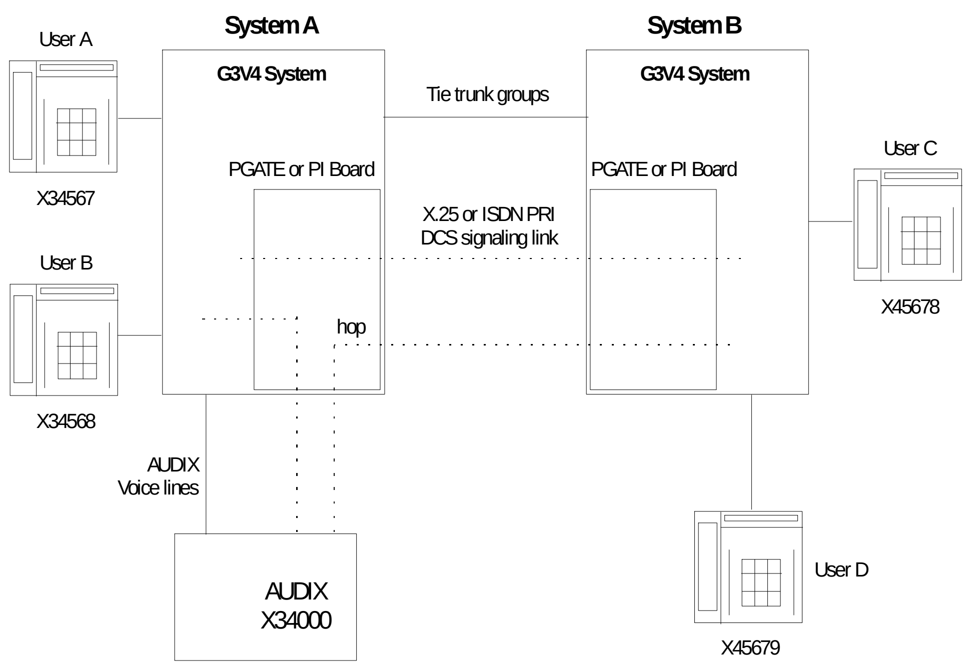


Figure 3-6. G3V4 DCS Call Coverage

In this scenario using DCS Call Coverage, calls to user A can be covered first by user B, then by user C, and then finally by the AUDIX on system A. Alternatively, calls could be covered by user C, then user B, and so on. If the principal party answers after the call has gone to coverage and the coverage point has answered, the calling party, the principal, and the coverage point are conferenced together. In this same case, if the coverage point has not answered, the call to the coverage point is dropped and the calling party and principal are connected.

Detailed Operation

To illustrate how DCS Call Coverage works in more detail, consider the following scenario

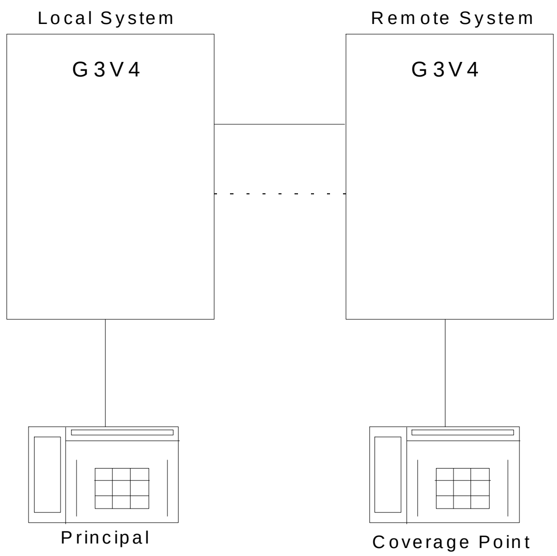


Figure 3-7. Detailed Operation Scenario

The following sections describe the DCS message flow for possibilities in this scenario.

Operation under Normal Conditions. In a normal DCS Call Coverage scenario where the DCS link is up, a DCS trunk group is available, and DCS Call Coverage is activated on the local and remote systems, DCS Call Coverage works as follows:

1. A call comes into the principal.
2. The principal does not answer within the administered number of rings or has Send All Calls activated.

3. The call is redirected to the first coverage point, which is on the remote system. Also, at this time the DCS distinctive ringing message is sent to the coverage point along with two messages conveying the caller's name and the principal's name.
4. The coverage point rings with internal ringing if the caller is on a DCS node, or with external ringing otherwise.
5. The covering user answers and receives the calling party's name and principal's name on the display if equipped.

Several other scenarios can occur deviating from this normal situation, as follows:

- A call to the principal redirects to the remote coverage point, which is unavailable. The coverage point is considered unavailable in the following cases:
  - The coverage point is not a valid extension, QDN, or VDN.
  - The coverage point is busy with no hunting, forwarded, or has send all calls activated, or activates send all calls after ringing.
  - The coverage point has no staffed agents or an invalid vector.

  In the case of an unavailable coverage point, the local system learns of the availability status either through a time-out or from a message from the remote system. When the local system determines that the coverage point is unavailable, it tries the next coverage point. If the last coverage point is unavailable, the previous coverage point rings until answered or until the caller hangs up. If only one coverage point exists in the path and it is unavailable, the principal's station rings until answered or until the caller hangs up.
- A call to the principal is forwarded and the forwarded-to extension is not available. In this case, the first coverage point in the principal's path will be tried. Note that the coverage does not follow the forwarded-to extension's coverage path.
- A call to the principal redirects to the remote coverage point, which answers. Subsequently, the principal goes off hook. In this case, the local system bridges the principal onto the call between the calling party and coverage point creating a conference among the three. The principle receives the call on the same call appearance as the original call.
- A call to the principal redirects to the remote coverage point. While the remote coverage point is ringing, the principal answers the call. In this case the call is not cut through to the coverage point. Instead, ringing and ringback is removed from the coverage point and the call is cut through to the principal.

Operation Under Error Conditions. The following table describes the operation of DCS Call Coverage under error conditions

Table 3-52. Error Conditions

Error Condition	Action
DCS link not up. or DCS trunk is not available. or DCS Call Coverage feature is not activated on the remote system.	The call is routed to the remote coverage point. If the call is answered, it is treated as Call Coverage Off Premises (also called Remote Call Coverage.) If the call is redirected at the remote coverage point before the DCS SRI expires, the remote point's path will be followed. If the call is not answered within the DCS SRI time-out period, the next coverage point is tried with DCS Call Coverage from the local system.
All trunks to the remote system, DCS or otherwise, are busy	The next coverage point is tried with DCS Call Coverage from the local system.

In addition, call consult operates as follows when the DCS link is down: If user A calls user B but the call covers to user C, then if user C consults back to user B, user B receives the consult call on the next call appearance.

User Interface

The user interface for DCS Call Coverage consists of station alerting and terminal displays, including hospitality displays

Station Alerting. Station alerting operates the same as in Call Coverage, as follows:

- If the call to the principal originated from a station within the DCS network, the coverage point receives internal ringing.
- If the call to the principal originated from a trunk, which means from a phone on a switch outside the DCS network, the coverage point receives external ringing.
- If the call to the principal originated from an attendant within the DCS network, the coverage point receives external ringing.

Displays. Displays at the coverage point work the same as for Call Coverage with the exceptions caused by DCS. In general, the display at the coverage point takes the form

a=<Calling Party Name> <to> <Principal Name> <reason for redirection flag>

For call coverage, this format is always true because the system has all the information it needs from the calling party, principal, and coverage point. Under most circumstances, DCS also has all the information it needs, providing it in messages. In some DCS circumstances, however, some information can be lacking or substituted with other information. Following are the formats for displays on the coverage point's terminal under these special conditions

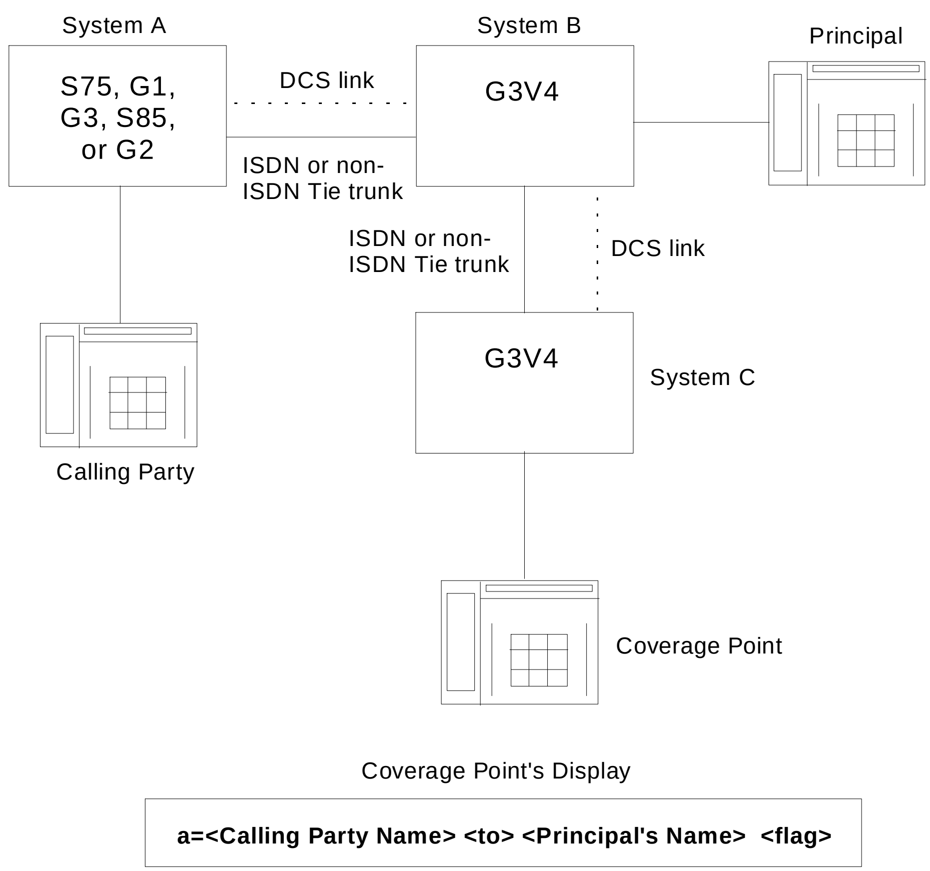


Figure 3-8. Display — DCS Trunks End-to-End

In the Figure 3-8, a call is made from the station on system A to the principal on system B and covered by the station on system C. The tie trunks between A and

B and B and C are DCS trunks transported over any kind of facility (ISDN or non-ISDN).

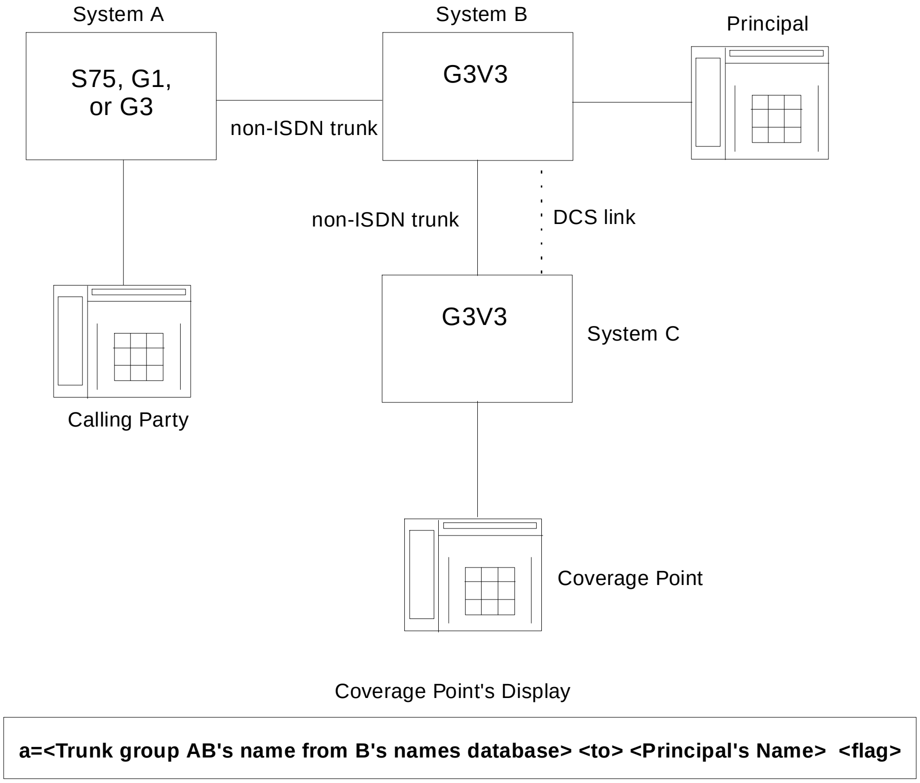


Figure 3-9. Display — Non-DCS, Non-ISDN Trunk to DCS Trunk over Non-ISDN Trunk

In Figure 3-9, a call is made from the station on system A to the principal on system B and covered by the station on system C. The tie trunk between A and B is not a DCS trunk and is not an ISDN PRI trunk. The tie trunk between B and C is a non-ISDN PRI DCS trunk.

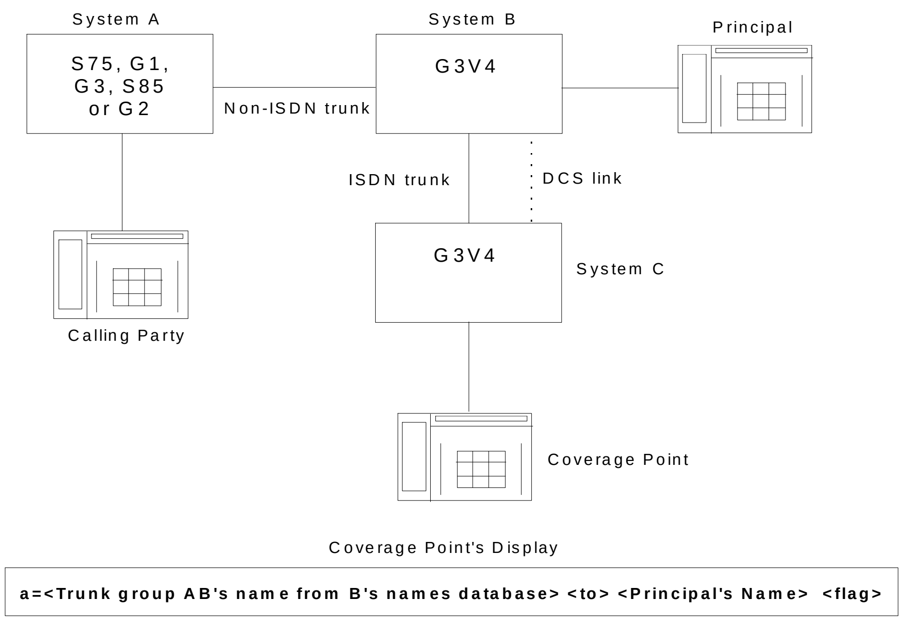


Figure 3-10. Display — Non-DCS, Non-ISDN Trunk to DCS Trunk over ISDN Trunk

In Figure 3-10, a call is made from the station on system A to the principal on system B and covered by the station on system C. The tie trunk between A and B is not a DCS trunk and is not an ISDN PRI trunk. The tie trunk between B and C is a DCS ISDN PRI trunk.

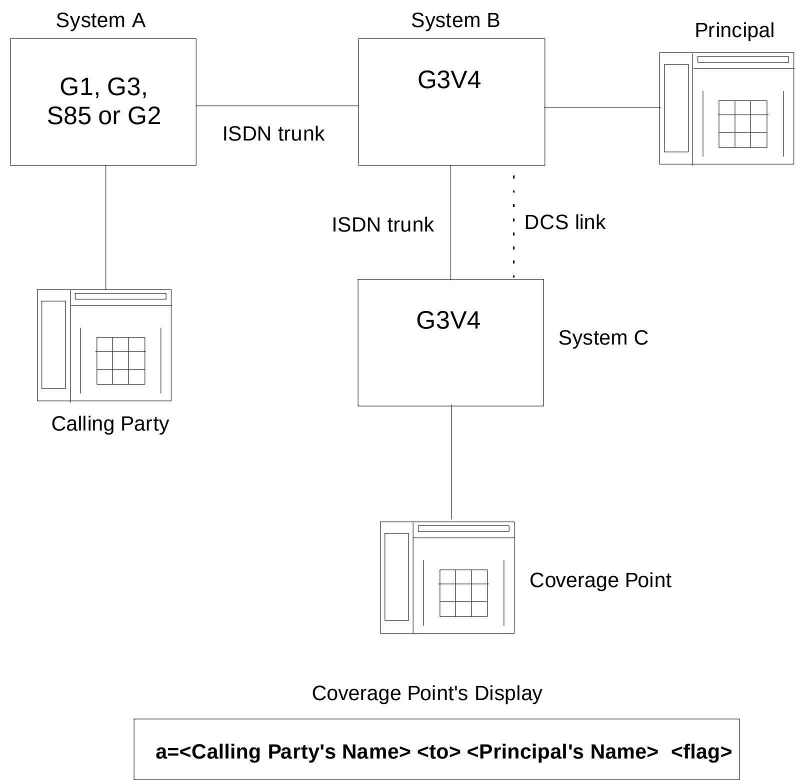


Figure 3-11. Display — Non-DCS ISDN Trunk to DCS ISDN Trunk

In Figure 3-11, a call is made from the station on system A to the principal on system B and covered by the station on system C. The tie trunk between A and B is not a DCS trunk but is an ISDN PRI trunk. The tie trunk between B and C is a DCS ISDN PRI trunk.

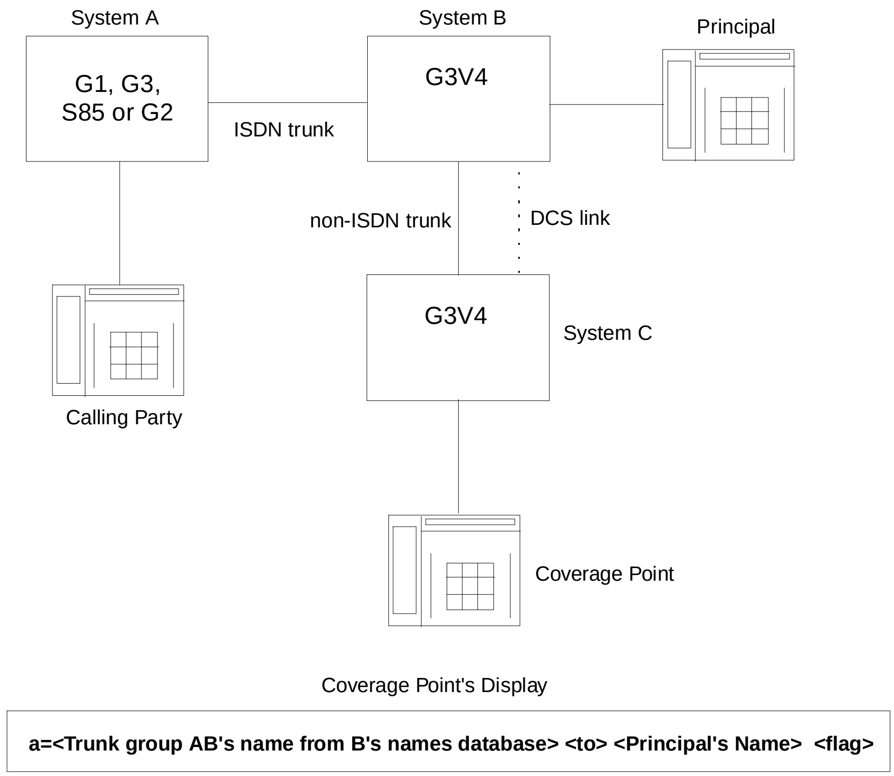


Figure 3-12. Non-DCS ISDN Trunk to Non-ISDN DCS Trunk

In Figure 3-12, a call is made from the station on system A to the principal on system B and covered by the station on system C. The tie trunk between A and B is not a DCS trunk but is an ISDN PRI trunk. The tie trunk between B and C is a DCS trunk but not an ISDN PRI trunk.

If the calling party is a trunk on system A instead of a station, the displays are the same as above, substituting the calling trunk group’s names database for the station’s names database. The exception to this is when system A is a System 85 or G2, as follows:

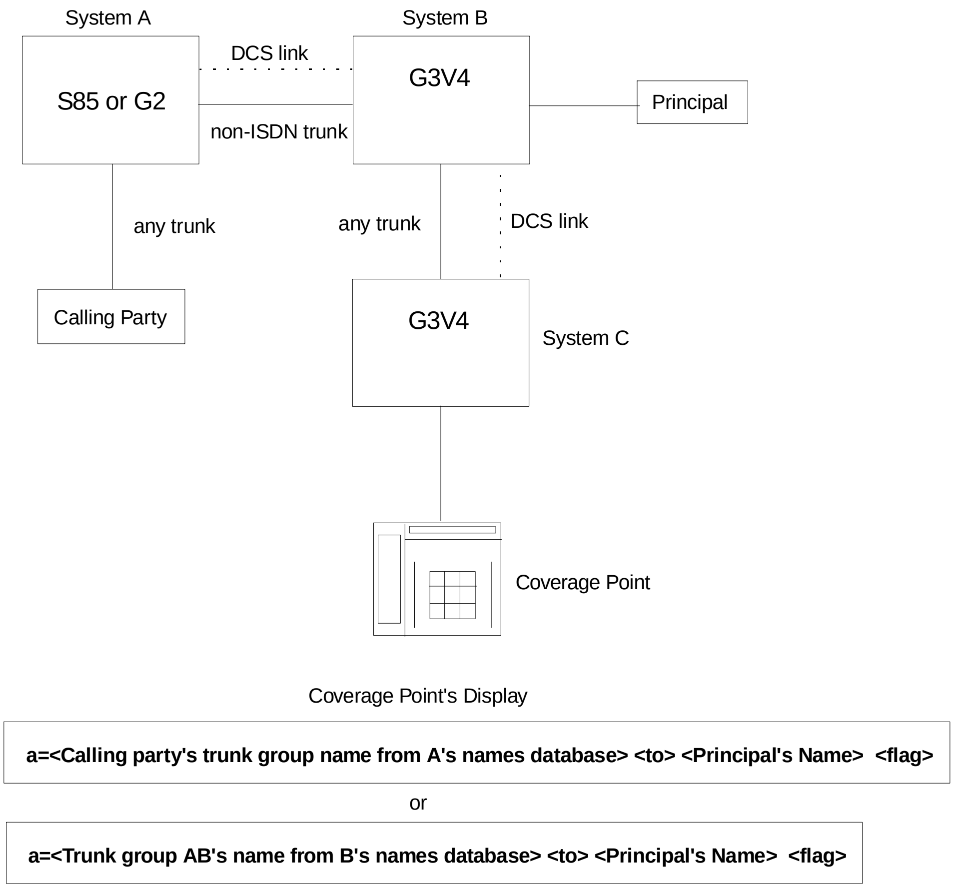


Figure 3-13. Display — Call Initiated by a Trunk on System 85 or G2

The display in this case depends on the administration on the G2. If the call is routed to trunk group AB via the internal dial plan (PROC 354 word 2), the G2 sends a DCS name message for the calling trunk and the first display results. If the call is routed directly to a pattern without going through the internal dial plan, however, a name message is not sent, resulting in the second display. If trunk groups AB and BC are both ISDN PRI trunk groups, however, the first display will result because the coverage point will display the contents of the ISDN PRI display information element.

The situations described above assume outgoing trunk group displays are turned off in the “Trunk Group” form, that the “Send Name” field is **yes** in the” ISDN Trunk Group” form, that the name message and display IEs have contents, which are taken from the station or trunk names databases, and that these messages are sent and received.

Hospitality Displays. If the hospitality feature is in use, the display rules can vary from those shown above depending on administration. The displays on terminals follow the same rules as above except for one case as described in the following list.

- The hospitality feature is enabled on the calling party's system, the principals' system, and on the coverage point's system on page 2 of the "System Parameters Customer Options" form.
- The client room option is set to **y** on the "Class of Service" form for the calling party's and the principal's stations.
- The "Display Client Redir?" field is set to **n** on the "Coverage Point's Station" form.

If the above conditions exist, the coverage point displays the following:

a=<Calling party's name from A's names database><to><Principal's extension>

In most hospitality applications, the extension is administered as the room number. If this is not the case, however, the administered room number can be displayed instead of the extension. This is accomplished by doing the following:

- Set the "Display Room Information?" field to **y** in the "Hospitality-Related System Parameters" form.
- Enter up to five characters in the "Room" field in the "Principal's Station" form.

Interactions

DCS Call Coverage has the same interactions as Call Coverage plus interactions with the following features. For information on the Call Coverage interactions, see the "Call Coverage" feature description in this manual.

- Call Coverage Off Premises

  DCS Call Coverage interacts with Call Coverage Off Premises as follows. If the coverage point is a non-UDP number in the remote call coverage table, Call Coverage Off Premises is applied to the call rather than DCS Call Coverage, even if a DCS link exists to the remote system.

- Coverage Answer Groups

  DCS Call Coverage to Coverage Answer Groups on remote systems are not supported by DCS Call Coverage. Coverage answer groups cannot be administered on a system other than the principal's system.

- Coverage Call Back

  DCS Call Coverage does not support Coverage Call Back from a remote node. This is because the coverage point's system has no way of knowing whether the caller can respond to Leave Word Calling. For example, the coverage point's system does not know whether the calling party is a trunk or a station.

- Displays

  DCS Call Coverage interacts with displays as follows. Displays on the coverage point's terminal can appear different than they do with Call Coverage in the following situations:

  — When the call from the calling party to the principal or the redirected call to the coverage point travel over ISDN PRI trunk groups.

  — When the calling party is on a System 85 or Generic 2.

  — When the DCS name message is not received by the remote (coverage point's) system.

  For more detail see the Displays section of this feature description.

- Goto Cover

  Goto Cover is not supported over DCS and therefore is not supported with DCS Call Coverage.

- Leave Word Calling (LWC) Back to Principal

  With DCS Call Coverage a covering user on a different node cannot press their LWC button to leave a message for the principal to call the covering user.

- Queuing

  DCS Call Coverage interacts with queuing in the following way. If a call is queued to a coverage point, such as a queue to a hunt group or an ACD split, and the queue is not full, the call will stay in the queue without subsequent redirection until answered or until the caller hangs up.

Interworking

DCS Call Coverage contains considerations associated with interworking with ISDN PRI and with non-G3V4 systems. The interworking issues with ISDN PRI are actually issues related to DCS, not to DCS Call Coverage, but they do have consequences for how displays appear at the coverage point's display terminal.

The following list describes the interworking between DCS and ISDN PRI:

- When an incoming call on a DCS trunk is routed to an outgoing ISDN PRI-only trunk, the contents of the display information element in the ISDN PRI SETUP message, which contains the calling party name, are taken from the contents of the DCS name message.
- When an incoming call on an ISDN PRI trunk is routed to an outgoing DCS trunk over an ISDN PRI trunk, the DCS name message is suppressed in G3V2 and later systems. The reason for this is that the DCS name message in this case contains only the trunk group name, if administered, of the incoming ISDN PRI trunk. Since DCS messages normally take precedence, the more informative ISDN PRI display IE would not be used. Thus, the DCS name message is turned off, allowing the ISDN PRI display IE to be used.

DCS Call Coverage does not work in a mixed system of G3V4 and pre-G3V4 systems. The reason for this is that a pre-G3V4 system will not tandem call associated messages that they do not understand. Since the call coverage DCS messages are new starting with G3V4, earlier systems cannot understand them. The following figure illustrates this

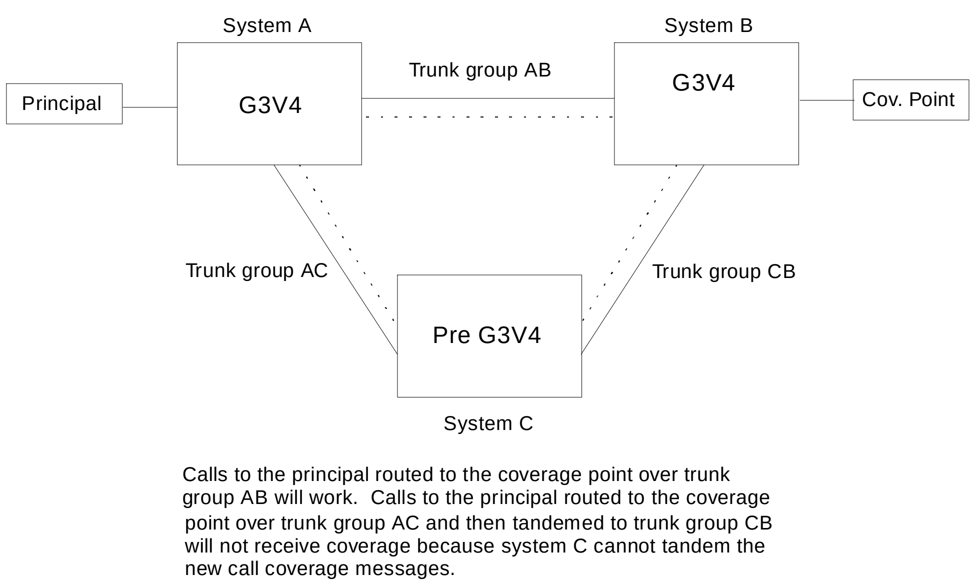


Figure 3-14. Interworking

Hardware and Software Requirements

The following software is required for DCS Call Coverage:

- G3V4 or later system software
- DCS software
- UDP or PNA software
- DCS Call Coverage must be enabled on all systems having principals or coverage points and on all intervening systems.

DCS Call Coverage does not require any additional hardware beyond what is required by DCS or call coverage.

DCS Call Forwarding All Calls

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows all calls to an extension number to be forwarded to a selected extension number within the DCS network or to an external (off-premises) number. If the Call Forwarding and DCS Call Forwarding All Calls features are both active, and if a call is forwarded between extensions on the same node, the Call Forwarding coverage path is used. If the nodes are different, the DCS Call Forwarding All Calls coverage path is used.

This feature is activated or deactivated by dial access code or by a Call Forwarding button. The feature can be activated or deactivated only by voice terminal users within the DCS.

Activation and deactivation of the feature is the same as described for the "Call Forwarding All Calls" feature, described earlier in this chapter.

Considerations

With DCS Call Forwarding All Calls, voice terminal users can have their calls follow them to any location within the DCS network or outside the DCS network.

Calls to an attendant cannot be forwarded. However, an attendant can activate or deactivate the feature for other extension numbers within the DCS.

Interactions

If the forwarding extension and the designated extension are at different nodes, and the designated extension's coverage criteria are met on a forwarded call, the call is redirected to a point in the designated extension's coverage path.

If the forwarding extension and the designated extension are at different nodes, LWC and Coverage Callback cannot be activated at the designated extension for a forwarded call.

There is a 30-second interval during which calls forwarded from the DEFINITY switch to another DCS node is denied. This prevents forwarded incoming trunk calls from being forwarded ad infinitum between two extensions.

Administration

The administration for DCS Call Forwarding All Calls is the same as that for the "Call Forwarding All Calls" feature, which is fully described elsewhere in this chapter.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Call Waiting

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Allows calls from one node to busy single-line voice terminals at another node to wait until the called party is available to accept the call.

DCS Call Waiting includes the following features:

- Attendant Call Waiting
- Call Waiting — Termination
- Priority Calling

Attendant Call Waiting, Call Waiting Termination, and Priority Calling function the same between the DEFINITY system Generic 1, Generic 3, or System 75 nodes in a DCS as they do from a single DEFINITY system Generic 1, Generic 3, or System 75. These features are detailed in Chapter 3, "Feature Descriptions".

Considerations

With DCS Call Waiting, a single-line voice terminal user, by knowing a call is waiting, can quickly process calls from locations within the DCS.

Call Waiting — Origination can only be received if Generic 3i not activated.

DCS priority calling from the attendant station is *not* available.

Interactions

DCS Call Waiting is denied when the following features are activated at the single-line voice terminal:

- Automatic Callback (to or from the voice terminal)
- Data Privacy
- Data Restriction

On incoming trunk calls to the attendant extended over DCS trunks, Attendant Call Waiting interacts with the EDCS feature.

Administration

None required.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Distinctive Ringing

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Activates the alerting, or ringing, device of a called terminal so that the user is aware of the type of incoming call before answering it. Distinctive Alerting functions in a DCS environment as it does within a System:

- For G3sV1 and G3iV1, internal calls are identified by a 1-burst ringing pattern; external calls by a 2-burst ringing pattern; and priority calls by a 3-burst ringing pattern.
- For G3i-Global, G3rV1, G3V2, and later releases, these values are administrable; the defaults are the same as those given above for G3sV1 and G3iV1.

A detailed description of the "Distinctive Ringing" feature is given earlier in this chapter.

Considerations

DCS Distinctive Ringing allows the user to identify the type of incoming calls. By knowing the type of incoming call, the user can answer each call properly.

When DCS transparency is lost for any reason, terminal-to-terminal calls made between nodes produce external ringing instead of the usual internal ringing. Loss of transparency may occur when the data link between nodes is down or when data transmission delay exceeds the trunk signaling time.

Interactions

The following features interact with the DCS Distinctive Ringing feature:

- Distinctive Ringing

  Distinctive Ringing treats a call from another switch in a DCS arrangement as external; DCS Distinctive Ringing treats such calls as internal. If both features are administered, DCS Distinctive Ringing takes precedence. If EDCS is activated, DID treatment may be different. See the "Enhanced DCS (EDCS)" feature.

- Intercom — Automatic

  This feature and its distinctive ringing are not provided between nodes in a DCS.

- Intercom — Dial

  This feature and its distinctive ringing are not provided between nodes in a DCS.

- Manual Signaling

  This feature and its distinctive ringing are not provided between nodes in a DCS.

- Tie Trunk Access

  On a DEFINITY Generic 3 switch, tie trunk groups can be administered as either internal or external tie trunk groups. Calls from internal tie trunk groups are treated as terminal-originated calls and receive one-burst ringing. Calls from external tie trunk groups are treated as externally originated calls and receive two-burst ringing.

Administration

None required.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Leave Word Calling

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Enables terminal users to leave preprogrammed ‘‘call me’’ messages at other terminals within the DCS network. Messages can be left by calling, called, or covering users.

LWC transparency in a DCS configuration allows messages from a DEFINITY switch to another node, depending on the storage capability of the remote node.

Considerations

DCS LWC lets users within the DCS leave short, simple messages for other users.

LWC cannot be successfully activated toward any system that is not capable of storing the messages, either internally or in an associated adjunct.

Messages from one node, through an intermediate node, to a remote node do not require storage capability at the intermediate node.

LWC transparency is supported for all DCS configurations except for cases when either the activating node or the remote node is either an ENHANCED DIMENSION PBX or a System 85 R2V1.

Retrieval of LWC messages is permitted only from a terminal at the node where the messages are stored.

DCS LWC cannot be activated from an attendant console.

Interactions

The following features interact with the DCS LWC feature.

- DCS Multi-appearance Conference/Transfer

  Activation of LWC is denied after a DCS call has been conferenced or transferred.

- DCS Call Forwarding All Calls

  If the forwarding extension and the designated extension are at different nodes, LWC cannot be activated at the designated extension for a forwarded call.

Administration

The administration for DCS LWC is the same as that for LWC ("Leave Word Calling"), which is fully described elsewhere in this chapter.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Multi-Appearance Conference/Transfer

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Provides transparency of transfer of calls, and of conference calls within a DCS network. A user in the DCS can make conference calls or transfer calls originated from any extension in the DCS network to another extension within the DCS. (For transferred calls, the destination need not be within the DCS.)

In a DCS, if a party in a conference hangs up or completes a transfer leaving only outgoing trunks on the call, an attempt is made to preserve the connection if any of the remaining parties on the call is a DCS tie trunk. This can be accomplished if the DCS tie trunk can signal the remote node when the party hangs up. The remote node sends a reply to the originating node, and disconnect supervision is provided for that trunk.

Conference Calls can be placed and calls can be transferred to users within the DCS by dialing the UDP extension number.

A detailed description of the "Conference — Attendant", "Conference — Terminal", and "Transfer" features is given elsewhere in this chapter.

Considerations

DCS Multi-Appearance Conference/Transfer is useful when it is necessary to talk to more than one party at one time within a DCS. Multiappearance voice terminals must have an idle appearance in order to transfer a call.

Interactions

The following features interact with the DCS Multi-Appearance Conference/Transfer feature.

- Voice Terminal Display

  No display transparency is provided for DCS Multi-Appearance Conference/Transfer.

- EDCS

  On calls to or from Public Network Trunks, calling/called party restrictions are checked when EDCS is active.

Administration

None required.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Over ISDN-PRI D-Channel

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Enhances the DCS feature by allowing access to the public network for DCS feature connectivity between DCS switch nodes. With this feature, DCS features are no longer restricted to private facilities. The ISDN-PRI B-channel is used for voice communications, and the ISDN-PRI D-channel is used to transport DCS control information. The only difference between DCS networks that do not utilize the DCS Over ISDN-PRI feature and those that do is in the method of signaling. The DCS Over ISDN-PRI D-Channel feature uses Message-Associated User-to-User Information (MA-UUI) and Temporary Signaling Connections (TSCs) to transport certain DCS control information. MA-UUI allows additional user-specific information to be transported along with certain ISDN call-control messages.

A TSC provides a temporary signaling path through ISDN switches for exchanging user-user information. There are two types of temporary signaling connections: Call Associated Temporary Signaling Connections (CA-TSC) and Non-Call Associated Temporary Signaling Connections (NCA-TSC).

A CA-TSC refers to a service for exchanging USER INFORMATION messages that are associated with an ISDN B-Channel connection by the call reference value of the call control data packets.

An NCA-TSC is a connection not related with any ISDN B-channel connections. It is an administered virtual connection established for exchanging USER INFORMATION messages on the ISDN D-channel for the DCS Over ISDN-PRI D-Channel application as well as for the DCS AUDIX application. Once an NCA-TSC has been administered and enabled, it will be active for an extended period of time. There are two types of administered NCA-TSCs depending on their setup mechanism: *Permanent* and *As-needed*.

Once enabled, a permanent NCA-TSC will remain established while the system is running. If the permanent NCA-TSC is dropped for any reason, the system will attempt to reestablish the connection. An as-needed administered NCA-TSC is established based on user request and the availability of TSC facilities. The connection will be dropped after an administered period of inactivity.

The system can transport DCS or DCS Audix messages over an ISDN-PRI

D-channel and over BX.25 data links when functioning as a gateway between a PBX equipped with the DCS Over ISDN-PRI D-Channel feature and a PBX equipped with traditional DCS using BX.25 data links. In this situation, the messages travel from the gateway through the %NCA-TSCs or %CA-TSCs to %TSC-capable PBXs and from the gateway to PBXs that support only traditional DCS via a BX.25 logical channel.

At least one PBX must be configured as an ISDN DCS Gateway node in a DCS network that consists of PBXs that support DCS Over ISDN-PRI D-Channel and PBXs that do not support the feature.

For examples of various DCS network configurations and how they are administered, see *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

Considerations

System users should notice no difference between DCS features over ISDN-PRI and traditional DCS features.

Interactions

Most feature interactions with DCS Over ISDN-PRI D-Channel are the same as those with traditional DCS features. However, some interactions are unique to the DCS Over ISDN-PRI D-Channel feature. These unique interactions are listed below:

- Attendant DXS with Busy Lamp Field

  An attempt by the attendant to directly select an extension that has been previously administered as belonging to a administered NCA-TSC will result in intercept tone being received.

- D-Channel Backup

  In the event of a D-channel switchover (primary to secondary or secondary back to primary) in a private network, administered NCA-TSCs that were active are assumed to have remained active. Any unacknowledged user-user service requests are assumed to be rejected, and administered NCA-TSCs which were in the process of being established at the time of the switchover are dropped when the switchover occurs. Those administered NCA-TSCs that were dropped will be reattempted again.

  If a D-channel switchover occurs on a D-channel going to the public network then all TSCs will be dropped. A maintenance-provided ‘‘heartbeat’’ message will periodically be sent over each permanent administered NCA-TSC to ensure that such a situation is detected and recovered from.

- Distributed Communications System AUDIX (DCS AUDIX)

The DCS over ISDN-PRI D-Channel feature can be used to support DCS Audix. (The connection between G3i and AUDIX should be BX.25.)

- GRS

  GRS will select TSC compatible facilities when routing NCA-TSCs. In other words, a NCA-TSC request can only select a routing preference that supports TSCs.

  In a tandem node, GRS will first select facilities that support TSCs if the call falls into any one of the following two conditions:

  — It requests a CA-TSC explicitly

  — It contains a DCS information element in the SETUP message

  Once a trunk group with available members is selected, the call will proceed even if all the TSCs belonging to the associated signaling group are active. In other words, the completion of a call is given priority over DCS transparency.

- ISDN-PRI

  This feature uses ISDN-PRI call control protocol and messages.

- SDN

  The DCS over ISDN-PRI D-Channel feature allows the system to access public networks such as SDN. SDN supports all DCS features except for the following:

  — "DCS Attendant Control of Trunk Group Access"

  — "DCS Attendant Direct Trunk Group Selection"

  — "DCS Busy Verification of Terminals and Trunks"

- CDR

  CDR will record both the status and the utilization of TSCs. Both CA-TSCs and NCA-TSCs can be recorded. For more information, consult the CDR description in this manual or the CDR manual.

- Voice Terminals

  An attempt to dial an extension that has been previously administered as belonging to a administered NCA-TSC will result in intercept tone being received.

Administration

The ISDN-PRI option must be enabled on the “System-Parameters Customer-Option” form before associated forms and fields on forms can be administered.

To implement this feature, the following form(s) must be completed.

- Signaling Group Form

- ISDN TSC Gateway Channel Assignment Form — Used to map administered NCA-TSCs to (BX.25) processor channels in an Integrated DCS Network. (Form only required for PBX location serving as an ISDN Gateway in an Integrated DCS Network.) Complete all fields as required.
- ISDN Trunk Group Form
- Routing Pattern Form
- Communications-Links Processor Channels Form — The “Appl” field specifies to which application channel this processor channel is connected. Enter gateway in the associated “Appl” field if the processor channel is used as one end of the gateway channel assigned in the “ISDN TSC Gateway Channel Assignments” form. The gateway node serves as the terminating node to the D-Channel DCS network as well as the terminating node to the traditional DCS network.

  A PBX serving as an ISDN DCS Gateway node introduces some interesting situations when administering processor channels in an associated traditional DCS PBX. In a traditional DCS network, (BX.25 processor channel links) the “Remote Port” field in the “Processor Channel Assignments” form refers to the processor channel of the destination PBX. In an Integrated DCS network, the “Remote Proc Chan” field in the “Processor Channel Assignments” form refers to the processor channel of the Gateway PBX (if the destination PBX is an ISDN DCS PBX), *not* the destination PBX.

  On the contrary, the “Machine-ID” field in the “Processor Channel Assignments” form refers to the destination PBX, either an ISDN DCS PBX or a traditional DCS PBX. The Gateway PBX number *must* not be used in this field if the destination PBX is an ISDN DCS PBX.

NOTE:
There are several differences in between G3i and G3r administration. For example, PRI is translated a little differently in G3r when traditional DCS and this feature are used in combination. On systems with AUDIX in a DCS environment, an additional column has been added to the “Signalling Group” form so the administrator can specify which AUDIX on which switch to use. When traditional DCS and DCS over ISDN are used in combination, translations are also different. Refer to the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for further information.

Hardware and Software Requirements

The DCS over ISDN-PRI D-Channel feature requires the same hardware as other ISDN-PRI features.

ISDN-PRI software is required.

DCS Trunk Group Busy/Warning Indication

Feature Availability

DCS is available with all Generic 3 releases except G3vs/G3s ABP.

Description

Provides attendants with a visual indication that the number of busy trunks in a remote group has reached an administered level. A visual indication is also provided when all trunks in a trunk group are busy.

If an attendant has a Trunk Hundreds Select button assigned to a remote trunk group, the button's Busy lamp lights when all trunks in the trunk group are busy. The lamp goes dark when one of the busy trunks becomes available.

If an attendant has a three-lamp Trunk Hundreds Select button assigned to a remote trunk group, the button's Warn lamp lights when the number of busy trunks in the trunk group reaches the Busy Warning Threshold. The lamp goes dark when the number of busy trunks in the trunk group falls below the Busy Warning Threshold.

To ensure that the busy, warning, and control status of all Trunk Hundreds Select buttons in the DCS remain consistent with the status of the corresponding trunk groups, some nodes in the DCS broadcast the status of a different local trunk group, every 50 seconds, to all directly connected nodes. A pair of DCS nodes are directly-connected if the voice tie trunks between them are not connected through another switch. For example, a node with 30 trunk groups would take 1,500 (50 x 30) seconds to broadcast the status of all 30 trunk groups. This is called a lamp audit. When a node receives a lamp audit message, its TGB/TGW lamps are updated accordingly.

Considerations

Trunk Group Busy and Trunk Group Warning Indication is particularly useful with the Attendant Control of Trunk Group Access feature. The indicators alert the attendant when control of access to local and remote trunk groups is necessary.

Except for remote S75 R1V3, S85/G2 and S75 R1V2 switches, this feature is only transparent if the remote switch is directly connected by voice tie trunks. For S75 R1V3 and S85/G2 remote switches, Trunk Group Busy and Trunk Group Warning Indication is provided regardless of whether the voice tie trunks are directly-connected or in tandem through an intermediate node; For S75 R1V2 remote switches, no Trunk Group Busy and Trunk Group Warning Indication transparency is provided.

Interactions

If Trunk Hundreds Select buttons are assigned for Loudspeaker Paging Access zones, Trunk Group Busy Indicators will provide a visual indication of the busy or idle status of the zones at the remote location as well as at the local node.

Administration

Administration for DCS Trunk Group Busy/Warning Indication can only be administered for remote trunk groups that are directly connected to the local switch. The trunk group access codes for such trunk groups must be three digits or less.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

Default Dialing

Feature Availability

This feature is available with all Generic 3 releases.

Description

Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination by either entering Return at the DIAL: prompt (for data terminals using DCP data modules) or typing d and entering Return at the CMD: prompt (for data terminals using ISDN-BRI data modules). The data terminal user with a DCP data module can still place calls to other destinations by entering the complete address after the DIAL: prompt (normal Data Terminal Dialing or Alphanumeric Dialing). The data terminal user with an ISDN-BRI data module can still place calls to other destinations by typing d, entering a space, typing the complete address, and entering Return after the CMD: prompt.

Default Dialing provides data terminal users who dial a specific number the majority of the time a very simple method of dialing that number. Normal Data Terminal Dialing and Alphanumeric Dialing are unaffected.

For the AT command interface supported by the 7400A data module, the **ATD** command may be entered instead of a carriage return to dial the default destination.

The default destination is administered by the System Manager. Default Dialing can be assigned to any data endpoint that has the Data Terminal Dialing capability.

Considerations

Default Dialing provides data terminal users who dial a specific number the majority of the time a very simple method of dialing that number.

Interactions

The following features interact with the Default Dialing feature.

- Data Call Setup

  Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination by either Return at the DIAL: prompt (for DCP) or typing d and entering Return at the CMD: prompt (for ISDN-BRI).

Administration

Default Dialing is administered by the System Manager. In addition to those items listed in the "Data Call Setup" feature description, elsewhere in this chapter, the following items must be administered:

- The preadministered default destination must be stored in an abbreviated dialing list. This list can be a system list, group list, personal list, or enhanced list. Therefore, before this feature can be activated, users' terminals need to be administered with an abbreviated dialing list that stores the default destination Then, each data module needs to specify the index number of the entry which stores the default destination. The following fields from the "Data Module" form are needed to activate the Default Dialing feature.
  - — Special Dialing Option (default/hotline) — For Default Dialing, ''default'' must be selected. Default Dialing and the Data Hotline feature cannot both be assigned to an extension.
  - — An Abbreviated Dialing List — One entry in the data module's abbreviated dialing list needs to store the default destination.
  - — Dial Code — An index to the entry of the abbreviated dialing list that stores the default destination.

Hardware and Software Requirements

No additional hardware or software is required.

Dial Access to Attendant

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows voice terminal users to access an attendant by dialing an attendant access code. Attendants can then extend the call to a trunk or to another voice terminal.

In G3vsV1/G3sV1 and G3iV1, the access code is always the single digit 0. For G3i-Global, G3rV1, G3V2, and later releases, this code is administrable and may be any one or two digit number; the default is 0.

Considerations

With Dial Access to Attendant, voice terminal users can access the attendant whenever attendant aid is needed by dialing the attendant access code.

A voice terminal user calling the attendant by dial access cannot be added to an existing conference by the attendant.

Interactions

Restriction — Origination (administered to a voice terminal by the COR) prohibits placing any calls, including Dial Access to Attendant calls.

For G3v1/G3sV1 and G3iV1 only, the attendant group dial code must be zero and cannot be administered to another value. (This dial code can be administered for G3i-global, G3rV1, G3V2, and later releases.)

Administration

The administrator may select an extension number other than zero for this feature. However, in the United States 0 has traditionally been used for operator assistance in both the public network and in PBXs, Also, consideration should be given to the fact that prior to the emergence of 911 systems, the dialing of 0 was customary in the United States for emergencies and other urgent assistance requests. Using an extension number other than 0 for dial access to the attendant could create confusion and delay response to an emergency.

Hardware and Software Requirements

No additional hardware or software is required.

Dial Plan

Feature Availability

This feature is available with all Generic 3 releases.

Description

The Dial Plan is the system's guide to digit translation. When a digit is dialed, the system must know what to expect, based on that digit. For example, if a voice terminal user dials a 4, the system must know how many more digits to expect before the call is processed.

There are a number of enhancements and changes for G3i-Global, G3rV1, G3V2, and later releases. There are two digit tables giving more flexibility in the dial plan. The number of extensions is greatly increased and, for G3rV1, G3V2, and later releases, there is the introduction of Extension Number Portability. The number of trunk access codes increased significantly as well as other capacities. See the System Capacity Limits table in for more details.

The Dial Plan, or first-digit (and in G3rV1, G3i-Global, G3V2, and later releases, the second-digit) table, established during administration for each system, provides information to the switch on what to do with dialed digits. The tables define the intended use of a code beginning with a specific first digit (and or first two digits for G3i-Global, G3rV1, G3i-Global, G3V2, and later releases) and relates to the system how many digits to collect before processing the code. The choices of a first digit are 0 through 9, *, and #. Permissible code uses and the allowable number of digits are listed below.

- Extension Numbers

  Flexible numbering allows one, two, three, four, or five-digit extension numbers. The first digit in the extension number tells the system how many digits to expect the extension number to have.

  Extension numbers can have a first digit of 1 (0 for G3rV1, G3i-Global, G3V2, and later releases) through 9. For example, if a three-digit extension number is administered and the first digit is a 4, the extension numbers can range from 400 to 499. Also, if a four-digit number with a 6 as the first digit is administered, the extension numbers can range from 6000 to 6999. If multiple extension number lengths begin with the same digit, G3i-Global uses a timer to identify when all the digits have been entered.

- Attendant

  Dial access to the attendant group may be achieved by dialing the single digit ‘‘0.’’ It is also possible to set the dial plan so that some other digit is used to reach the attendant (for example, 9 in Italy). For G3rV1, G3i-Global, G3V2, and later releases, access can range from 0 to 9 with a length of one or more digits. In addition, Generic 3 provides for Individual Attendant Access by assigning each attendant an individual extension number.

- Trunk Access Codes

  A minimum of one digit and a maximum of three (G3iV1) digits or four (G3i-Global and G3rV1, G3V2, and later releases) can be used. Trunk access codes can have a first digit of 1 (0 for G3rV1, G3i-Global, and G3V2) through 9. For example, 9 could be used for local trunks, 8 for WATS trunks, and 7 for tie trunks. For G3r and G3i-Global, * and # can be the first digit for TACs. TACs may be a maximum of 4 digit. TACs for DCS trunk groups and trunk groups controlled by DCS can be a maximum of 3 digits (or 4 if no DCS).

- Feature Access Codes

  A minimum of one digit and a maximum of three digits can be used. The * and # buttons can be used as part of a feature access code and, when used, must be the first digit. The * or # counts as one digit. For example, * 2 could be used to activate Call Forwarding All Calls and # 2 used to deactivate Call Forwarding All Calls.

  Feature access codes can also have a first digit of 1 through 9. For example, 3 2 could be used to activate Call Forwarding All Calls and 3 3 used to deactivate Call Forwarding All Calls.

- Prefixed Extensions

  The dial plan does not route on the first digit prefix (0 through 9) but routes on the rest of the digits.

- Miscellaneous Code

  Used to indicate that any dialed digit string starting with the first digit uses the second digit table for interpretation. For G3i-Global, G3V2, and later releases only, the length of this code can only be one. It can have a feature access code, trunk access code, attendant access code, and extensions on the second digit table.

For G3rV1, G3V2, and later releases, UDP has its own forms in administration. This information is given in the UDP and ENP Tables.

A UDP may also be established during administration as part of the Dial Plan. This plan provides a common extension number plan that can be shared among a group of switches. If a UDP is to be established, all extension numbers (in the UDP numbering plan) must be the same length (four or five digits but not both). So that calls route to the desired switch, a UDP requires the following information:

- A PBX Code, which represents the first 1 to 4 digits (five digits for G3rV1, G3V2, and later releases) of a 4- or 5-digit extension number and can range from 1 to 9999 with a maximum of 240 PBX Codes.
- Whether or not the PBX Code is local to this system — this information is required for each PBX Code for G3vsV1/G3sV1, G3iV1, and G3i-Global.

- An RNX, which is associated with the PBX code and is used to select an AAR pattern for the call — this information is required for each PBX code. With G3i, the RNX is flexible (no fixed form). (With G3rV1, G3V2,and later releases, an ENP can also be used to route UDP calls.)
- A PBX ID (1 to 63), which represents a specific switch — this information is required for G3V1 (except G3rV1) for each PBX Code when the switch is located within a DCS. For G3rV1, G3V2, and later releases, the PBX ID is administered in the Node Number column of the ''AAR Analysis'' form.'
- A Local PBX ID, which represents the PBX ID of the local switch.

Considerations

The entire Dial Plan is dependent on the first and second digit dialed. The 12 possible choices of a first digit are 0 through 9, *, and #.

Interactions

- All dial access features and services provided by the system require the Dial Plan.
- When an ISDN/BRI station dials sufficient digits to route a call, but could route differently if additional digits were dialed, the station will not recognize the Conference or Transfer buttons. The user must delay dialing for three seconds or dial a # to indicate the call can be routed based on the digits already dialed. The Conference or Transfer buttons are then recognized and the operation is completed by the switch.
- Flexible numbering is supported in countries using R2-MFC trunk signaling without Group II tones. Different length extension numbers can exist as long as the extensions have different first digits. Contact your AT&T representative for more information.

Administration

The Dial Plan is administered on a per-system basis by the System Manager. The following items require administration:

- Area code where the system is located (in North America only)
- Whether or not the serving central office requires the digit 1 to indicate a long-distance call (in North America only)
- Whether or not a UDP is to be established
- The type of code and the number of digits in the code for each first digit
- Optionally, a second digit table

If a UDP is to be established on G3i (G3r contains additional features as described in Uniform Dial Plan feature and is administered on a different form), the following items can be administered:

- Number of digits in plan (4 or 5)
- PBX Codes
- Whether or not each PBX Code is local to the PBX being administered
- RNX (Per PBX Code)
- PBX ID (Per PBX Code)

Hardware and Software Requirements

No additional hardware or software is required.

Digital Multiplexed Interface

Feature Availability

This feature is available with all Generic 3 releases.

Description

Supports two signaling techniques: Bit Oriented Signaling and Message Oriented Signaling for direct connection to host computers.

Message Oriented Signaling is used with ISDN-PRI.

The DMI provides twenty-three 64 kbps data channels, plus one 64 kbps channel for Common Channel Signaling. Within the data channel, DMI provides control information exchange and data formats supporting data transport at all standard data rates; each data channel can be used in one of the following transfer modes:

- Mode 0 — 64 kbps Channel
- Mode 1 — 56 kbps Channel
- Mode 2 — 0-19.2 kbps Synchronous/Asynchronous
- Mode 3 — Multiple Virtual Channels

ISDN-PRI can be assigned as the signaling mode. In this case, for G3i only, the TN767 DS1 or TN464C/D DS1 circuit pack and the TN765 Processor Interface circuit pack must be used. For G3r, the TN464B/C/D circuit pack is required. The ISDN-PRI is a 1.544 Mbps digital interface that consists of a 1.536 Mbps signal multiplexed with an 8 kbps framing channel. The 1.536 Mbps signal is divided into 24 channels of 64 kbps each (23 ‘‘B’’ voice or data channels and 1 ‘‘D’’ signaling channel). The D channel multiplexes signaling messages for the 23 B-channels.

NOTE:
The TN767 and the TN464C and later revisions of the DS1 circuit packs provide a DSX1 interface.

DMI trunks are accessed the same as tie trunks. The only difference is that DMI trunks are connected to host computers while tie trunks are connected to another switch. Each trunk functions like a PDM since the DMI protocol is identical to the DCP format used by the data modules.

Considerations

System DMI support offers high-volume (high-speed, high-capacity) data transmission, via DS1 digital facilities, between host computers and analog or digital data endpoints.

DMI is widely supported. To date, more than 100 data processing suppliers, communications equipment suppliers, and device manufacturers have licensed DMI specifications and have obtained the rights to implement DMI in their products.

DMI trunks with ISDN-PRI signaling can be connected to a host computer, another PBX, or a public or private network.

Interactions

The following features interact with the Digital Multiplexed Interface feature.

- Data Restriction

  DMI trunks cannot be data restricted.

- Modem Pooling

  Data calls dialed from a local analog data endpoint to a DMI trunk must contain the Data Origination Access Code to obtain a conversion resource. Data calls on DMI trunks to local analog data endpoints will automatically obtain conversion resources, if available.

Administration

DMI support is assigned on a per-system basis by the System Manager. The following items require administration.

- DS1 Circuit Pack — Assign the circuit pack to the system before administration of the associated trunks.
- Processor Interface Circuit or Packet Control pack (for G3i) — If ISDN-PRI Signaling is used, a TN765 Processor Interface circuit pack or TN778 Packet Control must be assigned to work in tandem with the TN767 DS1 circuit pack or TN464C/D.

NOTE:
The TN767 and the TN464C and later revisions of the DS1 circuit packs provide a DSX1 interface.

- DMI Trunk Group — Associate the trunks to the groups.

Hardware and Software Requirements

One TN722B, TN767, or TN464 is required for every 23 DMI trunks. The TN464 is needed for G3r. If ISDN-PRI signaling is used, a TN765 Processor Interface or TN778 Packet Control is required for G3i.

No additional software is required.

Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows direct inward access to an answering group other than the attendant even if the system does not have the DID feature.

This feature was called ‘‘Enhanced Uniform Call Distribution’’ in G2.

A DDC or UCD answering group can consist of voice terminals and individual attendants (described in the Individual Attendant Access feature elsewhere in this document). In addition, a UCD group can consist of data modules, data line circuit ports, or modems.

One extension number is assigned to all voice terminals, individual attendants, data modules, data line circuit ports, or modems in a group or department, that is, to a set that serves the same function and requires call distribution among the members of the group. Incoming calls to a DDC group or UCD group can be internal or external.

The hunting algorithm used by the system to select an idle terminal or console is the only difference between DDC and UCD.

With DDC, an incoming call rings the first available voice terminal or individual attendant in the administered sequence. If the first group member in the sequence is active on a call (busy), or has had his or her calls temporarily redirected (via Send All Calls, Call Forwarding All Calls, or the Hunt Group Busy Function discussed later), the call routes to the next group member, and so on. In other words, incoming calls always try to complete at the first group member in the administered sequence. Therefore, the calls are not evenly distributed among the DDC group members.

With UCD, an incoming call rings the member of the group that has not received a UCD group call for the longest period of time (the most idle member). In other words, incoming calls to a UCD group extension number are distributed evenly among the group members.

When DDC or UCD is not provided, incoming LDN calls, international exchange calls, 800 service calls, and automatic tie trunk calls are normally directed to an attendant who must extend the call. When DDC or UCD is provided on a trunk group, incoming calls are automatically directed to the desired DDC group by the switch. Attendant intervention is not required.

Any voice terminal or individual attendant can be a member of one or more DDC and/or UCD groups. Data modules, data line circuit ports, and modems are limited to UCD groups and can be a member of one or more groups. Each member of a group also has its own unique extension number and can be called individually. Multi-appearance voice terminals and attendant consoles can have an assigned status lamp that identifies an incoming DDC or UCD call. However, the voice terminal or individual attendant must be idle (not active on any call appearance) before a group call is directed to the terminal or console. Therefore, a voice terminal can receive only one DDC or UCD call at a time.

A queue can be established for a DDC or UCD group. When all members of the group are active, the queue allows incoming calls to await an idle terminal.

When a call enters the queue, a delay announcement interval is started. This interval (0 to 99 seconds) indicates how long a call remains in queue before the call is connected to a recorded announcement. If Call Coverage is provided, the Don’t Answer Interval (one to 99 ringing cycles) may also begin when the call enters the DDC or UCD group queue. After these intervals have begun, one of the following occurs:

- If the Coverage Don’t Answer Interval expires before the delay announcement interval expires, the call is redirected to coverage. If no coverage point is available to handle the call, the call remains in queue and may then be connected to delay announcement.
- If the delay announcement interval expires before the Coverage Don’t Answer Interval, the call is connected to a delay recorded announcement, if available. Once a call is connected to a delay announcement, it remains in queue until a group member becomes available. If the announcement is already in use, the delay announcement interval is reset. This process (as described above) continues until the call is answered, goes to coverage, is connected to a delay announcement, or the calling party hangs up.

If the delay announcement interval is administered as 0 seconds, the incoming call automatically is connected to the announcement, if available. The result is a ‘‘forced first announcement,’’ and the call is not attempted to access a hunt group member until after the announcement is heard.

Calls connected to a delay recorded announcement remain in queue while the announcement is heard by the caller. If the call has not been answered by the time the announcement is over, the call is connected to music (if provided) or there is silence, as long as the call remains in queue. When the call begins ringing a member of the hunt group, the calling party hears audible ringing. Music is not provided after a forced first announcement.

The queue length can be set if queuing is provided. If queuing is not provided, the queue length must be set to zero. If queuing is not provided, if the queue is full, or if all group members (voice terminals or individual attendants only) have activated the Hunt Group Busy option (discussed later), calls to a busy group receive busy tone (unless using a Central Office trunk) or redirect via the Call

Coverage feature. Lamp indicators may be used to give a warning when the number of calls waiting in the queue reaches a predetermined limit (queue warning limit). The queue warning level cannot exceed the queue length.

When the queue warning level is reached, the indicator lamp lights and remains lighted until the calls waiting in queue are fewer than the queue warning level. A queue warning level lamp may be provided for each DDC or UCD group queue. The lamp can be installed at any location convenient for the group.

As an example of queue warning level and delay announcement operation, assume that there is an incoming call to a DDC or UCD group with the following parameters.

- Queue length is 10 calls.
- Queue warning level is five calls.
- Recorded announcement delay is 20 seconds.

Also assume the following:

- All DDC or UCD group members are busy.
- The call is the fifth call in the queue.

Since all members in the DDC or UCD group are busy, the incoming call enters the queue. The incoming call, being the fifth call in the queue, causes the queue warning level to be reached. This causes the queue warning level lamp to light.

From the indicator lamp, the DDC or UCD group members know the queue warning level has been reached and try to complete their present calls. Meanwhile, the incoming call has been in the queue for 20 seconds and hears the delay recorded announcement. The caller may decide to hang up or may decide to remain in the queue. Assume the caller remains in the queue. When a DDC or UCD group member becomes idle, the longest queued call is directed to that group member. The queue warning level lamp may or may not be lighted at that time, depending on the number of other calls that have been queued. Also, the first four calls in the queue have heard the delay announcement after being queued for 20 seconds. The queue warning level and delay announcement capabilities are independent of each other.

A Coverage ICI button can be assigned to a hunt group member's multiappearance voice terminal. The Coverage ICI button allows the user who is a member of more than one hunt group to identify a call that is directed to a specific hunt group. When a hunt group member receives a call that is directed to the hunt group assigned to that button, the button's status lamp lights.

Considerations

DDC and UCD are particularly useful when the answering group assigned receives a high volume of incoming calls. Call completion time is minimized and attendant assistance is not required. This feature can also minimize the use of DID trunks.

If DDC and UCD groups are both used in the system, the number of combined groups and the number of voice terminals per group are determined by the size of the system and call traffic requirements.

Each system recorded announcements. Each group queue can be assigned one of these announcements as a delay announcement. A delay announcement can be shared among the DDC groups, UCD groups, or a combination of these groups. Delay announcements may be either analog or digital (integrated). Only one caller can be connected to an analog announcement at any one time. Multiple callers can be connected to the same integrated announcement. Callers are always connected at the beginning of the announcement. More efficient use of the announcements is realized if the announcements are brief. The capability to install multiple Integrated Announcement boards is available with G3V4 and later releases. See the "Recorded Announcement" feature.

Calls incoming on a non-DID trunk group can route to a DDC group instead of to an attendant. Calls incoming on any non-DID trunk group can have only one primary destination; therefore, the trunk group must be dedicated to the DDC group.

If a delay announcement is used, answer supervision is sent to the distant office when the caller is connected to the announcement. Charging for the call, if applicable, begins when answer supervision is returned.

Multi-appearance voice terminals can receive only one DDC or UCD call at a time. A voice terminal is idle for a DDC or UCD call only if all call appearances are idle.

A Hunt Group Busy option can be administered for the system. When a voice terminal user or individual attendant dials the Hunt Group Busy activation code followed by the DDC or UCD group number, or presses the Auxiliary Work button, the terminal or console appears busy to the DDC or UCD group. This effectively removes the member from the group until the user dials the Hunt Group Busy cancellation code or presses the button again. The Auxiliary Work button can be assigned to multiappearance voice terminals only. Calls to a busy hunt group receive a busy signal if the caller is internal or incoming on a DID, tie, or DS1 tie trunk. A Caller to a busy hunt group hears ringing if the caller is incoming on a CO trunk.

The last available member of a DDC or UCD group cannot activate the Hunt Group Busy option if any calls are remaining in the queue. An attempt by the last available group member to activate the Hunt Group Busy option results in the following:

- New calls to the DDC or UCD group either receive busy tone or redirect to coverage.
- Calls already in the queue continue to route to the last available voice terminal until the queue is empty.
- At the last available voice terminal or console, the status lamp associated with the Auxiliary Work button, if provided, flashes until the queue is empty. When no more calls remain in the queue, Hunt Group Busy is activated and the status lamp, if provided, lights steadily. (The same sequence applies when Hunt Group Busy is dial activated instead of button activated, except there is no status lamp.)

LWC messages can be stored for a DDC or UCD group and can be retrieved by a member of the DDC or UCD group, a covering user of the group, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization must be assigned to the message retriever. Also, a remote Automatic Message Waiting lamp can be assigned to a group member to provide a visual indication that a message has been stored for the group. One remote Automatic Message Waiting lamp is allowed per DDC or UCD group. The status lamp associated with this button informs the user that at least one message has been left for the group.

Members of a UCD group used for data communications must be of the same type and serve the same function. Either data modules or analog modems can be used in a UCD group, not a mixture of the two, and the group must be dedicated to a specific, intended use.

Since any member of a data UCD group can be used on a given call, option settings must be the same for all group members. This minimizes call setup failures because of incompatible options between the origination data module or modem and the UCD group data module or modem selected for the call.

A Data Extension button can be used to access the associated data module, even if the module is in a UCD group. Individual data modules or modems can originate and receive calls.

Each UCD group and each individual UCD member is assigned a COR. Miscellaneous Restrictions, described in this chapter, can be used to prohibit selected users from accessing certain UCD groups. Either Miscellaneous Restrictions or restrictions assigned through the COR can be used to prohibit the group members from being accessed individually. Unless such restrictions are administered, each group member can be accessed individually as well as through the group.

When a hunt group is changed from ACD to non-ACD, the agent has to enter the Hunt Group Busy deactivation code in order to receive calls in that hunt group. If an Aux-Work button has been administered for that station, then the lamp associated with that button lights, and the button can be pressed to make the agent available for hunt group calls.

Agents should not be used for hunt group calls and ACD split calls simultaneously.

Oldest call waiting termination is only supported on ACD calls, not on hunt group calls.

Interactions

The following features interact with the DDC and UCD feature.

- Attendant Call Waiting

  An attendant can originate or extend a call to a DDC or UCD group. Attendant Call Waiting cannot be used on such calls. However, such calls can enter the group queue, if provided. Attendant Call Waiting can be used on call to the individual hunt group members.

- Automatic Callback

  Automatic Callback calls cannot be activated toward a DDC or UCD group.

- Call Coverage

  Calls can redirect to or from a DDC or UCD group.

  If a user has an Auxiliary Work button, and activates or deactivates Send All Calls, the Hunt Group Busy function associated with DDC or UCD is activated or deactivated simultaneously.

  If a user has no Auxiliary Work button, activating or deactivating Send All Calls still makes the user available or unavailable for DDC or UCD calls, but Hunt Group Busy is not activated or deactivated. The Hunt Group Busy activate or deactivate code and the DDC or UCD extension must be dialed to activate the Hunt Group Busy function.

  Activating or deactivating the Hunt Group Busy function does not activate or deactivate Send All Calls.

  For a call to a DDC or UCD group to be directed to Call Coverage, each voice terminal in the group must be active on at least one call appearance and the queue, if there is one, must be full. If the queue is not full, a call enters the queue when no voice terminal is available. Queued calls remain in queue for a time interval equal to the Coverage Don't Answer Interval before redirecting to coverage. If any voice terminal in the group is idle, the call directs to that voice terminal.

  When a call is redirected via Call Coverage to a hunt group, the calling party does not hear a forced first announcement, if administered. In order for the redirected call to receive an announcement, the announcements must be administered as first or second delay announcements.

- Call Forwarding All Calls

  When activated for a hunt group member, the activating voice terminal appears busy to the DDC or UCD group.

  When activated for the hunt group extension, calls directed to the hunt group are forwarded away from the hunt group. No announcements (other than a forced first announcement, if administered) associated with that hunt group are connected to the call.

- Data Call Setup (to or from a member of a UCD group)

  Voice Terminal Dialing of Data Terminal (Keyboard) Dialing can be used on calls to a UCD group.

- DID

  If DID is provided and the DDC or UCD group extension number is within the range of extension numbers that can be dialed directly, then the group can be called the same as any voice terminal.

- DCS

  If a call to a hunt group is forwarded to a hunt group at another DCS node, the caller does not hear the forced first announcement of the forwarded-to hunt group.

  If a hunt group is in night service, with a hunt group at another DCS node as the night service destination, a call to the first hunt group is connected to the first forced announcement of the hunt group serving as the night service destination.

- Individual Attendant Access

  Individual Attendant Extensions can be assigned to DDC and UCD groups. Unlike voice terminal users, individual attendants can answer DDC and UCD calls as long as there is an idle call appearance and no other DDC or UCD call is on the console.

- Internal Automatic Answer (IAA)

  Internal calls to a DDC or UCD group member are eligible for IAA.

- Multi-Appearance Preselection and Preference

  All assigned call appearances must be idle before a DDC or UCD group call is directed to a voice terminal.

- Music-on-Hold Access

  A call placed in a DDC or UCD group queue can receive a delay announcement followed by music.

- Night Service — Hunt Group

  When Hunt Group Night Service is activated for a hunt group and the night-service destination is a hunt group, the caller hears the first forced announcement, if administered. The call is then redirected to the night service destination hunt group. When a member of the night service hunt group becomes available, the call goes to that member.

- Priority Calling

  A priority call directed to a DDC or UCD group is treated the same as a nonpriority call, except that the distinctive three-burst ringing is heard.

- CDR

  The system can be administered to record the called number on incoming calls as the particular hunt group extension number or hunt group member extension number.

- Terminating Extension Group

  A Terminating Extension Group cannot be a member of a DDC or UCD group.

- Voice Terminal Display

  On calls dialed directly to a DDC or UCD group extension number, the DDC or UCD group's identity is displayed at the calling extension.

Administration

DDC and UCD are administered by the System Manager. The following items can be administered for each DDC or UCD group:

- Delay announcement (not applicable for vector-controlled hunt groups)
- Delay announcement interval (not applicable for vector-controlled hunt groups)
- Group extension number, name, and type (DDC and UCD)
- Coverage Path (not applicable for vector-controlled hunt groups)
- COR
- Four-digit security code
- Whether or not the group is served by a queue
- Queue length (one to 200 calls for G3i, 999 for G3r)
- Queue Warning Threshold (one to 200 calls for G3i, 999 for G3r)
- Port Number assigned to queue warning level lamp
- Group Members (extension numbers)

Also, the system can be administered to record (via CDR) incoming calls to a particular hunt group or hunt group member.

Hardware and Software Requirements

Each queue warning level lamp requires one port on a TN742, TN746B (A-law), or TN769 Analog Line circuit pack. A 21C-49 indicator lamp may be used as a queue warning level lamp. This lamp is approximately two inches in diameter and has a clear beehive lens. The lamp operates on ringing voltage and can be mounted at a location convenient to the group.

Each delay announcement requires integrated or external announcement equipment and one port on a TN742 or TN746B (A-law) Analog Line circuit pack. If music is to be heard after the delay announcement, a music source and a port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) is required, or an auxiliary port (for external announcement equipment). Announcement equipment and music sources are not provided by the system.

No additional software is required. However, if a hunt group is to be vector controlled, Call Vectoring software is required.

Direct Inward and Outward Dialing (DIOD) — International

Feature Availability

This feature is available with G3i-Global, G3V2, and later releases.

Description

Provides a 2-way service, with both inward and outward dialing features, allowing calls from an international public exchange to be made directly to the PBX. The Dialing Inward and Outward Dialing (DIOD) feature is a combination of the Direct Inward Dialing (DID) feature and the Direct Outward Dialing (DOD) feature via a common analog or digital trunk. The Japanese version of DID, however, implies a 2-way service, with both inward and outward dialing features.

An analog DIOD trunk has the capability of being used in any of the following ways:

- DID trunk
- CO trunk — 2-way, 1-way incoming, or 1-way outgoing CO
- DIOD trunk with incoming DID operation and outgoing CO trunk capability

A digital DIOD trunk has the ability to be used with incoming DID operation and outgoing CO operation.

NOTE:
Two-way CO allows one-way outgoing CO calls plus one-way incoming calls via a listed directory number (LDN). The one-way incoming is *not* DID. Two-way DIOD allows two-way CO plus DID.

Considerations

Analog DIOD requires either the TN429 (Japan) or TN2184 (Germany) circuit boards. The TN429 is an 8-port loop-start circuit pack. The TN2184 is a 4-port loop-start circuit pack that sends addressing information for incoming and outgoing calls and detects Periodic Pulse Metering (PPM) signals for metering outgoing calls.

Digital DIOD must be from a TN464D V2 (or later version) circuit board and must have a bit rate of 2.048 Mbps.

NOTE:
Individual countries must specify boards and configurations. Refer to each country's ''Application Notes'' for the appropriate hardware and software requirements.

Interactions

When the DIOD trunk is being used as a DID trunk, the following interactions apply:

- The Inward Restriction, Manual Terminating Line Restriction, and Termination Restriction features (administered by the COR) prevent receiving DID calls at the restricted voice terminal.
- When a DID trunk is accessed via a LDN, the call is routed to the attendant. The attendant display indicates that the call is an LDN call. If Night Service is activated, DID LDN calls route to a designated DID LDN night extension.
- If an incoming DID call is forwarded to another extension and answered by the forwarded-to extension, any other calls to the same DID extension within the next 30 seconds receive busy tone.

When the DIOD trunk is being used as a DOD trunk, the calling party restrictions (assigned by the COR) prevent placing DOD calls from the restricted voice terminal.

Administration

The System Manager must administer a “DIOD Trunk Group” form.

Hardware and Software Requirements

No additional software is required. Requires one of the following circuit packs:

- TN429 analog (Japan)
- TN2184 analog with PPM detection (Germany)
- TN464 for digital trunks with PPM. Refer to the Application Notes in Appendix D of the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-6555, manual for the minimum suffix and/or vintage needs for each country.

NOTE:
The TN767 and the TN464C and later revisions of the DS1 circuit packs provide a DSX1 interface.

Distinctive Ringing

Feature Availability

Distinctive Ringing is available with all Generic 3 releases. This feature is called “Ringing Distinctive’’ in G2.

Description

Helps voice terminal users and attendants distinguish between various types of incoming calls.

The ringing cycle, which begins when a voice terminal or attendant console receives an incoming call, is heard by the voice terminal user or attendant. There are three basic call types - external, internal, and priority. There are three types of distinctive ringing cycles (1-, 2-, and 3-burst); and with G3iV1, and G3vsV1/G3sV1 the association between basic call types and ring cycle types is FIXED (1-burst = internal, 2-burst = external, 3-burst = priority). With G3i-global, G3rV1, G3V2, and later releases, this association between basic call types and ring cycle types is administrable with the default being the same as G1. In addition to these basic call ringing patterns, there are ringing signals for (Automatic and Dial) Intercom calls, Manual Signaling, and Redirection Notification. There are administrable distinctive alerting patterns that can be administered for internal, external, and priority calls. These are system-wide patterns which are administered on the “System Parameters Features” form.

The associated call types, types of users, and ringing cycles are as follows:

NOTE:
Table 3-53 shows the default values used in the United States. For further details, see the "World Class Tone Generation" section later in this chapter

Table 3-53. Distinctive Ringing Defaults in United States

If the Default Ringing Cycle Is:	Then the Call Type Is:	User
One-burst ringing (1.2 on, 4.0 off repetitive)	Internal voice terminal, internal tie trunk, and Remote Access	All voice terminals
	Intercom	Single-line voice terminals
Two-burst ringing (0.2 on, 0.4 off; 0.6 on, 4.0 off repetitive)	Attendant-extended, attendant-originated, and incoming trunk, including external tie trunk	All voice terminals
Three-burst ringing (0.2 on, 0.1 off; 0.2 on, 0.1 off; 0.6 on, 4.0 off repetitive)	Automatic Callback, Priority Calling, and Ringback Queuing callback	Single-line voice terminals
Three-burst ringing (0.1 on, 0.1 off; 0.1 on, 0.3 off; 0.6 on, 4.0 off repetitive)		Multi-appearance voice terminals
Single tone (0.6 on, 4.6 off repetitive)	Intercom	Multi-appearance voice terminals
Single tone (0.2 on)	Manual Signaling	Multi-appearance voice terminals
Single tone (0.2 on)	Redirection Notification	All voice terminals

Table 3-54 lists the call types and their ringing cycles received at attendant consoles.

Table 3-54. Distinctive Ringing Defaults in United States Received at Attendant Consoles

If the Default Ringing Cycle Is:	Then the Call Type Is:
Low-pitched tone (0.4 on, 1.2 off repetitive)	Incoming call
High-pitched tone (0.4 on, 1.2 off repetitive)	Attendant Recall call and when any call associated with a timed reminder interval returns to the console
Low-pitched tone (0.25 on, 0.8 off repetitive)	Calls waiting in queue

Table 3-55 lists the call types and their ringing cycles received at attendant consoles.

Table 3-55. Call Types and Ringing Cycles Received at Attendant Consoles

Call Type	Default Ringing Cycle (In Seconds)
Incoming call	Low-pitched tone (0.4 on, 1.2 off repetitive)
Attendant Recall call and when any call associated with a timed reminder interval returns to the console	High-pitched tone (0.4 on, 1.2 off repetitive)
Calls waiting in queue	Low-pitched tone (0.25 on, 0.8 off repetitive)

Considerations

Distinctive Ringing allows the user to identify the type of incoming call. By knowing the type of incoming call, the user is able to answer each call in a suitable manner for that type of call.

The two- and three-burst ringing is optional only on single-line voice terminals. If Distinctive Ringing is disabled, the user hears a one-burst repetitive tone for all incoming calls. This is useful for equipment interfaced by analog lines, especially if the Off-Premises Station feature is used.

A single distinctive ring cycle is used for each new incoming call when a voice terminal is off-hook or a headset is being used. The CALLMASTER. terminal is alerted with a single ring cycle whenever either the headset or handset is plugged into the headset jack.

Interactions

The Distinctive Ringing cycles are altered when the Personalized Ringing feature is used.

Administration

Ringing is a standard system feature. No administration is required except for the following items which are set by the System Manager:

- For G3i-Global, G3rV1, G3V2and later releases, the various ringing cycles and what they are for.

  NOTE:
  Typically, the one-, two-, and three-burst distinctive ringing patterns are administered for internal, external, and priority calls, respectively.

- Redirection Notification can be assigned for any voice terminal.
- Distinctive two- or three-burst ringing can be disabled for single-line voice terminals.

DS1 Trunk Service

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides a digital interface for the following:

- Voice-Grade DS1 Tie Trunks
- AVD DS1 Tie Trunks
- Robbed-Bit Alternate Voice/Data (RBAVD) DS1 Tie Trunks
- DMI Tie Trunks
- ISDN-PRI Trunks
- CO Trunks
- FX Trunks
- Remote Access Trunks
- WATS Trunks
- DID Trunks
- Off-Premises Stations
- Access Endpoints (G3i)

Voice Grade DS1 Tie Trunks

The Voice-Grade DS1 tie trunks are an alternative to four-wire analog E&M tie trunks and may be used to interface with other properly equipped switching systems.

Voice-Grade DS1 tie trunks can also be used as the following:

- ETN or TTTN tie trunks
- Main/Satellite tie trunks
- Tie trunks used to interface with EPSCS and CCSA networks
- Release link trunks for CAS
- Access Trunks.

The TN722B, TN767, or TN464C/D DS1 circuit pack is used to support Voice-Grade DS1 tie trunks in the Robbed Bit Signaling mode. The Robbed Bit Signaling mode supports 24 trunks for transmission on the circuit pack because the least significant bit (robbed) in every sixth frame of data transmission is replaced by a signaling bit.

This type of tie trunk uses DS1 transmission facilities in the USA and Canada DS1 format, which is a 1.544 Mbps digital signal that consists of a 1.536 Mbps signal multiplexed with an 8 kbps framing signal.

NOTE:
The TN767 and the TN464C and later revisions of the DS1 circuit packs provide a DSX1 interface.

AVD Tie Trunks

AVD DS1 tie trunks permit alternate voice and data calling between a Generic 3 and another switch. The other switch can be a System 75 R1V2 or later,

System 85 R2V1 or later, a DEFINITY system Generic 1, Generic 2, or Generic 3.

AVD DS1 tie trunks can be used to connect the system with other digital switches.

The TN722B, TN767, and TN464 DS1 circuit pack (TN464 supports A-law) is used to support AVD DS1 tie trunks in the Common Channel Signaling mode. The Common Channel Signaling mode supports 23 trunks for data transmission and 1 trunk for signaling purposes. (32 channel signalling mode is supported with the TN464B/C/D boards.)

This type of tie trunk uses DS1 transmission facilities in the USA and Canada DS1 format, which is a 1.544 Mbps digital signal that consists of a 1.536 Mbps signal multiplexed with an 8 kbps framing signal.

Tie Trunks

Robbed-Bit Alternate Voice/Data DS1 tie trunks permit alternate voice and data calling between Generic 3 and another switch. The other switch can be a DEFINITY system Generic 1, Generic 2, or Generic 3.

Robbed-Bit Alternate Voice/Data DS1 tie trunks can be used to connect the system with other digital switches.

Robbed-Bit Alternate Voice/Data DS1 tie trunks can be used to dynamically access voice/data SDN Services.

For normal AVD facilities, modem pool resources are not automatically inserted. With Robbed-Bit Alternate Voice/Data facilities, a data origination access code can be used to force modem pool insertion on the call.

The TN722B, TN767, or TN464 DS1 circuit pack is used to support Robbed-Bit Alternate Voice/Data DS1 tie trunks in the Robbed-Bit Signaling mode.

This type of tie trunk uses DS1 transmission facilities in the USA and Canada DS1 format, which is a 1.544 Mbps digital signal that consists of a 1.536 Mbps signal multiplexed with an 8 kbps framing signal.

DMI Tie Trunks

DMI tie trunks use Bit-Oriented Signaling or ISDN-PRI Message-Oriented Signaling to interface with a host computer, another PBX, or a public or private network.

The TN722B DS1 Tie Trunk circuit pack supports Voice-Grade DS1, AVD DS1, and DMI tie trunks in the Bit-Oriented Signaling modes. In addition, the TN767 and TN464 DS1 circuit packs supports DMI tie trunks in the ISDN-PRI signaling mode. For details on the DMI signaling modes, see the "Digital Multiplexed Interface" feature description, elsewhere in this section.

This type of tie trunk uses DS1 transmission facilities in the USA and Canada DS1 format, which is a 1.544 Mbps digital signal that consists of a 1.536 Mbps signal multiplexed with an 8 kbps framing signal.

When the DS1 circuit pack is assigned ISDN-PRI signaling, Robbed Bit Signaling and ISDN-PRI signaling can be used over the same DS1 interface. Trunk groups administered with a communication type of ‘‘voice’’ can use Robbed Bit signaling and trunk groups administered with a communication type of ‘‘isdn’’ can use the ISDN-PRI signaling.

CO, FX, and WATS Trunks

When the TN767 or TN464 DS1 circuit pack interface is used to provide CO, FX, or WATS trunk group service with incoming and outgoing types of ground start or loop start, Robbed Bit Signaling must be used.

When the DS1 interface is used to provide CO, FX, or WATS trunk group service with incoming/outgoing dial types administered as auto/auto, auto/delay, auto-immediate, or auto/wink, either Common Channel Signaling or Robbed Bit Signaling can be used.

When the DS1 interface is used to provide CO, FX, or WATS trunk group service with incoming and outgoing types of ground start or loop start, outgoing trunk calls do not receive answer supervision. Instead, the answer supervision is faked by the DS1 circuit pack based on a timer administered on the “Trunk Group” form.

Remote Access Trunks

Signaling for remote access trunks is depends on the trunk group and incoming/outgoing dial types.

DID Trunks

When the DS1 interface is used to provide DID trunk group service, Robbed Bit Signaling must be used.

Off-Premises Stations

DS1 off-premises stations do not receive system message waiting indications.

When the DS1 interface is used to provide off-premises stations, Robbed Bit Signaling must be used.

Access Endpoints

Individual channels of a DS1 can be assigned extensions with the Access Endpoint function. Access Endpoints are discussed in the Administered Connections feature description elsewhere in this manual.

Considerations

DS1 tie trunks offer voice and data transmission, via DS1 digital facilities, at lower cost and faster speed than conventional analog trunks. Data transmission costs are lower than if large analog trunk groups are used. In the future, digital transmission is expected to cost less, thus adding to savings over analog facilities.

Interactions

The following features interact with the DS1 Trunk Service feature.

- CAS

  AVD tie trunks and RLTs can share the same DS1 interface. RLTs cannot, however, be administered or a communication type of avd on the RLT Trunk screen.

- DCS

  AVD DS1 tie trunks can only be used in a DCS between two DEFINITY system Generic 1s or between a DEFINITY system Generic 1 or Generic 3 and System 75, System 85, or DEFINITY system Generic 2.1.

- ETN

  AVD DS1 tie trunks can only be used in an ETN between two Generic 1s or between Generic 1 or Generic 3 and System 75, br System 85, or DEFINITY system Generic 2.1.

- Modem Pooling

  When AVD DS1 tie trunks are used, a conversion resource is not automatically inserted into the connection because the system cannot determine whether the transmission is voice or data. A conversion resource is connected between Voice-Grade tie trunks and digital endpoints.

- Private Network Access

  AVD DS1 tie trunks cannot be used as EPSCS or CCSA access trunks.

Administration

DS1 trunks are assigned on a per-line basis by the System Manager. The following items require administration:

- DS1 Circuit Pack — Assign the circuit pack to the system before administration of the associated trunks.
- Synchronization Plan — Administer to provide synchronization between the switch's DS1 circuitry and the digital facilities that the switch is connected to.
- Trunk Groups — Associate the trunks to groups, if desired.

Hardware and Software Requirements

One TN722, TN767, or TN464 DS1 circuit pack is required for every 24 trunks using Robbed Bit Signaling or for every 23 trunks using Common Channel Signaling. If ISDN-PRI signaling is used, a TN765 Processor Interface circuit pack is required for G3vs/G3s and G3i (or for G3i the TN778 Packet Control) in addition to the DS1 circuit pack. G3r requires the TN1655 packet interface (PKTINT) circuit pack.

A TN464 circuit pack is required to support 32 trunks.

No additional software is required.

Do Not Disturb

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows guests, attendants, and authorized front desk voice terminal users to request that no calls, other than priority calls, terminate at a particular extension number until a specified time. At the specified time, the system automatically deactivates the feature and allows calls to terminate normally at the extension.

Do Not Disturb is a form of Termination Restriction that is associated with an automatic deactivate time. When Do Not Disturb is active at an extension number, the user will receive only those calls associated with the Automatic Callback, Automatic Wakeup, and Priority Calling features, and those calls that are redirected to that extension via the Call Coverage and Call Forwarding All Calls features. All other call attempts will redirect to a recorded announcement, an attendant, or intercept tone, as specified by the System Manager through administration.

This feature may be dial or button accessed from voice terminals equipped with touch-tone dialing or button accessed from attendant consoles and front desk terminals. Users with rotary-dial terminals must call the attendant or front desk user to request Do Not Disturb.

When the Do Not Disturb feature is activated, the system supplies voice prompting to voice terminal users or display prompting to attendants and front desk users.

Feature Activation by Voice Terminal Users

A voice terminal user can activate Do Not Disturb by dial access or by button access if a Do Not Disturb button is assigned to the voice terminal. If dial access is used, the system automatically deactivates the feature at the requested time. If button access is used, deactivation is not automatic.

- Dial Access

  When a user dials the Do Not Disturb feature access code, the system generates voice messages (through the use of a Speech Synthesis circuit pack) that direct the user to enter a deactivate time. The DTMF buttons on the voice terminal are used to enter this information. The user may later change or delete the request by dialing the Do Not Disturb FAC again and entering the required information.

If invalid entries are made (such as 32 for the deactivate time) or if system conditions (such as all voice synthesis ports busy) prevent entry of a Do Not Disturb request, the system informs the user to dial the attendant or front desk assistance.

- Button Access

  If a voice terminal has a Do Not Disturb button, the user can press the assigned button to activate the feature. The handset may be on-hook or off-hook; voice prompting is not required. The user must press the button a second time to deactivate the feature.

  The lamp associated with the Do Not Disturb button lights when the feature is activated and remains lighted until the feature is deactivated. An automatic deactivate time is not provided through button access.

Feature Activation by Attendant

The attendant (or authorized front desk user) can activate Do Not Disturb for a user or a group of users. (The assigned COR determines which users are in the group.) The attendant presses the Do Not Disturb — Extension or the Do Not Disturb — Group button. After pressing the Extension button, the attendant enters an extension number; after pressing the Group button, the attendant enters an appropriate COR number.

System prompts appear on the display to direct the attendant on what information to enter, and a displayed message notifies the attendant when the request is confirmed. If a Do Not Disturb request is denied, a displayed message, including a reason for the denial, informs the attendant.

The attendant can cancel a Do Not Disturb request by activating the feature, entering the desired extension number or group COR number, and pressing the Delete button.

Activation of Do Not Disturb Through a PMS

The system provides an interface to a PMS. This interface can allow activation and deactivation of controlled restrictions. Activation of Do Not Disturb through a PMS is similar to that of Termination Restriction. A scheduled deactivate time is not specified.

Audit Trail Reports

The system keeps an audit trail record of all voice terminals that are in the Do Not Disturb mode and that have Termination Restriction activated. The System Manager or other delegated administration personnel can request a listing of this information to be displayed at a terminal or to be printed at a designated printer.

The following reports can be administered for printing on a daily basis:

- Do Not Disturb Status Report — This report lists all extension numbers with Do Not Disturb active. The specified Do Not Disturb deactivate time for each extension number is also listed.
- Do Not Disturb Plus COR Status Report — This report lists all extension numbers as defined above, plus a list of those extension numbers whose Controlled Restriction level is Termination Restriction. (Termination Restriction is activated by the attendant for a specific extension or COR. A deactivate time is not associated with Termination Restriction.)

Audit trail records do not include Do Not Disturb information for extensions that are both termination and Outward Restricted.

Considerations

The Do Not Disturb feature lessens the attendant's work load since each voice terminal user can activate the feature. Also, through the voice messages supplied by the system, the user is assured that his or her request is confirmed.

The Do Not Disturb deactivate time may be requested using standard time or military time. If standard time is entered, the system will prompt the user to enter a.m. or p.m.

A front desk user must have console permission COS in order to activate this feature.

The number of available speech synthesis ports is the only limit on the number of voice terminal users receiving voice prompting at the same time.

Interactions

The following features interact with the Do Not Disturb feature.

- Automatic Callback

  Do Not Disturb does not block an Automatic Callback call. Return calls will terminate at a voice terminal in the normal way.
- Automatic Wakeup

  An Automatic Wakeup call deactivates Do Not Disturb and alerts the guest at the specified time.
- Call Coverage

  If a point in a coverage path has Do Not Disturb active, calls covering to that coverage point extension still alert that extension unless it has controlled restriction termination activated. When the feature is set to "coverage" and a station does not have a coverage, the calls are routed to the attendant.

- Call Forwarding All Calls

  If Do Not Disturb is active at the forwarding extension, the caller will receive intercept treatment. If Do Not Disturb is active at the forwarded-to extension, the call alerts the forwarded-to extension.

- Controlled Restriction

  When a station has total controlled restriction, it cannot receive or place any calls. However, it can receive a call if another station has an auto-icom button pointing to the controlled restriction station.

- Internal Automatic Answer (IAA)

  Activation of the Do Not Disturb feature preempts IAA at the called voice terminal.

- PMS Interface

  Check-Out from either a PMS or the switch automatically deactivates Do Not Disturb for the specified extension number.

Administration

Do Not Disturb is administered by the System Manager. The following items require administration:

- Feature Access Code (one code can be used to activate and deactivate Do Not Disturb)
- Do Not Disturb button (per voice terminal, optional)
- Do Not Disturb — Extension button (per attendant console or front desk terminal)
- Do Not Disturb — Group button (per attendant console or front desk terminal)
- Intercept treatment for call attempts to a terminal with Do Not Disturb active — Choice is one of the following: intercept tone, recorded announcement, coverage, or attendant

Hardware and Software Requirements

Requires a TN725 Speech Synthesis circuit pack if voice prompting is used. Each circuit pack has four ports to provide voice prompting.

No additional software is required.

E1 Trunk Service

Feature Availability

This feature is available with G3i-Global, G3V2, and later releases.

Description

Provides the same service as DS1 Trunk Service at 2.048 mbps rate and with 32 channels. See the "DS1 Trunk Service" feature.

EIA Interface

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides an alternative for host connections and analog voice terminal users who use simple data terminals or Personal Computers (PCs) which emulate simple data terminals. Also for interconnection between an EIA-232 compatible DTE and the system. However, it is not for use with PCs using PC communications packages. The EIA Interface consists of a Data Line circuit pack port and an ADU.

The EIA Interface supports speeds of LOW, 300, 1200, 2400, 9600, and 19200 bps.

A data line port differs from a data module in that the functions (options) are set in the system rather than at the physical hardware. The user does not have physical access to the data line port, but has access to all data module related functions; that is, the user can examine and change such items as speed, parity, and so on, via a menu-driven selection mode at the DTE, if permitted by the administrator. Also, the Management Terminal can be used to examine and change the functions.

A data line port in conjunction with an ADU can be used to connect the system to the ISN. The ISN consists of packet data switches that support data calls between data endpoints. Data line ports provide the most economical access to the ISN. Available ADUs are the Z3A1, Z3A2, Z3A3, and Z3A4.

DCE may also be connected to a data line port by use of a null modem.

Considerations

The system’s EIA Interface support offers a convenient and lower-cost alternative for host connections and for analog users with simple data terminals. DTEs can connect directly to a Data Line circuit pack which functions as a data module connected to a Digital Line port. Since the user does not have physical access to the data module, all related data module options are settable from the DTE. With a density of eight data line ports per circuit pack, each port provides connections of user’s asynchronous EIA-232 compatible DTE.

There is no limit to the number of Data Line circuit packs the system can support, subject to slot availability and the system limit of digital data endpoints.

Flow control signaling is not provided.

Interactions

The following features interact with the EIA Interface feature.

- Data Terminal Dialing

  Data Terminal (Keyboard) Dialing permission must be granted before Keyboard Dialing can be accessed.

  Access to ISN endpoints requires two-stage dialing, the first stage consisting of dialing a hunt group extension number to access ISN, then the second stage consisting of an ISN address.

Administration

EIA Interface support is assigned on a per-data terminal basis by the System Manager. The following items require administration:

- Data Line circuit pack — Assign a vacant port, port options, and permissions on the circuit pack to the associated DTE.
- Data Extension buttons — Assign Data Extension buttons to multi-appearance voice terminals.

The following permissions can be administered on a data line port to allow DTEs to be used.

- Keyboard Dialing (KYBD) — Must be set to allow data endpoints to receive and send text during data call origination or termination. Text prompts are provided.
- Configuration — Must be set to allow DTEs to change their data module options; that is, examine and change options, such as speed, from the DTE. Keyboard Dialing permission must be granted first.
- Busy-Out — Should be set for DTEs that are members of a hunt group, and to allow busy out (when DTE turns power off) so that calls will not terminate on that DTE.

The following options can be examined and changed from the DTE if the configuration permission has been granted:

- Speed — All speeds (up to 19.2 kbps) at which the DTE can operate are selectable, including Autoadjust. Autoadjust is the capability of the data line port to determine what speed and parity the associated DTE is transmitting at and match it for terminal dialing and/or text feedback purposes.
- Parity — All choices of parity (even, odd, mark, or space) can be selected.
- Permit Mismatch — The EIA Interface may be operated at a different transmission speed rate than the rate of the far end data module. This allows for calls between digital endpoints with different speeds without changing the speed of the DTE.

- Dial Echoing — Can be set to echo typed characters back to the DTE during dialing.
- Disconnect — Set the signal to indicate disconnect. Choices are one break greater than two seconds or two breaks within one second.
- Answer Text — Can be selected when DTE is an intelligent device to allow text messages to be delivered to the DTE from the system when a call is being answered; also, applies to text generated by the data line circuit. The following call progress messages may be answered:
  - — INCOMING CALL
  - — PLEASE ANSWER
  - — TRANSFER
  - — FORWARDED
  - — ANSWERED
  - — ABANDONED
  - — DISCONNECTED
  - — OTHER END
- Connected Indication — Can be set to allow the text ‘‘CONNECTED SPEED= XXXX’’ to be sent to the DTE when the data call has been established.
- Other Characteristics — The Data Line circuit always operates in automatic answer (provided the DTE is on), asynchronous, and full duplex modes. The ‘‘Loss of Carrier Disconnect’’ is set to off; that is, the Data Line circuit, unlike other data modules, does not disconnect upon loss of EIA updates in the previous four seconds.

Hardware and Software Requirements

One TN726 Data Line circuit pack is required for each eight EIA interfaces provided. One ADU is required for each port on the circuit pack.

No additional software is required.

Emergency Access to the Attendant

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by the system or can be dialed by system users. Such calls can receive priority handling by the attendant.

Emergency calls to the attendant can be placed in the following ways:

- Automatically by the system

  If a voice terminal has been assigned the Off-Hook Alert option via COS, and if that voice terminal is left off-hook until the intercept timeout occurs, the (administrable) off-hook alerting timer is started. If the station is still off-hook when the off-hook alerting timer expires, an emergency call is automatically placed to the attendant.

- Dial access by a system user

  A system voice terminal user can place an emergency call to the attendant by dialing the Emergency Access to the Attendant feature access code.

When an emergency call is placed, one of the available attendant consoles receives visual and audible notification of the call. However, if all attendants are busy, the call enters a unique queue for emergency calls. This queue allows attendants to handle emergency calls separately from other calls. If the queue is full, the call, if administered to do so, can be redirected to another extension.

An emergency call causes the following to occur:

- The system selects the first available console to receive the call, even if the call first entered the emergency queue.
- The Emergency tone alerts the selected attendant and the lamp associated with the Emergency button, if assigned, lights at that attendant console. If the console is an older console (does not have emergency tone capability), normal ringing is heard and the display flashes.
- When the call arrives at an available console, the attendant display shows the following:
  - The call appearance that received the call
  - The calling party identification

— The calling party extension number

— The number of emergency calls remaining in queue

An attendant can place a normal call on hold in order to receive an emergency call.

An audit record is created for each emergency call event. This record includes the following:

- The extension number and name where the call was originated
- The extension number of the attendant or attendant group that answered the call
- The time the call was originated
- One of the following known call results:

  — Call Completed

  Call answered at attendant or LDN night extension.

  — Queue Full

  Emergency access queue is full; tries to redirect call to an emergency access redirection extension.

  — No Attd

  No active attendants are available to receive the emergency call; tries to redirect the call to an emergency access redirection extension.

  — Redirected Answered

  Call answered by emergency access redirection extension.

  — No Redirection Ext.

  Could not redirect call to emergency access redirection extension because none are administered.

  — Attd Night Service

  System is in night service. Will try to redirect call to attendant night service.

  — Call Abandoned

  Caller drops call before it was answered. Call was either waiting in the attendant emergency queue, ringing at an attendant console, or ringing at the LDN night extension.

  — Redirected Abandoned

  Caller drops call before it was answered. Call had previously been redirected to the emergency access redirection extension.

The emergency audit records are used to generate an Emergency Access Activity Report. This report summarizes each emergency access call that was attempted during the past 24 hours. This report can be scheduled for printing once a day at a designated printer. Also, if the system has a journal printer, Emergency Access to the Attendant events are printed as they occur.

The System Manager can monitor emergency access call events by displaying them at the administration terminal. The command for listing emergency call events is **list emergency.** Also, a **from** and **to** time option can be used with the command. For example, if the command **list emergency from 8:00 a.m. to 12:00 p.m.** was entered, all emergency call events are printed at a designated printer.

Considerations

Emergency Access to the Attendant provides a way for users to quickly and easily get in touch with an attendant. This results in more efficient call handling for users.

When the emergency queue is full, any overflow should be redirected to an extension number or another queue.

The Emergency button on the console has no function other than to provide a visual indication of an incoming emergency call.

The unique Emergency tone cannot be silenced except by answering the emergency call.

The system should have at least one day and one night attendant (or night service station) for this feature to be useful at all times of the day.

Interactions

The following features interact with the Emergency Access to the Attendant feature.

- CAS

  If the system is a branch location and if CAS is in effect, an emergency call is rerouted to the branch attendant group. If the branch does not have an attendant or if the branch is not in CAS Backup Service, the call is denied.

  If the Branch PBX is in CAS Backup Service, an emergency call routes to the backup position and is be treated the same as any other non-emergency call.

- COR

  If the calling voice terminal is assigned Origination Restriction, an emergency call attempt is denied. However, an Emergency Access to the Attendant call overrides any other calling party restrictions, including any controlled restrictions.

- Individual Attendant Access

  An Emergency Call to the Attendant cannot be placed to an individual attendant.

  An Emergency Call to the Attendant does not have priority over a call to an individual attendant (For G3iV1 and G3vsV1/G3sV1).

- Intercept Treatment

  The Intercept With Off-Hook Alert option automatically activates Emergency Access to the Attendant.

- Inter-PBX Attendant Service

  If the system is a branch location and if Inter-PBX Attendant Service is in effect, an emergency call is rerouted to the local Branch PBX attendant group. If the branch does not have an attendant or if the attendant is not on duty, the call is denied.

- Night Service

  When Night Service is in effect, Emergency Calls to the Attendant route to the night destination. Such calls are included on the Emergency Audit Record, and the call is designated as “Emergency Night’’ in the audit trail.

  When an attendant is in night service, either a night station or redirect extension must be assigned. Otherwise emergency access calls to the night atendant hear a busy tone.

- Priority Queue

  The priority queue administration can change the priority of an Emergency Access to the Attendant call to an equal or lower priority than other types of calls. (For G3r, G3i-Global, G3V2, and later releases)

- Remote Access

  An Emergency Call to the Attendant cannot be placed through Remote Access.

- Restriction—Controlled

  An Emergency Access to the Attendant call overrides any Controlled Restriction.

Administration

The Emergency Access to the Attendant feature is optional on a per-system basis. The following items require administration by the System Manager:

- Emergency feature access code
- Emergency button and associated lamp (per attendant console)
- Emergency queue length
- Permission to activate Emergency Access to the Attendant via off-hook alert (per Class of Service)
- Extension number where the emergency queue overflow redirects the call.
- Interval the intercept tone is applied before the emergency call is placed

In addition, the journal printer and the time for the system to print the Scheduled Emergency Access Activity Report are administered through the "Feature-Related System Parameters" form.

Hardware and Software Requirements

No additional hardware or software is required.

End-to-End Signalling

Feature Availability

This feature is available with all G3 releases.

Description

End-to-end signalling allows a DTMF station to send DTMF digits over a trunk after the far-end has answered. This allows a rotary station user to access equipment, such as AUDIX, that is controlled by DTMF digits. The trunk can be either rotary or tone signalling type. For a rotary signalling type, addressing digits are sent over the trunk as rotary. After addressing is complete and the call is connected, any additional digits are sent as DTMF. For most situations the end-to-end signalling DTMF digits are generated at the calling DTMF station and passed directly to the outgoing trunk, but switch-generated DTMF end-to-end signalling digits are transmitted if the calling station is a 73nn-series hybrid terminal and/or the calling station user had used either the Abbreviated Dialing or Last Number Dialed feature to dial the end-to-end signalling digits. See the "Last Number Dialed" or "Abbreviated Dialing (AD)" features for examples of using end-to-end signalling.

Interactions

None.

Considerations

None.

Administration

None.

Hardware and Software Requirements

None.

Enhanced 84xx Display

Feature Availability

Enhanced 84xx Display is available with G3V4 Release 3 and later releases.

Description

The Enhanced 84xx Display feature allows users to administer the switch to display information and soft key labels using the Katakana alphabet and various European characters on 84xx series display terminals. The system accepts entries as Roman characters preceded by a tilde (~), and maps them to corresponding Katakana or European characters.

The OPTREX character set that displays on 84xx series voice terminals can be divided into two different groups. Group 1 of the OPTREX character set contains the Roman alphabet, numerals and special characters, found on the standard US English keyboard. Group 2 contains Katakana characters, as well as some European characters and other symbols. The enhanced display feature allows users to administer the system to display either group.

End User Operation

Users activate the feature by setting the value of the Enhanced 84xx Display field on the System-Parameters Country-Options form to “y”. Once this value is set, the administrator can enter the Roman characters that produce the corresponding Group 2 characters on the terminal display.

In order for Group 2 characters to display, the administrator must make the proper entries at the form level and submit the form. Users can administer name fields and soft key labels to display in the character set of choice.

Security Measures

This feature does not require any additional security measures.

Considerations

After the feature is turned on at the system parameter level, the user must still enter the display field values on the appropriate forms and submit those forms before the enhanced character set will display. The reverse is also true, in that if the feature is turned off at the system level after Group 2 display has been administered, the user must change the display field values to the appropriate Group 1 characters.

Interactions

Internal Feature interactions

Directory . The Enhanced 84xx Display feature will interact with the Directory feature to include names administered with OPTREX characters in the Directory. Terminal users can select directory entries displayed in the Group 2 character set by using the “Directory” and “Next” buttons on their display terminals. However, they cannot input directory entries using the voice terminal because of the special characters used.

ISDN. OPTREX characters are supported for ISDN calls, and if administered properly, will display correctly for calls between two DEFINITY PBXs.

Data Call Setup Dialing. Not supported.

Message Retreival - Print Messages (Demand Print). Not supported.

Leave Word Calling - Adjunct. Not supported.

OSSI. Does not malfunction, but displays the literal value of the display field, not the Group 2 characters.

Features with Display Interactions. All features that display administered data are supported for OPTREX, provided the terminal hardware is compatible.

External Feature Interfaces

AT&T Adjunct Switch Interface. The information sent from the terminal to any adjunct is useless because the adjunct receives the literal value of the display field, not the OPTREX display format. The tilde is not a recognized character, so the resulting display appears as a string of random characters, for example, as “2<@^.“

The ASAI-Query Names Database feature also will also recieve names as mentioned above, making this feature useless. The same is true of the ASAI-Accessed BCMS Data feature.

Call Management System. Not an issue, since CMS maintains its own database.

AUDIX. If AUDIX uses DCIU and/or Control Link Integration, special characters are not a problem and the AUDIX platform and software version are irrelevant. However, for any release earlier than 3.2, special characters cause the system to hang. Older releases should be upgraded to DEFINITY AUDIX R3.2.

The following AUDIX products operate normally with OPTREX enabled:

- Basic R1 AUDIX

- Embedded DCIU Audix
- Embedded DCP Audix (R3.2 or greater)

AUDIX Voice Power/Audix Voice Power Lodging. Not supported.

Property Management System. Not applicable.

Cost Mangement (CDR). Not applicable.

Miscellaneous Adjuncts. Not applicable.

Message Manager. Not an issue, since Message Manager receives its information from AUDIX.

System Management (G3-MA). Not applicable.

Passageway Direct Connect. Not supported.

Novell Telephony Services. Not supported.

Look Ahead Interflow. Supported.

VUStats. If this feature is used with non-84xx terminals, OPTREX may cause screens to be cleared or display false information.

DCS Interworking. OPTREX is supported in a DCS environment. All switches in the network must have the enhanced character set enabled *and* the same software load must be installed on each switch. The software versions do not have to be the same. For example, V2 could have a bugfix load installed and function properly.

Conversant. Not applicable.

Monitor 1 and OneVision. Not supported.

Trouble Tracker. Not applicable.

ECMA QSIG Networking. Not supported between a DEFINITY and non-DEFINITY switch. If two DEFINITY switches are administered for OPTREX, this feature is suppproted.

Administration

This feature is activated on a system-wide basis using the System-Parameters County-Options form. Once activated, the administrator must update those forms that determine the contents of display fields.

The following forms contain fields that will accept the enhanced display characters:

- access-endpoint
- agent-loginID
- announcements
- attendant
- bcms-vustats loginIDs
- console-parameters
- data-module
- display-messages
- display-messages auto-wakeup-dn-dst
- display-messages call-identifiers
- display-messages date-time
- display-messages leave-word-calling
- display-messages malicious-call-trace
- display-messages miscellaneous-features
- display-messages property-management
- display-messages softkey-labels
- display-messages time-of-day-routing
- display-messages vustats
- hunt-group
- listed-directory-numbers
- paging loudspeaker
- personal-CO-line
- pri-endpoint
- station
- system-parameters-hospitality
- term-ext-group
- trunk-group
- vdn
- vustats-display-format

Hardware and Software Requirements

The Enhanced 84xx Display feature requires one of the following voice terminals:

- 8400 Series DCP Terminals (8410D, 8411D, 8434D)
- 603E1 Callmaster III
- 9400 Series DCP Terminals (9410, 9434); used only in Europe

NOTE:
Aliased 84xx series terminals are not supported with this feature.

Enhanced Abbreviated Dialing (EAD)

Feature Availability

Enhanced Abbreviated Dialing is optional with all releases except G3vs/G3s ABP. Enhanced Abbreviated Dialing is not available with G3vs/G3s ABP.

NOTE:
Abbreviated Dialing is available with all Generic 3 releases.

Description

Enhanced Abbreviated Dialing (EAD) option supplements the standard Abbreviated Dialing feature and provides the Enhanced Number List. Provides an upgraded version of Abbreviated Dialing via inclusion of the Enhanced Number List option. One Enhanced Number List is allowed per system in addition to the System Number List. The Enhanced Number List can contain any number or dial access code. The System Manager programs the Enhanced Number List and determines which users can access the list.

List entries for the Enhanced Number list are numbered 000 through 999 in G3V3 and earlier releases. With G3V4 and later releases, the system can be administered to have list entry numbers of either 000 through 999 or 0000 through 9999.

An Enhanced Number List can be designated as Privileged by the System Manager. Calls automatically dialed from a Privileged List are completed without Class of Restriction or FRL checking. (FRLs are associated with the Automatic Route Selection and Automatic Alternate Routing features.) This allows access to selected numbers from which certain voice terminal users might otherwise not be able to dial manually. For example, a voice terminal user may be restricted from making long-distance calls. However, the number of another office location may be long-distance, and it could be entered in a list designated as Privileged. The user could then call the office location using Abbreviated Dialing while still being restricted from making other long-distance calls. For details, see the "Abbreviated Dialing (AD)" feature description earlier in this chapter.

Considerations

See the "Abbreviated Dialing (AD)" feature.

Interactions

See the"Abbreviated Dialing (AD)" feature.

Administration

See the "Abbreviated Dialing (AD)" feature.

Hardware and Software Requirements

Optional software is required to receive this feature.

Enhanced ICSU

Feature Availability

This feature is available with Generic 3 V3, release 3.0. The AT&T CPE Loopback Jack equipment is not available with this release, but is planned for a future release.

NOTE:
It is not necessary to upgrade to release 3.0 to receive default CSU functionality from the ICSU. However, the default integrated CSU options on page two of the "DS1 Circuit Pack" form cannot be changed without release 3.0.

Description

The Integrated Channel Service Unit (ICSU) is the combination of an AT&T 120A CSU module and a TN464E, TN767D, or later suffix DS1 circuit pack.

NOTE:
The term 120A refers to the 120A1 and later suffix CSU modules.

NOTE:
The TN767 and the TN464C and later revisions of the DS1 circuit packs provide a DSX1 interface.

The DS1 circuit pack automatically initializes and tests the 120A, and detects on-line alarms. The 120A CSU in combination with the DS1 circuit pack provides the essential functions that have historically been provided by external CSUs. For more information about the ICSU, see *DEFINITY Communications System Generic 1, Generic 2, and Generic 3 V1 and V2 Integrated CSU Module Installation and Operation*, 555-230-193.

The Enhanced ICSU functionality provides for the administration and maintenance of the ICSU by local and remote personnel. It also provides for additional performance measurements of a DS1 facility and for new loopback tests. Enhanced ICSU is only available for use in the USA with 1.544 Mbps DS1 service. It requires a 120A CSU module with the TN767E, TN464F or later suffix DS1 board.

NOTE:
A CSU module is not required for the Enhanced ICSU functionality for performance measurements or the on-demand long-duration loopback tests.

The following testing and performance measurement functions can be performed with Enhanced ICSU.

Testing

Provides for:

- On-demand short-duration loopback testing of the DS1 board and CSU Module, including board loopback, equipment loopback, and repeater loopback tests
- On-demand long-duration loopback testing of building wiring to the CPE Loopback Jack
- On-demand long-duration loopback testing to the far-end CSU
- On-demand long-duration one-way span testing to and from remote test equipment or another DEFINITY Communications System
- The ability to inject bit errors in long-duration loopback test patterns

The enhanced feature allows users to view long-duration loopback/span test results during testing. Enter the list measurements ds1 summary command to view these results on the DS-1 Link Performance Measurements Summary Report. The summary report provides loopback/span test information about: the type of test being run; the pattern being sent; whether the DS1 board has synchronized to the looped test pattern; the bit-error count; and the test duration. It also provides performance information, described below. See the *DEFINITY Communications System Generic 3 Version 4 Traffic Reports*, 555-230-511, for more detailed information including a sample summary report.

At any time during testing, the Loopback/Span Test Bit-Error Count and Test Duration, and/or “ESF Error Events” fields on the summary report can be cleared. Also, at any time during testing the loopback test can be terminated.

Performance Measurements

The following performance measurements have been added to the DS-1 Link Performance Measurements Summary Report. An **N/A** in any of these fields indicates that enhanced functionality is not activated.

- Controlled Slip Seconds (CSS)
- Loss Of Frame Count (LOFC)
- The “Failed Seconds (FS”) field is replaced by Unavailable/Failed Seconds (UAS/FS)
- ESF Error Events

The following performance measurements have been added to the DS-1 Link Performance Measurements Detailed Log Report. An “N/A” in any of these fields indicates that enhanced functionality is not activated.

- CSS (Controlled Slip Seconds)
- LOFC (Loss Of Frame Count)
- The “FS (Failed Seconds)” field is replaced by UAS/FS (Unavailable/Failed Seconds)

Performance measurement reports can be created for local, carrier-local and remote facilities.

- Local reports display user measurements. These measurements are the user copies of the local (near-end) performance measurements and can be cleared by the user. They cannot be cleared by the carrier.
- Carrier-local reports display network measurements. These measurements are the carrier copies of the local (near-end) performance measurements. They can only be cleared by the carrier.
- Remote reports display remote CSU measurements. These measurements are available from the CSU at the far end of the link. They can be cleared from the near end of the link.

See the *DEFINITY Communications System Generic 3 Version 4 Traffic Reports,* 555-230-511, for additional information.

Considerations

The DS1 circuit pack must be taken out of service with the busyout board command before loopback/span tests can be run on-demand.

Only one loopback/span test can be active at any one time on a DS1 facility.

If the DS1 circuit pack is currently functioning as part of a network requested loopback test, loopback/span testing will be denied.

If the DS1 span is being used as the clock reference for the switch, a different reference must be selected before the DS1 span is looped.

Interactions

None.

Administration

The ICSU can be administered for connection to any DS1 service.

The Enhanced ICSU is administered, on the “DS1 Circuit Pack” form, by entering the add ds1 or change ds1 command. On the “DS1 Circuit Pack” form, when the Near-End CSU Type is identified as “integrated,” additional CSU module administration options become available. If the Framing Mode is “esf,” the user can administer options for the ESF Data Link. See *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for more detailed instructions for administering the Enhanced ICSU.

Hardware and Software Requirements

Enhanced ICSU requires the use of a 120A CSU Module with the TN767E, TN464F, or later suffix DS1 board.

Enhanced DCS (EDCS)

Feature Availability

This feature is available with G3i-Global, G3V2, and later releases. EDCS is not available with G3vs/G3s ABP.

Description

Enhanced DCS adds features to the existing DCS capabilities. Additional features include exchanging information to provide class of restriction checking between PBXs in the EDCS network, providing call progress information for the attendant, allowing attendant intrusion between a main and a satellite, and allowing a main PBX to provide DID/CO intercept treatment rather than the satellite PBX.

Considerations

If the DCS link fails, the administrator can choose to allow calls to continue without class of restriction checking or to block all DCS calls to inward-restricted stations.

Enhanced DCS is not available with DCS over an ISDN-PRI D-channel. When EDCS is used, all nodes in the DCS network must use EDCS.

Interactions

- Class of Restriction

  When a call goes to coverage, it is the called party's (not the covering party's) restrictions that will be used.

Administration

This feature is enabled with the System Parameters Feature form.

Hardware and Software Requirements

This feature requires the same hardware as DCS.

Expert Agent Selection (EAS)

Feature Availability

Expert Agent Selection (EAS) is available with Generic 3 Version 2 and later releases.

Purpose

The EAS feature is an optional feature that allows certain EAS Skill types to be assigned to a call type or Vector Directory Number (VDN). Routing incoming calls through a VDN then allows the system administrator to direct calls to agents who have the particular agent skills required to complete the customers' inquiry successfully.

The EAS provides a method for call managers to match the needs of a caller to the skills/talents of the agents. This arrangement ensures the best possible service to the caller.

The EAS consists of the following capabilities:

- Call distribution based on Skill provides for matching the VDN Skills required to handle a particular call (based on VDN) to an agent who has at least one of the agent Skills that the caller requires.
- EAS agent capabilities are not administered in relation to a physical voice terminal. Instead, agents are administered in relation to their personal login IDs, each with its own assigned capabilities. EAS agents can use their login IDs to login to any voice terminal in the system. This ability is referred to as the EAS Logical Agent Capability.
- Direct Agent Calling from a voice terminal allows users to call a particular agent and have the call treated as an ACD call.

Identifying Caller Needs

Caller needs are identified by information passed from the network in DNIS digits or ISDN messages, by call prompting digits or digits entered at a VRU (voice response unit), or using ASAI in a host database lookup. A Skill number is assigned to each need or group of needs. For example, a caller may need to place a catalog order (Skill 22), ask about a bill (Skill 31), or talk with someone who speaks Korean to place an order (Skill 122).

AT&T Network DNIS/ISDN Called Party

The AT&T network routes the call to a number based on the type of service dialed (e.g., 800 number), the state/city the call originates in, the time of day, a percentage allocator, or combinations of these. Information on the type of service dialed and the origin of the call is used to determine the needs of the caller.

Call Prompting/VRU Digits

The call prompting/VRU digits are entered in response to any question about callers' needs. For example, a hotline for a product may request that a product code be entered, or a travel service may request a two-digit state code to indicate the state the caller is interested in traveling to.

ASAI Host Database Lookup

The ASAI host database lookup uses DNIS and ANI (calling party's number) to determine what skills are required or even the agent desired. For example, the database may show that the caller speaks Spanish or that the caller has been working with Agent 1367.

Routing Digits

Routing digits received by the switch from DNIS, ISDN, prompting, a VRU, or ASAI routing are typically assigned a VDN. A VDN is an extension number that points to a vector routing table. Each VDN can have up to three skill preferences (skill numbers) corresponding to the callers' needs.

Agents Skills

Agents are trained or hired based on meeting the callers' needs. Agents are assigned Skill numbers based on training or knowledge, access to systems or information, language (multi-lingual skills), interpersonal skills, etc. Agents are assigned up to four skills or sets of skills. Examples of agents' skills are: speaks Spanish, knows about widget X, can handle complaint calls, knows about insurance laws for New York, or has access to a database.

If a single caller speaks with an agent who has multiple skills (e.g., speaks Spanish and has access to the billing system, or knows about widget X and can handle complaint calls), then a single skill group is created for each set of skills.

EAS Application

Assume a call makes reservations for a chain of resorts. Callers have historically said that they prefer to talk to someone who personally visited the resort. The company also determines it is easier to sell additional sightseeing packages if the agent has a regional accent.

The resort company places a variety of ads in magazines for information on a particular resort or a state. Callers responding to these ads dial a number that directly routes them to a VDN for that state's resorts.

Callers also dials a general number for the resort chain. A vector prompts the callers to enter a two-digit state code for the state they wish to visit.

If the caller dials the general number and enters the Texas state code, the VRU returns routing digits 3222 (a VDN) to the vector. Callers who respond to an ad for a Texas resort are directly routed to VDN 3222. Table 3-56 shows the skill preferences assigned to Texas VDN 3222.

Table 3-56. Skill Preferences

Texas VDN 3222 - Skill Preferences		
1st:	30	Agent who has Texas accent and has been to resorts in Texas
2nd:	31	Agent who has been to resorts in Texas
3rd:	100	Any agent who can take a reservation

Each state resort has a VDN and a set of skill preferences.

When callers make a reservation at a resort they are given a call-back number. When someone dials the call-back number, call prompting requests the reservation number and passes this to the host in an ASAI adjunct routing request. The host does a database lookup for the agent who made the reservation and the resort corresponding to the reservation. If the agent is currently logged in, the vector routes the call to the agent, or else the vector routes the call to the state's VDN for the resort. When the call is delivered to an agent, the current reservation is automatically displayed at the agent's data terminal. This provides quicker and more personal service.

Writing Vectors

The VDN routes the call to a vector. The steps in the vector determine how the call is processed.

In the following application, Vectors 1 and 3 are written in the same way traditional vectors are written. A single VDN for each of these two vectors is used for all states. A single VDN for each state is assigned to Vector 2. The call queues to the first VDN skill preference at the beginning of vector 2. After allowing some time for the best-matched agents to answer, the call queues to the second VDN skill preference. Later the call queues to the third preference as shown in Figure 3-15.

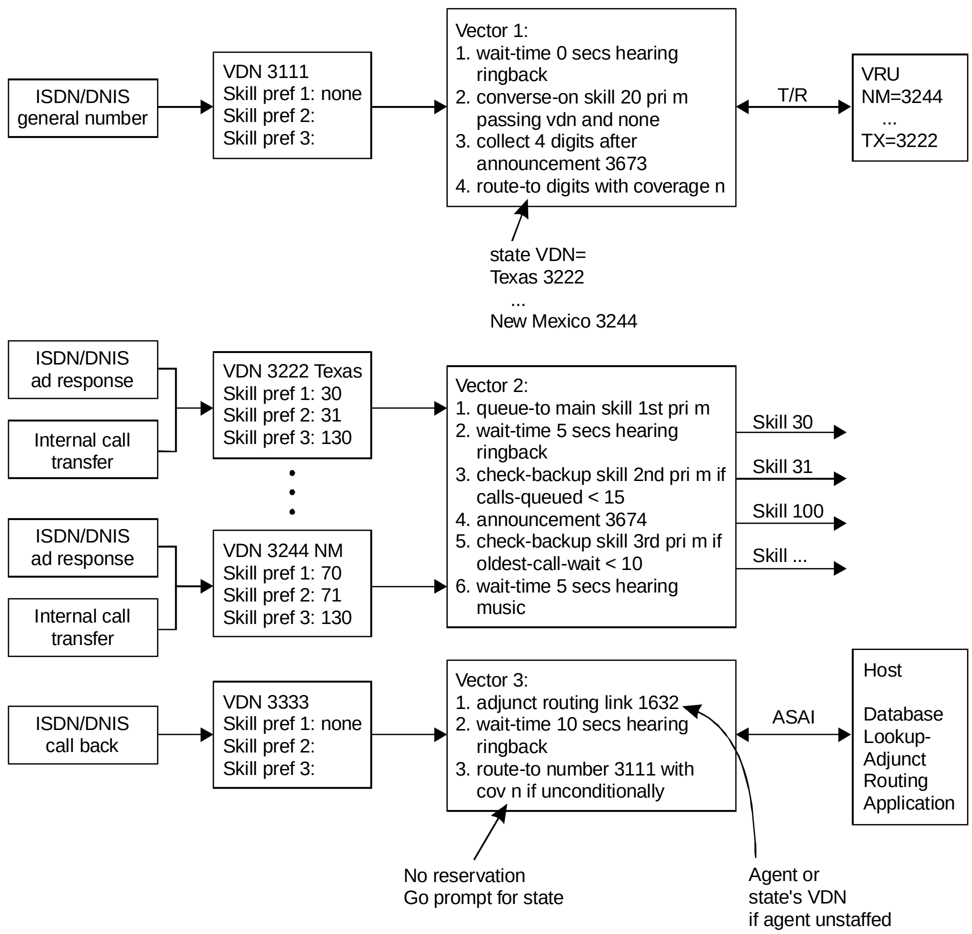


Figure 3-15. Writing Vectors Example

EAS Agent Operations

In an EAS environment, the DEFINITY Communications System treats agent login IDs as extension numbers. Therefore, the switch treats agents based on their individual login IDs. Agents are no longer associated with a particular voice terminal. Agents can use any voice terminal and multiple agents can use the same terminal on different shifts.

Agent login IDs are assigned up to four skills and each skill is assigned primary or secondary rank. With EAS, agents can be given preferences in the type of calls they handle. In the EAS application example, all agents have Skill 100 as a secondary skill. Agents with a Texas accent who have been to the Texas resorts and New Mexico resorts would have Skill 30 as a primary skill and Skills 71 and

100 as secondary skills. When no calls are waiting for Skill 30 for a particular agent, that agent receives a call for Skill 71 or 100.

Agents use a single set of agent work mode buttons for all their skills. Work mode buttons no longer have particular splits (or skills) assigned to them. This provides much greater flexibility in that any voice terminal with work mode buttons can be used by any agent. Also this requires fewer work mode buttons and simplifies the agent operation.

Physical aspects of the voice terminal, such as the set type and button layout, are associated with the voice terminal and not the Login ID. This allows agents to use any voice terminal with the buttons as are labeled on that set.

In addition to skills, the following capabilities are now associated with agents' login IDs.

- Calls - calls to the Login ID reach the agent independent of the voice terminal the agent is currently using. These can be Direct Agent calls (calls to a particular agent treated as an ACD call).
- Name - calls to the Login ID display the name associated with the Login ID and not the name associated with the voice terminal. This is also true for calls made from a voice terminal with an agent logged in.
- Coverage - when the agent is logged out, or when calls go to coverage because the agent is busy or does not answer, calls to the Login ID go to the coverage path associated with the agent and not the voice terminal.
- Restrictions - calls to the Login ID or from the agent use the restrictions associated with the agent and not the voice terminal.

Voice terminals are fully functional when an agent is not logged in. The restrictions, coverage, and name revert to the voice terminal administration when the agent logs out.

Other Login ID Applications

Anytime persons share a voice terminal, for example when working shifts or job sharing, each person can have their own extension and have their calls go to their coverage mailbox when they are logged out.

Also when persons use multiple phones, because they have multiple offices or rotate desks (such as guards), using login IDs allows these persons to be reached independent their desk location.

Administration of Skills

Skills are used in three distinct areas:

- Skills administered for a VDN
- Skills administered for an ACD Agent Login ID
- Skills active for an ACD Caller via vector commands

It is important to note that a particular Skill number always has the same meaning, whether it is called an Agent Skill, VDN Skill, or Caller Skill. For example, if Skill 127 means Spanish speaking, then Agent Skill 127, VDN Skill 127, and Caller Skill 127 all mean Spanish speaking.

Up to three different Skills can be administered to a VDN in a prioritized manner. The first or primary Skill administered to a VDN would be the Skill that is required or desired to service a call to that VDN. The second (secondary) and third (tertiary) Skills are optionally administered to a VDN and represent other Skills that are allowed to handle calls to that VDN.

Call Distribution

Using Automatic Call Distribution (ACD) queuing and the vector commands ‘‘queue-to main’’ and ‘‘check-backup,’’ the incoming call is matched to an agent who has at least one of the three Skills required to handle the call.

Skill hunt groups replace ACD splits when EAS is optioned on the Hunt-Group Form.

The hunt group form supports Expert Agent Distribution (EAD) in the ‘‘Type’’ field when EAS is optioned. When EAD is optioned, a new call to the hunt group results in a search for the most-idle primary agent. Only if no primary agents are available, the call goes to the most-idle secondary agent. This distributes a call to an agent best able to handle the call if multiple agents (primary and secondary) are available.

The Multiple Split Queuing feature is used to allow calls to queue up to three Skill hunt groups simultaneously. Agents can log in with up to four Skills at any one time. Each of the agent’s Skills are administered as a primary Skill or a secondary Skill. The ACD software distributes any call waiting for one of the agent’s primary Skills when the agent becomes available. If no calls waiting are found for a primary Skill, the queued calls for secondary Skills are distributed to the agent.

With Multiple Call Handling, available with G3V3 and later releases, an agent can be interrupted with an additional ACD call either after putting a call on hold, or when the agent is active on another ACD call. Seethe "Multiple Call Handling" feature for more information.

EAS Logical Agent

With Logical Agent, an agent's ACD Login ID is associated with a particular voice terminal only when the agent actually logs in at that terminal. When the agent logs off, the association of an agent's ACD Login ID with a particular voice terminal is removed. The ACD Login IDs used by the EAS capability are extension numbers used out of the switch's normal station numbering plan, and are administered using a new Agent Login ID form.

When the ACD Login ID is called, the call is routed to the associated voice terminal. With the proper class of restriction (COR), the call could be treated as a direct agent call. Otherwise, the call is treated as a normal (personal) call.

Agent Login IDs are administered via an ''Agent Login ID'' form. The form includes items such as the agent's name, COR, coverage path, AUDIX, security code, login password, ''Skills,'' and ISDN call display option. These attributes override the physical voice terminal attributes when the agent logs in.

When EAS is optioned, an EAS agent becomes associated with one or more Skill hunt groups upon login.

The Call Management System (CMS) is able to change an agent's Skill assignments. With G3V4 and later releases, one skill can be added, deleted or moved simultaneously for a group of up to 32 agents. See the "Move Agents From CMS" feature for more information.

It is important to note that Generic 3 can have either splits or Skill hunt groups to determine where to route the call (not both simultaneously). Non-ACD hunt groups can exist with either splits or Skill hunt groups. Skill hunt groups are required when using EAS.

When EAS is optioned, a single work mode button applies to all the Skills assigned to the agent currently logged in. Hunt groups are not assigned to work mode buttons, therefore, logged-in EAS Agents need only a single set of work-mode buttons for all ''Skills.''

An EAS agent administered with at least one measured Skill is automatically measured by CMS independent of whether any other Skills assigned to the agent are measured.

Direct Agent Calling

ACD calls can be transferred by one agent to another agent using the Direct Agent Call feature which is initiated by the dialing of an EAS agent's Login ID or Adjunct Routing route-to number vector steps. A Direct Agent call can be originated to a particular agent using Call Prompting.

Calls to an agent's Login ID are treated as Direct Agent calls if the originator and the receiver have the ''Direct Agent Calling'' class of restriction. Direct Agent calls can be originated by stations or trunks (with the proper COR). If the originator or receiver does not have the proper COR, the call is treated as a normal non-ACD (personal) call.

Expert Agent Selection

The EAS feature matches a caller whose call requires certain VDN Skills to an agent who has at least one of the required Agent Skills. Each Skill has an associated Skill hunt group administered similar to a vector-controlled hunt group. As mentioned earlier, Skill hunt groups replace splits when EAS is optioned.

Agents can have up to four Skills administered for each Login ID. Each agent Skill is designated as a primary or secondary Skill.

The ‘‘queue-to main skill’’ and ‘‘check-backup skill’’ call vectoring commands allow the specification of which VDN Skill (1st, 2nd, 3rd), or which specific Skill (1 through 255) a call is to be queued to. Having Skills assigned to VDNs allows the same vector to be used by multiple VDNs which require different Skills. Allowing a specific Skill gives the user the same vectoring capabilities that non-Skill vectoring provides. Caller Skills become active for an ACD call when a Queue-To-Main-Skill command is executed or a Check-Backup-Skill command is executed and the threshold condition is met. Once a Skill is active for an ACD Caller, the ACD Caller cannot be delivered to an available ACD agent unless the agent also has one of the active Caller Skills.

An ACD call can be queued to up to three different Skill hunt groups at one time. This is done by executing multiple Queue-To-Main-Skill/Check-Backup-Skill commands to different Skills. Subsequent “queue-to main skill” and “check-backup skill” commands to the same skill can be used to change the priority of the caller.

Since the EAS feature adds to the existing Call Vectoring command set, the full flexibility of Call Vectoring can be used to the advantage of the EAS feature. R3 CMS supports EAS with the changed vector administration, VDN administration, as well as the ability to incorporate Skills into real-time and historical reports.

Example EAS applications are described in *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide*, 555-230-520.

Publishing Login ID Extensions Instead of Physical Extensions

Telemarketing managers may wish to only publish the Login ID for agents in applications where the agents move from desk to desk (voice terminal to voice terminal). Users are able to call the agent without knowing at which physical voice terminal the agent resides. An optional password has been added to the login sequence to provide additional security.

Direct Agent Calling

Direct Agent Calling allows users to call agents and have the call treated as an ACD call (for example, zip tone answer, queuing, after call work, CMS measured as ACD). Using EAS capabilities, any call (which has the proper class of restrictions) to the Login ID is treated as a Direct Agent call.

Internal and external users can originate Direct Agent calls. An application for external users is the insurance industry. People can call their particular claims adjuster and have the call treated as an ACD call.

Class of Restriction

A byproduct of the EAS capabilities allows for applications which are not calls These applications could use the EAS capability of allowing a Class of Restriction to follow a user. Suppose you are in a manufacturing environment where assembly-line workers are not allowed to make outgoing calls. The voice terminals in the manufacturing area can be restricted from making outgoing calls. If a manager tours the manufacturing area and wants to make an outgoing call, the manager can ''log in'' using a feature access code and make an outgoing call (assuming the manager's login is not outward restricted). The same is true for the facility restriction level (FRL); the manager's FRL and not the extension's FRL would be used to place the call.

Considerations

When EAS is enabled, all agents must be administered as EAS agents via the Login ID administration form. Login ID administration and hunt group administration can be done prior to enabling the feature.

Agents become logical entities and not physical entities. However, physical attributes of a voice terminal, such as button administration and automatic answer, do NOT follow the agent.

The Skill hunt group in which a Direct Agent call is queued is determined by the first skill the agent has been logged into. CMS tracks the queue time for the Direct Agent call to this Skill, unless it is not measured. If the agent's first logged into Skill is not measured, CMS credits the agent Login ID with the Direct Agent call, but no queue time is tracked.

EAS agent Login IDs are also tracked for personal calls. The first Skill an EAS Agent is logged into (whether primary or secondary) is used by CMS for measurements on personal calls. If the first logged-into Skill is unmeasured, CMS credits the agent Login ID with the personal call, but no Skill hunt group is credited with the personal call.

Caller Interactions

The caller does not experience any new interface or interactions when Expert Agent Selection is enabled. The full power of Call Vectoring can be used with EAS to customize the treatment to the caller. The call is delivered to an agent with the proper Skill, so in that sense the caller gets better service.

Agent Login Procedure

The login procedure for EAS agent is very similar to the existing login procedure.

The agent goes off hook on an idle call appearance. The agent dials the login feature access code and second dial tone is given. The agent dials a 1 to 5 digit EAS agent Login ID (the number must be consistent with the switch dial plan). If the password length is 0 digits, the agent hears confirmation tone if the Login ID is valid. If the password is greater than 0 digits, then the agent is given dial tone again. The agent then dials a password of 1 through 9 digits optionally followed by ‘‘#.’’ If the Login ID and password entered are valid (administered on the ‘‘Agent Login ID’’ form), the agent hears confirmation tone. If the agent is not logged into any of the administered skills, the intercept tone is given. The Skills an agent has assigned are displayed on the second line (or first line on single-line display terminals) of the agent’s voice terminal display in the format “Skills: Tnnn, Tnnn, Tnnn, Tnnn,’’ where the Skill type ‘‘T” is displayed as ‘‘p’’ or ‘‘s,’’ and ‘‘nnn’’ (left justified, no leading zeros) is the Agent Skill number. This display is shown upon completion of logging in and remains for five seconds unless the agent goes on-hook or the display is overwritten.

If the Login ID entered is not valid, the agent hears the intercept tone. If another agent is already logged in at that physical terminal, the agent hears intercept tone. If the maximum number of agents are currently logged in, then the login is denied and the agent hears reorder (in other words, fast busy) tone. If the Login ID is already logged in, the login is denied and the agent hears intercept tone.

Logout Procedure

To log out, the agent goes off hook on an idle appearance and dials the logout feature access code. The agent hears the confirmation tone if the agent is logged in. If the agent is not logged in, the agent is given an intercept tone.

NOTE:
The agent does not login and logout for each Skill.

Calling an EAS Agent

Independent of the CORs for Direct Agent calling, to call an EAS Agent, the originator dials the Login ID extension. The call is extended to the physical extension associated with the Login ID. The originator can be internal to the switch or external. The originator hears the same tones that the originator would hear when calling a physical extension. The call goes to the Login ID's coverage path if a busy condition or ringing time out occurs as indicated in Table 3-57.

Table 3-57. Call Progress Through Called Login ID Coverage Path

Agent Status	Call Progress Through Called Login ID's Coverage Path	
	With Coverage	**No Coverage**
Agent logged in:		
Ringing time out	ring coverage	continue ringing
Busy (no appearances)	busy coverage	busy tone
Agent logged out	busy coverage	busy tone

After dialing the Login ID, the originator's display shows the name associated with the Logical Agent and not the physical extension's name where the agent is logged in. All station features for the originator and the agent should work as if the call had been made to the physical extension, except for messaging features such as leave word calling, which apply to the Login ID extension.

Making a Direct Agent Call to the Agent

The user makes a Direct Agent call by calling the agent's Login ID. If the originator's or agent's COR restricts the call from being a Direct Agent call, the call is treated as a personal (normal) call (for example, ring on an idle appearance). Otherwise, the call is treated as a Direct Agent call.

Single Set of Work Mode Buttons

EAS Agents need only a single set of work-mode buttons for multiple Skills. With the EAS feature, agents always are allowed to go to the AUX work-mode state for all logged-into Skills by pressing a single AUX work-mode button even though the agent may have a call waiting in-queue (or be the last logged-in agent) for a particular Skill.

Message Waiting Lamp

With G3V4 and later releases, the Message Waiting Lamp by default tracks the status of messages waiting for the logged in EAS agent LoginID rather than messages for the physical terminal. The operation of the Message Waiting Lamp can be changed so that it tracks the status of messages waiting for the physical terminal where the agent is logged in. See the Feature-Related System-Parameters form in *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for more information.

Adjunct Interactions

ASAI

ASAI support for EAS may be broken into the following categories: call control, feature requests, value queries, event notification, and adjunct-controlled splits. This section provides a high level overview of the behavior of ASAI in the G3V2 EAS environment.

Call Control

Call control capabilities work exactly the same in the EAS environment as in the traditional ACD environment except for the following:

1. User classified third party make calls (calls classified by originator), may originate from an EAS Login ID, and terminate to a Login ID. User classified calls terminating to a Login ID are given Direct Agent treatment just as they would be if dialed from a station extension.
2. Switch classified third party make calls (calls classified by a call classifier board; answered calls are delivered to originating hunt group), may originate from or terminate to EAS Login IDs.
3. Direct Agent third party make calls (ACD call terminated to a selected member of an ACD split), may be requested by including a Direct Agent option, an agent’s physical extension and a split extension (compatibility mode), or by requesting a user classified third party make call with a Login ID destination. The primary differences between the two methods of requesting Direct Agent calls are that the compatibility mode allows the adjunct to specify the split that a given Direct Agent call is queued to, and the non-compatibility mode allows the adjunct to direct the call to a Login ID, regardless of which station an agent is logged into. Direct Agent calls may not originate from an EAS Login ID.
4. Supervisor assist third party make calls (supervisor assist calls originated by a selected member of an ACD split), may originate from a EAS Login ID and may terminate to an EAS Login ID. Unlike dialed Direct Agent calls, supervisor assist calls terminated to a Login ID behave as though they had been directed to the requested Login ID’s physical extension, for example, they do not cover if the requested agent is not logged, the originator’s display shows the agent’s physical extension and not the agent’s Login ID.
5. Extension (Domain) control may not be requested for an EAS Login ID but may be requested on behalf of a logical agent’s physical extension. Auto-dial calls (calls initiated by an extension controlled station) may be terminated to an EAS Login ID, in which case the call is given Direct Agent treatment.
6. Adjunct routing calls (vector calls routed by an ASAI adjunct via the adjunct routing vector command) are similar to third party make calls; they may include a Direct Agent option, an ACD agent’s physical extension,

and a split extension, in which case they are given compatibility mode Direct Agent treatment, or they may be terminated to an EAS Login ID (in which case they behave like dialed Direct Agent calls).

Feature Requests

In the EAS environment, agent login, logout and change work-mode requests are fully supported. Agent login requests must contain an EAS Login ID and optional password (delimited by a '#') in the login request's user code IE. Agent logout requests and change work-mode requests may contain the desired agent's physical extension or Login ID. Call forwarding and send all calls feature requests are denied for EAS Login IDs but may be requested for EAS's physical extensions where an EAS agent is logged in.

Value Queries

Value queries function identically in the EAS and traditional environments, except that the Extension Type/Class Information Query returns a new indication that a requested extension is an EAS Login ID along with an indication of whether the Login ID is currently logged in and where (in other words, at which physical extension).

Event Notification

Because all Skill hunt groups are vector controlled, event notification may not be requested on the basis of a Skill hunt group extension. Event notification may, however, be requested on the basis of a controlling VDN extension. Generally, all event reports involving EAS agents contain the agent's physical extension rather than the agent's Login ID.

Adjunct-Controlled Splits

Agents with adjunct controlled skills are considered to be adjunct-controlled agents. Adjunct-controlled agents exhibit the same behavior as agents in adjunct-controlled splits in the traditional ACD environment:

- Stations are locked for all logged-in adjunct-controlled agents. The only action an agent can take from the station is to go onhook (or unplug the headset) from an auto-answer station which causes the agent to be logged out.
- Stations are unlocked whenever the controlling adjunct's ASAI link goes down. Stations are locked again when the adjunct's link is reestablished.
- The adjunct controls all split/agent activities such as login, logout and change work-mode (with the exception of agent logout via onhook).
- Only adjunct-controlled calls can terminate to the extension of an adjunct-controlled agent.
- Only adjunct-controlled calls can terminate to an adjunct-controlled split/Skill extension.

- Agents may not log into more than one adjunct-controlled split/Skill.
- Agents logged into an adjunct-controlled split/Skill may not log into a non-adjunct-controlled split and vice versa.

Furthermore, adjunct-controlled EAS Agents can only be administered with one Skill. This implies that EAS agents may not mix adjunct-controlled and non-adjunct-controlled Skills.

AUDIX

Calls to the EAS agent Login ID can cover to AUDIX. Each agent must enter his/her Login ID when calling AUDIX to obtain messages.

AUDIX agents are assigned to EAS agent extensions. These Login IDs are used for CMS tracking if the associated AUDIX Skill hunt group is externally measured. The ‘‘aut-msg-wt’’ button can be used to indicate that Login ID has a message.

Speech Processing Adjuncts

Speech processing adjuncts which have ‘‘line’’ interface to G3V2 are able to initiate Direct Agent calls by dialing the Login ID for an agent.

Feature Interactions

EAS interaction with vector-controlled splits:

Unless otherwise specified, the feature interactions for Skill hunt groups are the same as for vector-controlled splits.

- Abbreviated Dialing

  Abbreviated dialing can be used to log in/out EAS agents. Abbreviated dialing lists/buttons can only be administered for stations.

- Add/Remove Skills

  In the EAS environment, agents have the ability to add and remove skills during a login session by dialing a FAC. Other voice terminal users with console permissions can add or remove an agent’s skill on behalf of the agent. (Note that the ability to add and remove skills depends on whether a user has a class of restriction (COR) that allows adding and removing skills.)

- Administration Without Hardware

  EAS Login ID extensions are extensions without hardware. Login ID extensions require space in the dial plan. If a coverage path is administered to a Login ID, calls to a Login ID are immediately directed to that path when the agent is not logged in. When a Login ID is logged in and a call is made to the Login ID extension, and the call is routed to the Login ID coverage path if coverage is administered for the Login ID.

- Agents in Multiple Splits Feature

  When the EAS feature is enabled, the ''Agents in Multiple Splits'' feature is referred to as ''Agents in Multiple Skills,'' which allows an EAS agent to be logged into up to four Skills.

- Agent Work Mode States

  With EAS, agents can only be in a single work state mode at any one time.

- Assist

  The Assist feature can be used for a Skill hunt group (for example, where there is one supervisor per Skill hunt group). A voice terminal can be administered with one or more Assist buttons for each Skill that agents using the terminal might have. The administered association of an Assist button with a particular split or Skill is not affected when an EAS agent logs into that station.

- Automatic Answering with Zip Tone

  The Automatic Answer option can only be administered for a physical extension.

- Auto-Available Splits

  If a Skill hunt group is administered as an Auto-Available Skill (AAS) the EAS Login IDs assigned to this Skill must also be administered as Auto-Available. When the switch reinitializes, these Login IDs are automatically logged in with the AUTO-IN work-mode. If any switch features attempt to change the work-mode to anything except to AUTO-IN, this attempt is denied. This feature is not intended for human agents.

- BCMS

  The Basic CMS user interface remains the same when EAS is enabled. Basic CMS is not modified when EAS is optioned, except to change the labeling of headings from ''split'' to "Skill.''

- Call Forwarding

  Skill hunt groups (since they are vector-controlled) cannot be call forwarded. EAS agent Login IDs can not be forwarded, but the physical extension where the EAS agent is logged in can be forwarded. If another station (with console permissions) tries to forward an EAS Login ID, an intercept is made.

- Call Management System (CMS) Interface

  When an EAS Login ID is administered as measured, the BCD format of that Login ID becomes the internal agent number (called ''position'') on CMS. During pump-up to CMS, the switch sends a list of the logged in agents and their internal agent numbers (position).

- Call Park

  Calls cannot be parked on the Skill hunt group extension.

- Call Pickup

  Skill hunt group extensions and EAS Login ID extensions cannot be members of a call pickup group.

- Call Vectoring

  The ‘‘goto vector’’ command carries the current VDN’s VDN Skills to the new vector. ‘‘route-to’’ vector which routes a call to another VDN uses the VDN Skills of the new VDN if VDN override is set on the current VDN. If VDN override is not set, the ‘‘original’’ Skills are used after a ‘‘Route to Vector’’ step to a VDN is executed.

  “Route-to’’ vector command with an EAS Login ID as the route to destination is treated as a Direct Agent call if the VDN and agent have the COR and the ‘‘Direct Agent’’ field is set to ‘‘y’’.

  The same vector can be used for many VDNs since the vector specifies the algorithm for call delivery, and the VDNs supply the Skills to use when selecting an agent.

  In the same way as for vector-controlled splits, a “queue-to main” “check-backup vector” command step associated with a skill hunt group is skipped and the next step is processed if any of the following conditions are true (where the term “Skill’’ applies to both a VDN Skill and an Agent Skill):

  — The Skill hunt group queue associated with the desired “Skill’’ is full

  — The Skill hunt group associated with the desired “Skill’’ is not vector-controlled

  — The call is already queued to the Skill hunt group at the specified priority level

  — The call has already queued to three different Skill hunt groups

  — The Skill hunt group associated with a desired ‘‘Skill’’ has no queue administered for it, and no agents are available in that Skill hunt group

  — A certain VDN Skill level (for example, 1st, 2nd, 3rd) is specified in the vector command, and no VDN Skill is administered for that level

- Class of Restriction

  Skill hunt groups do have a class of restriction. This is used if the Skill hunt group extension is called directly.

  The COR for an EAS agent Login ID overrides the physical extension’s COR of the voice terminal an EAS agent logs into.

- Class of Service

  EAS agents do not have a COS associated with their Login ID. Therefore, the COS of the voice terminal is not affected when an EAS agent logs into that terminal.

- Direct Agent Calling

  If a called EAS Agent Login ID and the call originator (extension or trunk) both have a COR that allows Direct Agent calls, the call to the Login ID is treated as a Direct Agent call. Calls to the voice terminal extension where an EAS agent is logged in are treated as personal (non-ACD) calls.

- Displays - Voice Terminal

  When an EAS agent logs in, the display for originators who call the Login ID shows the Login ID and agent name (as administered via the Agent Login ID form). Calls that the agent originates show the Agent Login ID and agent name at the receiving terminal's display. However, with G3V4 and later releases, the user can display the name of the physical terminal where the EAS agent is logged in. The user must be active on a call with the agent, and must have a terminal with an alphanumeric display and an inspect button. When the inspect button is pressed during a call to or from the EAS agent, the physical terminal name of the agent is displayed.

  Calls to the physical extension show the physical extension's number and name on the originator's display.

- Look Ahead Interflow

  Skills are not sent to another ACD/PBX when a call interflows using Look Ahead Interflow. If Skills have the same meaning on both ACDs, then a Look Ahead Interflow command to a VDN with the same Skills assigned can provide a mapping of the Skills.

- Multiple Split Queuing

  When EAS is enabled, the Multiple Split Queuing feature is referred to Multiple Skill Queuing, which has the same functionality.

- Queue Status Indications

  Physical extensions can be administered with Queue Status Indicator buttons and lamps for Skill hunt groups which operate in the same manner as split Queue Status Indicators for traditional ACD splits. Queue Status Indicators can be administered for all Skills needed by agents using that physical extension, given that enough buttons are available.

- Service Observing

  The Service Observing feature is activated in the EAS environment by dialing the physical extension of the terminal where a EAS agent is logged in, not the Login ID of an agent.

- VDN Override

  If VDN override is set on the ''original'' VDN, then the VDN Skills of the new VDN are used. If VDN override is not set, then the VDN Skills of the ''original'' VDN are used.

Administration

EAS is an optional feature. EAS is also a feature-related parameter that must be enabled. Once EAS is optioned, most of the EAS related administration can be done prior to enabling the feature. EAS requires that “Call Vectoring” and “ACD System Parameter/Customer-Options” fields be set to ‘‘y.’’

System Parameters

A new system parameter indicates the minimum number of digits required for the agents’ password. Valid entry are 0 through 9. Also, the system parameter for ‘‘Automatic Call Distribution (ACD) Log-in Identification Length’’ is removed when EAS is enabled. These parameters appear on Page 4 of the Feature-Related Parameters form.

Logical Agent

EAS agent Login IDs are administered as part of the station numbering plan and are administered using the Agent Login ID form as extensions without hardware.

In addition, the following is administered for EAS

- Login ID
- Name
- COR
- Coverage path
- Security code (for AP demand printer)
- Auto-avail split and AUDIX information
- Password (this field is not visible)
- ISDN Caller Display
- Up to four Agent Skills

If the maximum number of agents are already administered as internally measured, or the maximum number of agents are already administered, then an error is given. If the extension corresponding to the Login ID has already been assigned, an error is displayed.

If the password is fewer digits than the system parameter or contains non-numeric characters, then an error message is displayed.

Station Administration

On the Station form with the EAS featured optioned, when a work-mode button is selected, no ‘‘Gp’’ information can be entered. The assist and queue status buttons require that ‘‘Group’’ be entered.

VDN Administration

The VDN Skill preferences for handling a particular call type is administered on a per-VDN basis. Up to three VDN Skills can be administered for each VDN. The first, second, and third VDN Skills can be considered as the primary, secondary, and tertiary Skills respectively required for handling the call. All of the VDN Skills on the VDN form are optional. For example, only the first and third, or only the second and third VDN Skills might be assigned. Also, for example, if a vector encounters ‘‘Queue-To-Main-Skill 2nd’’ command, and no second VDN Skill is administered to the VDN, the step is skipped.

Vector Administration

Vector commands “queue-to main split” and “check-backup split” are changed to Queue-to-Main-Skill and Check-Backup-Skill when the EAS option is set to ‘‘y’’. These two queuing vector commands allow queuing the caller using the VDN Skills by administering ‘‘1st,’’‘‘2nd,’’ or ‘‘3rd’’ in the “Skill Number” field. These two queuing vector commands also allow a specific Skill number (1 through 255) to be entered in the “Skill Number” field. Also, the “check-backup skill” step supports all of the existing conditional checking (for example, available agents, staffed agents, oldest call, number of queued calls) for a Skill Hunt Group. Also the conditional checks for ‘‘goto’’ and the ‘‘messaging’’ command use the term “Skill’’ instead of split and allow entering values first, second, or third as well as the specific Skill number.

Hardware and Software Requirements

No new or changed hardware is required for the "Expert Agent Selection (EAS)" feature. This feature requires the appropriate call center package, but places no limitations on the type of voice terminals that can be used.

Extended Trunk Access

Feature Availability

This feature is available with Generic 3rV1 and all Generic V2 and later releases.

Description

Extended Trunk Access (ETA) is a software feature that provides a mechanism for routing calls that are not defined either in the first or second digit tables or the feature/trunk access code tables. This feature makes use of an ETA routing pattern and/or an ETA node number for determining how to route an unidentified call. Using ETA allows the user to fully use the capabilities inherent in the Automatic Alternate Routing (AAR) and Uniform Dial Plan (UDP) features.

Historically, ETA has been used by satellite switches to access stations, trunks, and features at the main switch. ETA frees the satellite switch administrator from having to enumerate the entire dial plan for the main/satellite complex.

Considerations

The user of ETA is usually unaware that it is being used because the feature is transparent to end users. To use ETA, a user dials a sequence of digits that are unadministered in the station, trunk, or feature access code dial plans, but may be successfully directed to an extension, trunk, or feature on the remote switch. This results in the local switch sending the digits to the remote switch that provides the appropriate audible signals a user would expect, such as the ringing of the calling party's voice terminal.

Capacity Requirements and Constraints

The purpose of the ETA Call-Screening Table is to allow a set of digit strings to be identified that should not be routed via ETA. For example, if the system administrator notices that a digit string was misdialed repeatedly and as a result was routed via ETA to a remote switch that returned intercept treatment, then that digit string would be a candidate for entry into the ETA call-screening table. Entry into this table would have the effect of returning intercept treatment to the caller without first attempting to route the digits to the remote switch for interpretation via ETA.

The maximum length of the ETA Call-Screening table is 10, each with a maximum length of seven digits with values ranging from one to 640 and/or special wildcard characters. The ETA routing pattern is a three-digit string. There is only one ETA routing pattern per switch. The ETA node number is a three-digit string with values ranging from one to 999. There is only one ETA node number per switch. The impact of the ETA feature on the performance of the switch as a whole is negligible because the extra processing needed by the feature is negligible.

The difference between normal call routing and ETA call routing is that extra processing is necessary at the point where the intercept treatment would have normally been applied during the normal call sequence. Specifically, calls that would have previously been given intercept treatment will now be routed to a remote switch across a trunk to be reprocessed at the remote switch if either an ETA routing pattern or an ETA node number is defined. This means that the following extra processing is required for each ETA call when ETA is administered:

- Determination that this is an ETA call because it fails all other possible routing possibilities and ETA is administered.
- The digit string dialed is not in the call-screening table.
- An available pattern preference is selected from the trunk group specified by the ETA routing pattern and/or ETA node number.
- The dialed digits are sent to the remote switch.

Interactions

The following features have interactions with ETA. Descriptions of features as they interact with ETA are provided.

- Abbreviated Dialing

  Any digit string that is undefined in the station or feature/trunk access-code dial plans should route via ETA.

- Attendant

  Attendants can access the ETA feature in exactly the same way as stations. There is no discernible difference in the ETA feature operation for attendants.

- Automatic Alternate Routing

  The ETA feature is implemented as part of Uniform Dial Plan (UDP). The routing preferences defined for the administered ETA routing pattern and/or ETA node number are individually checked to determine which trunk group will be used to route the dialed digit string.

- Data-Call Setup

  Analog and digital endpoints can access ETA. The digit string to be sent goes to the remote switch just as any other digit string is sent. The remote switch will handle the data-call setup from that point forward.

- DCS

  Applies only to ETA calls to an unknown extension (not FAC or TAC calls).

- Dial Plan Administration

  When administering ETA on separate switches, care must be taken to not create a circular ETA call setup; otherwise, all trunk members using ETA will be tied up until the calling party releases the call. If switch A routes calls to switch B and switch B has been administered to route calls to switch A, then any time an invalid set of digits is dialed, all trunk members using ETA will be busied in an attempt to route the ETA call. Since both switches point to each other, an infinite loop has been administered. Care should be taken not to have a receiving switch administered with ETA when the destination switch is also the sending switch.

- Dial Plan

  Any dialed digit string that is undefined in the station feature/trunk access-code dial plans will be routed via ETA to the remote switch, if ETA is administered. This implies that a feature access code is dialed, followed by some number of digits. For example, if you were dialing an AAR/ARS access code followed by digits and the dialed string code could not be routed by the local switch, then the entire digit string would be passed to the remote switch by the ETA feature. It is also important to note that undefined dialed digit-strings will not be routed via ETA unless the dial plan of the local switch recognizes the digits as they are being dialed during its own screening process as a valid digit-string type.

- Direct Inward Dialing (DID)

  Whenever a DID trunk presents undefined digit strings to the local switch and ETA is administered, the digits will be passed to the remote switch via ETA for proper handling.

- Facility Restriction Levels

  It is possible to restrict trunks that are being used in conjunction with the ETA feature by assigning restriction levels.

- Last Number Dialed (LND)

  If a number is routed via ETA to a remote switch and you want to reaccess that number, then the reaccess uses ETA and does so transparently.

- Main/Satellite/Tributary

  The environment of local and remote switches is the primary target for the ETA feature. It is within this environment that ETA is most effective in reducing maintenance of frequent administrative changes.

- Modem Pooling

  All trunks in the ETA pattern will be treated like all other trunks. Modem Pooling is not affected.

- Remote Access

  Remote-access trunks are able to access the ETA feature just as any other trunk or station does.

- Uniform Dial Plan

  The Uniform Dial Plan is extended by the ETA feature in that explicit definition of digit strings to be routed is not necessary. The internal operation of the UDP feature is not affected, however.

Administration

The switch administrator is able to add, delete, and display the ETA routing pattern and/or ETA node number as well as the Call-Screening table. The ETA routing pattern and/or ETA node number appears on the first-digit table.

The “ETA Routing Pattern” and/or “ETA Node Number” field accepts numerical values that are within the proper range and provides appropriate help and error messages as does the ETA node number. Likewise, the ETA call-screening table occupies the screen that allows up to10-digit strings to be enumerated. Any valid digit string can be entered including special wildcard characters ‘‘*,’’ ‘‘#,’’ and ‘‘x.’’ Appropriate help and error messages are provided.

If ETA routing pattern is administered, then there are two other screens that must be administered to make the feature work properly. Specifically, the route pattern for ETA must be administered along with the trunk group that will support ETA calls. The route-pattern screen is entered from the command line with ‘‘change route pattern xxx’’ where ‘‘xxx’’ is the ETA routing pattern.

Within this screen a trunk group is associated with the routing pattern, an FRL, an NPA, a Prefix Mark, a Toll List, and either absorption or deletion of digits if desired.

Trunk groups are administered with the ‘‘add/change trunk group xxx,’’ where xxx is the trunk group number. There is no special trunk group administration necessary for ETA. For additional information on the administration and operation of the AAR feature, see the "Automatic Alternate Routing (AAR)" feature description in this document. For G3r if ETA node number is administered, then node number routing must be administered along with a trunk group.

There are four scenarios that can be envisioned with respect to the above two fields either being administered or not administered. The following enumerates these possibilities and indicates what effect the administration will have on the way the feature either does or does not work.

CASE #1

- ETA Route Pattern — not administered
- ETA Node Number — not administered

In this case, the ETA feature is considered not active and no attempt will be made to utilize the ETA code to route otherwise undefined calls.

CASE #2

- ETA Route Pattern — administered
- ETA Node Number — not administered

In this case, the ETA route pattern will be used to determine how to route an otherwise undefined digit string. Since, however, an ETA node number is not specified, non-call associated DCS messages have no chance of succeeding in this scenario.

CASE #3

- ETA Route Pattern — not administered
- ETA Node Number — administered

In this case, the ETA node number will be used to get the routing pattern from the node number routing tables. DCS messages that are non-call related will also have a chance of succeeding since a node number will be supplied to the DCS message handling code.

CASE #4

- ETA Route Pattern — administered
- ETA Node Number — administered

In this case, the ETA route pattern will be used to route all undefined dialed digit strings while DCS messages will use the ETA node number to attempt to route. Nodes themselves do not have to be administered for ETA. ETA should not be used over Tandem Tie trunks.

Hardware/Software Requirements

There is no specialized hardware needed for ETA. However, trunks need to be administered for ETA. The feature usually uses tie trunks.

Extension Number Portability

Feature Availability

This feature is available with Generic 3rV1 and all Generic V2 and later releases.

Description

ENP provides the ability to assign any extension to any switch in an ENP subnetwork. The ENP Numbering Plan defines the setting of 4- or 5-digit extensions in the ENP subnetwork to a 7-digit (Automatic Alternate Routing) AAR-like number for sending to other nodes in that network. Only the first one or two leading digits of the extensions are significant in this action. This limits the number of ENP codes to 100. Each of these ENP codes should be administered in the AAR analysis table as home on all the nodes within the ENP subnetwork. Though up to 100 3-digit ENP codes can be used, only one ENP code is required for a 4-digit ENP, and a 5-digit ENP requires only one ENP code for each leading digit of extensions used within the subnetwork.

Considerations

This feature is compatible with System 85 and G2 ENP.

Interactions

The Uniform Dial Plan administration and that of Automatic Alternate Routing Analysis are affected. See “Administration” below. If DCS is used, the ENP node numbers must correspond to DCS node numbers.

Administration

Administering ENP for a group of extensions is accomplished in G3rV1, G3V2, and later releases using four administration forms. The UDP table must be administered so that Extension Codes for non-home extensions in the ENP subnetwork are node-number routed (one pattern per PGN).

The Node-Number Routing table must be administered to associate a routing pattern with each node in the ENP subnetwork.

The AAR Digit conversion table must be administered to assign all 3-digit ENP codes as home, and if using a 5-digit UDP, to associate the ENP codes with a leading 10 thousands digit. Note that ENP Codes are distinguished from AAR location codes because ENP Codes are home on every node within the ENP subnetwork. ENP Codes are administered in the ENP Numbering Plan table as well as in the AAR Digit conversion table. Since ENP Codes are home on every node, they cannot be used as AAR location codes.

The ENP Numbering Plan table must be administered to associate the leading one or two digits of extensions in the ENP subnetwork with a 3-digit ENP code, used to construct a 7-digit AAR-like ENP number.

Administering the ENP subnetwork in this manner enables stations to move from one node to another within the ENP subnetwork without having to change their extensions.

ENP Numbering Plan administration depends on both the UDP customer option and on the “Dial Plan Administration” field that asks if the administrator wishes to use UDP. If both of these fields are not administered to ’y’, then ENP Numbering Plan administration is not allowed. If the UDP customer option is not administered to ’y’, then ‘‘enp-number-plan’’ is not even listed as an option to the administrator; if UDP is administered to ’y’, then the option is available, but the command is denied if UDP is not also administered to ’y’ on the Dial Plan form. The UDP plan length is also administered on the Dial Plan administration form.

All ENP codes specified in the ENP Numbering Plan should be administered as home in AAR Analysis administration, but there is no automatic cross checking to verify that ENP codes have been administered correctly in AAR Analysis.

Hardware/Software Requirements

ENP Numbering Plan does not require any additional hardware or software. However, a UDP software license must be acquired.

Facility Busy Indication

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides multi-appearance voice terminal users with a visual indication of the busy or idle status of an extension number, a trunk group, terminating extension group, a hunt group (DDC or UCD group), or any loudspeaker paging zone, including all zones. The Facility Busy Indication button provides the voice terminal user direct access to the extension number, trunk group, or paging zone.

When the lamp associated with the Facility Busy Indication button is lighted, the tracked resource is busy. If the lamp is dark, the resource is idle. If the lamp is flashing, the tracked resource is placing a call to the voice terminal with the button.

Pressing the Facility Busy Indication button automatically selects an idle call appearance and places a call to the resource.

Considerations

With Facility Busy Indication, a user can monitor the busy or idle status of a frequently called extension number. By knowing when the monitored facility is busy or idle, the user can wait until the facility is idle to make a call. This reduces the time spent trying to call busy facilities.

Facility Busy Indication buttons can be administered to track the same resource. The maximum parameters for the DEFINITY Communications System Generic 1 and Generic 3 hardware and software items can be found in the System Hardware and Software Capacity Limits tables located throughout the DEFINITY documentation set. A new state of the tracked resource (a change from idle to busy) is updated within five seconds after the system detects the change.

Extension numbers, trunk group access codes, and Loudspeaker Paging Access codes can be stored in a Facility Busy Indication button. However, an access code followed by other numbers cannot be stored.

It is possible that an incoming call which causes the lamp to flash may go unanswered. If the lamp represents the status of a trunk group and all trunks in the trunk group become busy before the flashing call is answered, the system software lights the lamp steadily to indicate that all trunks are busy. When a trunk in that trunk group becomes idle, the system software turns off the busy indication and the lamp goes dark. Therefore, the lamp flashes, lights steadily, and goes out while the call has neither been answered nor dropped.

The Facility Busy Indication cannot monitor the status of the attendant console.

Interactions

None.

Administration

Facility Busy Indication is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign the Facility Busy Indication button to a voice terminal or attendant console.

Hardware and Software Requirements

No additional hardware or software is required.

Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs)

Feature Availability

Facility Restriction Levels (FRLs) are available with all Generic 3 releases; Traveling Class Marks is an optional feature available with all Generic 3 releases (except G3vs/G3s ABP) when Private Network Access (PNA) software is purchased. Traveling Class Marks are not available with G3vs/G3s ABP.

Description

Provides up to eight levels of restriction for users of the AAR and/or ARS features.

FRLs and TCMs provide a method of allowing certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed to use Central Office trunks to other corporate locations while other users may be restricted to the less expensive private network lines.

FRLs and TCMs are transparent to the user. Appropriate values are predetermined and programmed into the system. Dialing procedures are unaffected.

Call routing for each call is determined by the dialed Area Code and/or office code (either public or private network) or by the administered dial string. Translation on the first three or six digits of the called number yields Routing Patterns. More than one translation can point to the same pattern. A blank entry provides intercept treatment and is used for unassigned private network office codes. Each Routing Pattern contains up to six routing preferences (16 for G3r). Each preference includes the following information:

- Trunk Group Number
- Minimum FRL required to access the trunk group

No specific routing order is required.

Each facility, such as a trunk or voice terminal, capable of originating a call also has an associated FRL. Whether a given call is allowed or not depends on two things: compatibility between FRLs and availability of an idle trunk.

Compatibility is determined by a comparison of the minimum FRL associated with the trunk group and the originating-side FRL. Either can have a value of zero through seven. Access to the associated trunk group is permitted if the originating-side FRL is greater than or equal to the minimum FRL. Note that lower originating-side FRLs can access fewer routing preferences, whereas lower minimum FRLs permit greater access. Stated another way, a 0 originating-side FRL is the most restricted and a 7 is the least restricted. A 0 minimum FRL is the least restrictive, and a 7 is the most restrictive (when applied to the routing pattern's FRL). Compatibility checking begins with the first-choice route (the first one in the pattern). Assuming access is permitted, availability is checked; that is, is there an idle trunk in the group? If so, the call continues. If not, compatibility is checked on the next routing preference.

If the compatibility check fails on the first-choice route, the intercept tone continues.

NOTE:
Intercept is not received unless all possible routes are inaccessible.

If the compatibility check fails on the second or subsequent routing preference, or if all accessible trunk groups are busy, the call may queue on the first choice trunk group or first compatible trunk group. (See the "Ringback Queuing" feature for details.)

If the trunk group selected for a call is an intertandem tie trunk group, then a TCM is outpulsed as the last digit. A TCM is equivalent to the originating-side FRL. At the next tandem switch, compatibility and availability checking are done, as before. In this case, the FRL assigned to the incoming intertandem tie trunk group is used as the originating-side FRL (to compare with the TCM). If it is sufficient, then the call continues and no comparison on the TCM and the outgoing trunks TCM is made. If it is not sufficient, then the TCM is compared with the outgoing trunk's FRL to see if the call is allowed to continue. However, if this fails to yield a route and if the TCM is higher than the tie trunk FRL, then the TCM is used in another attempt to complete the call.

Call Originating Facilities

At a switch serving as the call origination point, any of the following can be the originator of an ARS or AAR call:

- Voice terminal
- Remote Access user
- Attendant
- Incoming tie trunk group from a subtending location
- Data terminal capable of Keyboard Dialing

At a tandem switch, either of the following can be the originator of an ARS or AAR call:

- Incoming Intertandem tie trunk group
- Incoming Access tie trunk group — links a remote main switch to a tandem switch

Each of these facilities is assigned an FRL via an associated COR, either directly or indirectly.

Voice terminals and all incoming tie trunk groups use the FRL contained within the assigned COR. Attendants use the FRL contained within the COR assigned to the attendant group for extended calls. If Individual Attendant Access is assigned, the individual attendant's COR FRL is used. Data terminals use the FRL contained within the COR assigned to the associated data module.

The Remote Access feature can be accessed via a DID trunk group, tie trunk group, dedicated central office trunk group, 800 Service trunk group, and/or dedicated foreign exchange trunk group. In the absence of a Remote Access Barrier Code, the applicable FRL is contained in the COR assigned to the trunk group. If a Barrier Code is required on Remote Access calls, the applicable FRL is the Remote Access default entry (none) contained in the COR assigned to the Barrier Code.

Call Terminating Facilities

Any of the following trunk types can serve as the termination point for an ARS or AAR call:

- Tie trunk — excluding RLTs, but including CCSA and EPSCS Access trunks
- WATS
- CO
- FX
- ISDN-PRI

Each of these outgoing trunk groups has an assigned COR that contains an FRL. However, this FRL is never used. Terminating-side FRLs are assigned in the Routing Pattern, not to the outgoing trunk group.

Considerations

FRLs provide the means to restrict certain users from placing selected calls while allowing other users to place the same calls.

Originating-side FRLs are assigned via the COR of the originating-side facility, such as an incoming tie trunk group or voice terminal. If an FRL is not assigned, the system assumes an FRL of 0 for all originating facilities except the attendant group. An FRL of 7 is assumed for the attendant group.

A COR is also assigned to each trunk group. If the COR specifies an FRL, the FRL is ignored. The minimum FRL specified in the Routing Pattern is the only FRL used on the terminating side of the call.

On attendant-extended calls, the attendant group FRL is used rather than the FRL of the calling party.

Interactions

- ARS and AAR

  FRLs apply only on ARS and AAR calls (including UDP).

- Authorization Codes

  Authorizations Codes can be used to raise a user's FRL.

- CDR Account Codes

  If CDR 15-digit account codes are used, the "FRL" field in the CDR record is overwritten.

- Intercept Treatment

  The TCM used to pass on the originating facility's FRL is sent by ISDN facilities in the SETUP message.

Administration

FRLs are assigned by the System Manager as a part of ARS and/or AAR administration. Originating FRLs are assigned on a per-COR basis. Terminating FRLs are assigned on a per-Routing Pattern basis. TCMs do not require assignment.

Assignment Guidelines

If there will be users within the system who are not allowed to make outside calls, use some value other than 0 as the value for the first-choice trunk group. By assigning these users an FRL of 0, none of the trunk groups can be accessed (since all trunk group FRLs are greater than 0). Such calls are denied.

Each Routing Pattern must be individually constructed. The same trunk group can be used in more than one pattern. The associated FRL is assigned within the pattern and is not associated with the trunk group itself. The same trunk group can have a different FRL in a different pattern.

Be consistent in FRL assignments. Do not use a range of 0 through 5 in one pattern and a range of 2 through 7 in another pattern if all users can access the first-choice route. Admittedly, the trunk group with an FRL of 2 may be more expensive than the trunk group with an FRL of 0, but there is no real reason to assign a 2 to a trunk group that everyone can access. For ease of assignments, always use a 0 for such a trunk group.

There should be a COR established for each FRL used in a Routing Pattern. The appropriate COR is then assigned to the users who can access the routes restricted by the FRL value. For example, a middle executive might be able to access all routes with an FRL of 5 or lower, whereas the president can access all routes. In this case, the executive is assigned a COR with an FRL of 5 and the president is assigned a COR with an FRL of 7.

Remote Access users can access the system's features and services the same as an on-premises user. FRL assignment is via Remote Access Barrier Codes. Up to 10 Barrier Codes, each with its own COR (and FRL) can be assigned. Although the COR defines other restrictions, 10 Barrier Codes are enough to also provide a range of FRL assignments. Assignment of Barrier Code FRLs is the same as if the user were on-premises. The simplest way to assign these FRLs is to duplicate the on-premises FRLs, then merely relate the appropriate Barrier Code to those that will be using Remote Access.

Hardware and Software Requirements

No additional hardware is required. The optional Private Network Access or ARS software is required.

Facility Test Calls (with Security Measures)

Feature Availability

This feature is available with all Generic 3 releases. Logoff Notification is available with G3V4 and later releases.

Description

Provides a voice terminal user with the capability of making test calls to access specific trunks, DTMF receivers, time slots, and system tones. The test call is used to make sure the facility is operating properly. A local voice terminal user can make a test call by dialing an access code. AT&T remote maintenance personnel may also use this feature to make test calls.

Four types of Facility Test Calls can be made:

- Trunk test call

  Accesses specific tie or CO trunks. DID trunks cannot be accessed.

  A user's Class of Restriction must be administered with the Facility Access Trunk Test option in order for the user to make trunk test calls.

- DTMF receiver test call

  Accesses and tests the DTMF receivers located on a Tone Detector or Call Classifier/Tone Detection circuit pack.

- Time slot test call

  Connects the voice terminal user to a specific time slot located on the Time Division Multiplex buses or out-of-service time slots.

- System tone test call

  Connects the voice terminal user to a specific system tone.

For detailed instructions for making test calls, see Chapter 5 of *DEFINITY Communications System G3i/s/vs Maintenance*, 555-204-105, *DEFINITY Communications System G3r Maintenance*, 555-230-105, *DEFINITY Communications System Generic 1 and Generic 3 Voice Terminal Operations*, 555-230-701, or *DEFINITY Communications System Generic 1 and Generic 3 Console Operation*, 555-230-700.

Security Measures

To help secure this feature from unauthorized use the following steps can be taken:

- Remove the code when not in use.
- Change the code from the factory default.
- Secure records of the code.
- Use COR to restrict which users can use the access code.

Logoff Notification, available with G3V4 and later releases, can be set to notify the system administrator at logoff that the Facility Test Calls feature is still enabled. Notification guards against inadvertently leaving the Facility Test Calls feature active. It can also alert the system administrator to unauthorized feature activation.

Consult the *BCSystems Security Handbook*, 555-025-600, for additional steps to secure your system.

Considerations

If a user has a problem with a specific system facility, Facility Test Calls can be used to test that facility for proper operation.

A DTMF voice terminal must be used to make test calls.

NOTE:
AT&T has designed the Facility Test Calls feature incorporated in this product that, when properly administered by the customer, will enable the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of test call features. In such events, applicable tariffs require that the customer pay all network charges for traffic. AT&T cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

Interactions

None.

Administration

Facility Test Calls is administered on a per-system basis by the System Manager. The Facility Test Calls access code must be assigned. Also, a user's Class of Restriction must be administered with the Facility Access Trunk Test option in order for the user to make trunk test calls.

Hardware and Software Requirements

No additional hardware or software is required.

Facility and Non-Facility Associated Signaling

Feature Availability

This optional feature is available with all Generic 3 releases, except G3vs/G3s ABP, when ISDN PRI software is purchased.

Description

Provides signaling for ISDN-PRI.

Facility Associated Signaling

Facility Associated Signaling (FAS) allows an ISDN-PRI DS1/E1 Interface

D-channel to carry signaling information for only those B-channels located on the same DS1/E1 facility (circuit pack) as the D-channel.

Non-Facility Associated Signaling

Non-Facility Associated Signaling (NFAS) allows an ISDN-PRI DS1/E1 Interface

D-channel (signaling channel) to convey signaling information for B-channels (voice and data channels) on ISDN-PRI DS1/E1 facilities other than the one containing the D-channel. As a result, a D-channel can carry signaling information for numerous B-channels located on different DS1/E1 facilities.

NOTE:
NFAS is only valid for DS1/E1 Country Protocol option 1 (U.S.).

D-Channel Backup

To improve reliability in the event of a signaling link failure, a backup D-Channel may be administered. If a signaling link failure does occur, a switch to a backup D-channel will then take place.

D-Channel Backup requires that one D-channel be administered as the Primary D-channel and that a second D-channel be administered as the Secondary

D-channel. These assignments insure that at certain times during D-Channel Backup procedures that both D-channels are in the same state. This avoids the occurrence of both switches at each end of the DS1/E1 interface selecting the same D-channel to be put into service. In these cases, the Primary D-channel is given precedence over the Secondary D-channel.

Figure 3-16 shows a possible configuration involving three ISDN-PRIs between a G3 switch and another DEFINITY system or the public network. With DS1 (24 channel) interfaces, two of the ISDN-PRIs contain a D-channel and 23 B-channels, while the other ISDN-PRI contains 24 B-channels. One of the D-channels is the Primary D-channel, and the other is the Secondary D-channel. Together, this pair of D-channels will signal for all 70 (23+24+23) of the B-channels that are part of the three PRIs.

Since the D-channels are signaling for more than one DS1/E1 facility, the D-channel Backup feature requires the use of the NFAS feature. At any given time, one of the two D-channels will be carrying Layer 3 signaling messages, while the other D-channel will be active at layer 2, but in a standby mode only. Any layer 3 messages received over the standby D-channel will be ignored. Since only one of the D-channels can be active at a time, load sharing between the two D-channels is not possible. The two D-channels can provide signaling for only a predefined set of B-channels and cannot dynamically back up other D-channels on other interfaces.

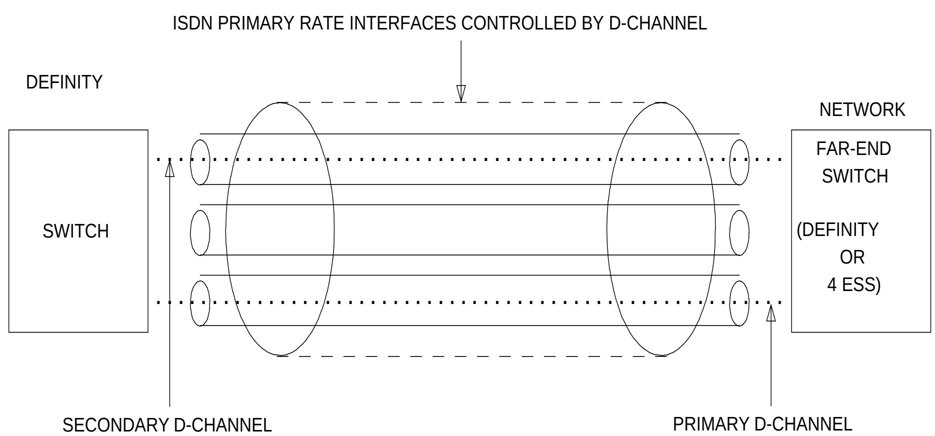


Figure 3-16. Example D-Channel Backup With Three ISDN-PRIs

D-Channel Backup Activation. D-Channel Backup can be invoked in response to the following events:

- D-Channel Failure

  If the signaling link fails on the active D-channel (D1) or the hardware carrying D1 fails, then the system will send a message over the standby D-channel (D2), which requests that D2 become the active D-channel. D2 then becomes the active D-channel and will carry all subsequent signaling messages. When the signaling link or hardware on D1 recovers from the failure, D1 becomes the standby D-channel.

- System Technician Commands

  If a system technician command requests that a D-channel switch-over take place, the first action taken by the system will be to tear down the signaling link on D1. After this has been completed, a message is sent on D2 to request that D2 become the active D-channel. D2 then becomes the active D-channel and the switch-over will be complete.

Considerations

NFAS allows a single D-channel to carry signaling information for numerous B-channels located on different DS1/E1 facilities, thus providing a more economical interface.

Only those ISDN-PRI facilities that use the NFAS feature will be capable of providing the D-Channel Backup feature. The reason for this limitation is that the two D-channels must be located on different PRI DS1/E1 facilities. As a result, the D-channels must support NFAS so that they can signal for B-channels on different ISDN-PRI DS1/E1 facilities.

When a transition from one D-channel to another occurs (D-Channel Backup is activated), all stable calls (calls that have been answered already) will be preserved. Some messages may be lost, resulting in a loss of call-related information, but the calls themselves will be maintained. The effect of the transition on unstable calls (those that have not been answered yet) is unpredictable since the results depend on which messages were lost and the contents of those messages.

Interactions

None.

Administration

The following provisioning and administration must be considered when implementing FAS and NFAS. Coordinate the following with the far-end switch for the DS1/E1 facilities to be used:

- Decide which DS1/E1 facilities will use FAS.
- Decide which of the remaining DS1/E1 facilities will carry D-channel signaling information on the 16th (E1) or 24th (DS1) channel. For those channels that have a D-Channel Backup, D-channel pairs must be allocated.
- Define Signaling Groups. A Signaling Group is a group of B-channels for which a given D-channel (or D-channel pair) will carry the signaling information. Each Signaling Group must be designated as either a FAS or NFAS Signaling Group.

- A FAS Signaling Group must contain all the ISDN B-Channels on the DS1/E1 interface associated with the group's D-channel, and cannot contain B-channels from any other DS1/E1 circuit pack. For 24-channel DS1 boards, some of the DS1 ports may use in-band (robbed-bit) signaling and be members in a tie trunk group rather than an ISDN trunk group. These tie trunks cannot be members of a Signaling Group.
- There is no restriction on which DS1/E1 ports can belong to an NFAS Signaling Group. Normally, an NFAS Signaling Group consists of one or two D-channels and several complete DS1/E1 interfaces.

  If a Signaling Group contains only a subset of a DS1/E1's B-channels (ports 1 through 12, for example), it is considered an NFAS Signaling Group, not a FAS Signaling Group. The remaining B-channels on the DS1/E1 will then be assigned as members of another NFAS Signaling Group.

■ An Interface ID must be assigned to each DS1/E1 facility in an NFAS Signaling Group. For example, if the B-channels in a Signaling Group span three DS1/E1 facilities, a unique Interface ID must be assigned to each of the three DS1/E1 facilities. This designation is required to uniquely identify the same B-channel (port) number on each of the DS1/E1 facilities in the signaling group. Therefore, this designation must be agreed upon by both sides of the interface and administered prior to initialization.

■ D-Channel Backup involves two or more ISDN-PRI DS1/E1 facilities that interconnect the switch to another PBX or to the network. Two D-channels must be present on the facilities. One of the D-channels is designated as Primary and the other as Secondary. This designation must be agreed upon by both sides of the interface and administered prior to initialization.

Hardware and Software Requirements

See the "Integrated Services Digital Network (ISDN) — Primary Rate Interface" feature for hardware and software requirements.

Flexible Billing

Feature Availability

Available with G3V4 and later releases. Flexible Billing is available in the United States for use with AT&T MultiQuest® 900 Vari-A-Bill™ Service.

Description

With Flexible Billing the DEFINITY switch or an adjunct can communicate with the AT&T network through ISDN PRI messages to change the rate at which an incoming 900-type call is billed. Rate change requests can be made anytime after the call is answered and before it disconnects.

To administer Flexible Billing, the Flexible Billing and ISDN-PRI fields on the System-Parameters Customer-Options form must be set to yes. Only an AT&T authorized representative can administer the System-Parameters Customer-Options form. You can display the form by entering the **display system-parameters customer-options** command.

All change requests are initiated by an Adjunct Processor (CPE Host or VRU) connected to the DEFINITY switch. The Adjunct Processor determines when to request a rate change based on information from customer-provided software residing on the processor.

The billing rate can be changed to any of the following:

- New Rate
- Flat Rate (not dependent on length of call)
- Premium Charge (a flat charge in addition to the existing rate)
- Premium Credit (a flat negative charge in addition to the existing rate)
- Free Call

There is never a negative charge for calls. A maximum is set for each 900 number as part of the provisioning process.

NOTE:
It is the responsibility of the customer to notify callers of rate changes. Before instructing the network to change the caller rate, the customer must inform the caller of the new rate and gain confirmation of the rate from the caller.

Applications

The following list gives examples of when a customer might decide to change the rate for a call that is in progress:

- A new rate might be established for higher valued services such as requests for particularly valuable information, conversion of text mail to speech, or technical support.
- A flat rate might be desired for consultative support, warranty registration, maintenance contracts or for the recovery of a fixed-cost service or product.
- A premium charge could be requested for callers who need specialized delivery of information such as overnight mail or FAX, recovery of the cost of a sample product, an account summary, or a transcript.
- A Premium Credit might be appropriate for promotional discounts, first-time callers, contest winners or those who use low overhead resources such as electronic media rather than human agents.

Detailed Operation

An incoming PRI call from the network contains messaging that indicates if the call can use the AT&T MultiQuest® 900 Vari-A-Bill℠ Service. This information is passed to the Adjunct Processor using ASAI messaging or to the PRI VRU adjunct. The Adjunct Processor will only be informed of the Flexible Billing capability of a call if the called VDN, split or agent is included in a domain for which notification has been requested by that Adjunct Processor (monitored domain).

The Adjunct Processor can initiate a rate change request message any time after the call has been answered and before the call disconnects. The rate change request specifies the new billing rate.

The DEFINITY switch can handle a limited number of change requests. See Appendix A, "System Parameters" for capacities information. If the switch cannot accept a change request, it notifies the Adjunct Processor. Any success or failure resulting from the request is returned to the adjunct.

Subsequent requests can be made for a call that has already had a rate change. In this case:

- If the original and subsequent requests were “flat charge” or if both were “new rate”, the previous change is overwritten. The new rate is effective from the time the original change took effect.
- If the original request was “premium charge” or “premium credit” the subsequent request must be either “premium charge” or “premium credit”. The previous change is overwritten. The new rate is effective from the time the original change took effect.

Considerations

Cellular End Offices and some other End Offices (NPA-NXX) do not have the necessary billing software to accept rate changes. Customers who receive a large percentage of calls from specific exchanges, geographical areas or cellular callers should do a careful analysis before using Flexible Billing.

Interactions

- Call Detail Recording (CDR)

  If the User Data item is being recorded, the count for this item will increase when Flexible Billing is used.

- Call Vectoring

  Incoming Flexible Billing calls can use Call Vectoring for routing.

  If a VDN Return Destination is assigned, and Flexible Billing is active for a call, the return destination is ignored and the call is forced to disconnect.

- Conference

  If a call arriving on a trunk that supports Flexible Billing is conferenced with a call from a trunk that does not support Flexible Billing, the switch sends billing change requests to the appropriate trunk. However, if calls are conferenced from different trunks that support Flexible Billing, only one trunk will be sent billing change requests.Therefore, never conference calls from two or more incoming Flexible Billing trunks.

- Transfer

  If a call is answered by a party on the local switch and is transferred to a second switch, the second switch will not receive an indication that the call has variable billing. However, Flexible Billing will be available when a call is transferred to another monitored domain on the same switch.

Administration

Flexible Billing must be enabled on the Customer-Options System Parameters form. In addition, ISDN-PRI must be enabled.

Hardware and Software Requirements

Flexible Billing requires an ASAI Adjunct Processor and customer-supplied application software.

Generalized Route Selection (GRS)

Feature Availability

This optional feature is available when ISDN PRI software is purchased with all Generic 3 releases, except G3vs/G3s ABP. Generalized Route Selection is not available with G3vs/G3s ABP.

Description

Generalized Route Selection (GRS) provides the customer voice and data call routing capabilities to select not only least cost routing, but also optimal routing over the appropriate facilities.

GRS is a capability built on the current Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) features. In AAR or ARS, routing is based on the dialed number, the Facility Restriction Level (FRL) of the call originator, the partitioning group number, and the time-of-day. By providing additional parameters in the routing decision, GRS enhances AAR and ARS and maximizes the chance of using the right facility to route the call. Also, if an endpoint incompatibility exists, it provides a conversion resource (such as Modem Pools) to attempt to match the right facility with the right endpoint.

GRS allows customers to use separate routes for voice and data calls. For data calls, GRS enables the switch to distinguish between restricted and unrestricted digital transmissions, allowing the switch to route data calls onto the appropriate facilities. GRS also provides the opportunity to integrate voice and data on the same trunk group, thereby providing certain economies.

GRS allows the system to use the Integrated Services Digital Network — Primary Rate Interface (ISDN-PRI) call-by-call selection of public network services. It also provides interworking between ISDN-PRI and non-ISDN-PRI entities.

ISDN-PRI interworking is the mixture of ISDN-PRI trunks and non-ISDN-PRI trunks in a call. A mixture of these signaling procedures is required to provide end-to-end connectivity when different type trunking facilities are used.

ISDN-PRI services add five additional routing parameters which are specified on each trunk group preference of the routing pattern. These parameters are:

- **Bearer Capability Class (BCC)**

  Identifies the type of call, such as voice calls and different type data calls.

- **Information Transfer Capability (ITC)**

  Identifies the type of data transmission (restricted, unrestricted, or both).

- **Network Specific Facility**

  Identifies the services and features to be used to complete a call.

- **Band**

  Identifies the OUTWATS band. Wide Area Telecommunications Service (WATS) is a voice-grade service providing both voice and low speed data transmission calls to defined areas (bands) for a flat rate charge.

- **Inter-Exchange Carrier (IXC)**

  Identifies the specific common carrier, such as AT&T, to be used for a call.

In GRS, there are five Bearer Capability Classes (BCCs). Customers may specify routing for each BCC according to their particular transmission needs.

Bearer Capability Classes (BCCs)

The BCCs are the mechanisms by which specialized routing is provided for the various type data calls and voice calls. Each trunk group preference in the AAR/ARS routing patterns contains a BCC parameter. When a call is originated, a route is selected based on the BCC of the originating facility. BCCs are used to classify the type of traffic permitted on this trunk in the outgoing direction. Details on how a trunk group preference is determined are given in the GRS Operation section of this description.

A set of ISDN-PRI bearer capability and low-layer compatibility parameters are defined by a BCC.

The system will determine the originating endpoint's BCC from one of the following:

- For an ISDN-BRI set, the switch determines the BCC by using information from the Bearer Capability Information Element (IE) and Low Layer Compatibility IE of the ISDN SETUP message.
- For a non-BRI terminal, the switch creates a BCC by using information about the station administration for the terminal and information obtained by performing a terminal query.
- From the administered value of the incoming trunk. For a non-ISDN trunk group, the switch creates a BCC by using the administered BCC value.
- From the ISDN-PRI bearer capability and low-layer compatibility parameters, if the call is an ISDN-PRI trunk-originated call.

The BCC associated with the routing preference in the routing pattern is administered by the system administrator. More than one BCC can be associated with each preference and the same facility can appear multiple times in a routing pattern and in multiple routing patterns.

The BCC of the originating endpoint (trunk or terminal) is matched with the BCCs of the routing preferences. An exact match is not always required. The system determines when conversion/insertion resources must be used to successfully complete a call via a compatible, but not identical, BCC.

The GRS capability will recognize one or more of the following five BCCs for each trunk group preference in the routing pattern (DCP/DMI mode is explained later).

Table 3-58. BCC Assignment

Endpoint	Voice/ Data Mode	BCC	Comments
Voice Terminal	Voice	0	
Data Line Circuit Pack	2	2	
Voice Data Set	2	2	
Modular Processor Data Module	0,1,2	1,2,4	See Note
Modular Processor Data Module-M1	1	1	For ACCUNET Switched 56 kbps Service
Modular Trunk Data Module	2	2	
Digital Terminal Data Module	2	2	
510D Personal Terminal	2	2	
Digital Communications Protocol Interface	0,2,3	2,3,4	See Note
7400A Data Module	2	2	
3270T Data Module	3	3	
3270C Data Module	3	3	
3270A Data Module	2,3	2,3	See Note

:

Legend

BCC	Type	DCP/DMI Mode
0	Voice-Grade Data and Voice	None
1	56 kbps Data (Mode 1)	1
2	64 kbps Data (Mode 2)	2
3	64 kbps Data (Mode 3)	3
4	64 kbps Data (Mode 0)	0

NOTE:
For all endpoints, the switch automatically determines its current operating mode when the data module originates. Before any call is originated, the default is Mode 2.

Since call origination from a data module determines the mode to be used on the call, it is recommended that the data module user press the Originate/Disconnect button once after changing data options. This way, the right mode is sure to be assigned to the next call.

ISDN-PRI BCC Parameters

The ISDN-PRI BCC Parameters are:

- Information Transfer Capability
- Low-Layer Compatibility
- DCP/DMI Mode

Information Transfer Capability

The information to be transferred (or type of call) requires different transmission facilities. For example, transmission needs for voice calls and data calls are generally different. Voice and voice-grade data calls can be sent over analog trunks, while high speed data calls require digital trunks.

The Information Transfer Capability parameter in the Bearer Capability Information Element (BC IE) and Low-Layer Information Element (LLC IE) have the following four values:

- Voice (speech)
- Voice-grade data (3.1 kHz transmission)
- Unrestricted digital transmission
- Restricted digital transmission.

With data calls, the switch distinguishes the information transfer capability (restricted or unrestricted) of the originating data endpoint (trunk or terminal), and then uses the information transfer capability of the data endpoint to route the call onto the appropriate facility. For BRI and PRI originating data endpoints, the information transfer capability is contained in the ISDN SETUP message. For non-ISDN data endpoints, the switch uses the information transfer capability specified by the system administrator. The default for the information transfer capability of an endpoint is *restricted*. The system administrator may change the information transfer capability to restricted or unrestricted for each non-ISDN originating endpoint.

More than one Information Transfer Capability can be supported by one BCC.

Table 3-59. Assignment of BCC Based on Information Transfer Capability

DCP/DMI MODE	Information Transfer Capability	BCC	Comments
—	Speech 3.1 kHz	0	Used for Voice/ Voice Grade Data.
M1	Unrestricted/ Restricted Digital	1	Used for Mode 1 Data (56 kbps).
M2	Unrestricted/ Restricted Digital	2	Used for Mode 2 Data (async data speed up to 19.2 kbps).
M3	Unrestricted/ Restricted Digital	3	Used for Mode 3 Data (64 kbps).
M0	Unrestricted/ Restricted Digital	4	Used for Mode 0 Data[1] (64 kbps clear channel).

1. Use BCC 4 for an unknown data mode that requires a 64 kbps channel.

Low-Layer Compatibility

The low-layer compatibility information element provides remote compatibility checking. This element is used with the bearer capability element and determines the mode of the originating caller. The low-layer compatibility information element is optional and sent only in case of data calls.

NOTE:
DCP Mode 0 does not send an LLC IE.

DCP/DMI Mode

The Digital Communications Protocol (DCP) and the Digital Mulitplexed Interface (DMI) modes are data parameters of the originating data facility. These modes are not applicable to voice.

The mode values (0, 1, 2, and 3) are administered for data and Alternate Voice/Date (AVD) non-ISDN-PRI trunk groups. These mode values determine the BCC of the trunk groups.

Determination of BCC at Tandeming or Terminating System

Determination of the BCC for an incoming call from a ISDN-PRI trunk to a tandem or terminating switch is based on the BCC parameters received on the signaling channel (D-channel) of the trunk. This includes the ITC (restricted or unrestricted) if the call is a data call.

Determination of the BCC for an incoming call from a non-PRI trunk will be as follows:

- If the incoming trunk is a voice trunk, then the BCC is defaulted to 0.
- If the incoming trunk is a data, AVD, or RBAVD (robbed-bit AVD) trunk, then the BCC and ITC are administrable.

GRS Operation

The AAR/ARS routing pattern will contain an indication for each trunk group preference showing which BCC or BCCs can use that trunk group. A trunk group preference may have more than one BCC.

GRS uses a *look-ahead* algorithm when determining which preference in a routing pattern to choose. GRS first attempts to find an exact match between the originator's BCC and the corresponding allowed BCC for any of the preferences in the routing pattern. Therefore, if preference 1 does not have an exact match (even though there are available compatible members in preference 1), it will be skipped over if a subsequent preference in the same pattern has an allowed BCC that exactly matches the originator's BCC.

After matching the BCCs, G3i will then match the ITCs. The originator's ITC is matched to the route preference ITC. Unrestricted (unre) matches on ''unr'' or ''both.'' Restricted (rest) matches on ''rest'' or ''both.''

NOTE:
ITC matching only applies to data calls (BCC 1 through 4).

As an example of how GRS chooses a trunk group preference, assume preference 1 in a pattern has BCC 0 and BCC 2 set to yes, while preference 2 has BCC 1, BCC 3, and BCC 4 set to yes. A voice or Mode 2 data call accessing this pattern will use the first preference, while a Mode 1, Mode 3, or Mode 0 data call will use the second, independent of the availability of trunks in the first preference.

When an exact match is not found in any of the routing pattern preferences, calls are treated as follows:

- Calls With an Originating BCC of 0

A BCC 0 originated call (such as voice or analog modem) will not be denied routing by GRS, even if the routing pattern lacks a preference with BCC 0 set to yes. This allows the user to use voice transfer to data when making a data call, without the need for data preindication.

If a BCC 0 originated call accesses a routing pattern for which no preference has BCC 0 set to yes, then GRS will choose a preference with BCC 2 set to yes, if one exists. If none exists, the next preferred order would be a preference with BCC 1 set to yes, followed by BCC 3, and finally, BCC 4. Since each preference must allow at least one BCC to be passed, a BCC 0 (voice) originated call will never be blocked by GRS. The call is of course still subject to other restrictions, such as FRL restrictions. The ITC does not help select a preference.

Since BCC 0 (voice) has no ITC, the switch selects an ITC from the routing pattern when a BCC 0 call is being routed as a data call. Table 3-60 shows how the ITC codepoint in the Bearer Capability IE is determined.

Table 3-60. Determination of ITC Codepoint

Originating Endpoint's ITC	Routing Preference's ITC				ITC codepoint in BC IE
	restricted	unrestricted	both endpoint	both unrestricted	
voice	x				restricted
voice		x			unrestricted
voice			x		unrestricted
voice				x	unrestricted

- Calls With an Originating BCC of 2

  If a BCC 2 originated call accesses a routing pattern for which no preference has BCC 2 set to yes, then GRS will choose a preference with BCC 0 set to yes, if one exists. If none exists, the call will be blocked with intercept treatment.

- Calls With an Originating BCC of 1, 3, or 4

  A DCP/DMI Mode 0 (BCC 4), Mode 1 (BCC 1), or Mode 3 (BCC 3) originated call requires an exact match on at least one preference in a routing pattern in order for GRS to allow the call to complete. For example, a Mode 1 originated call will complete only if the accessed routing pattern has a preference with BCC 1 set to yes. The ITCs must also match.

When an ISDN-PRI trunk group preference is accessed, the BCC information is encoded and sent in the outgoing ISDN SETUP message to the distant-end as shown below. The BCC information sent to the far end is important, because the

BCC information that the far-end receives in the SETUP message will become the originating BCC for the far-end's incoming trunk call.

- If an exact match of the originator's BCC and ITC has been found, then that Bearer Capability is encoded and sent in the ISDN SETUP message to the far-end. If the call is a data call, the system uses the ITC of the routing pattern to encode the SETUP message as shown in Table 3-61.

Table 3-61. Encoding of Setup message

Originating Endpoint's ITC	Routing Preference's ITC				ITC codepoint in BC IE
	restricted	unrestricted	both endpoint	both unrestricted	
restricted	x				restricted
restricted			x		restricted
restricted				x	unrestricted
unrestricted		x			unrestricted
unrestricted			x		unrestricted
unrestricted				x	unrestricted
voice[1]	x				restricted
voice		x			unrestricted
voice			x		unrestricted
voice				x	unrestricted

1. A voice originated call without data preindication that is routed to a routing pattern with data preferences only.

- If an exact match is not found, but the call is allowed to proceed, then the BCC encoded in the SETUP message sent to the far-end is that of the routing pattern. For example, if a BCC 2 (for example, DTDM) endpoint originates a call that accesses a pattern that has one preference with only BCC 0 set to yes, then the switch automatically inserts a modem pool for this call. In effect, the modem pool is converting BCC 2 to BCC 0. The far-end cannot distinguish this call from a BCC 0 originated call that has no modem pool inserted. Therefore, BCC 0 is sent in the SETUP message. This may in turn determine routing decisions by the far-end. Additional routing decisions are made as shown in the following tables.

BCC and ITC Determination on Calls from Endpoints to ISDN-PRI Trunks

Table 3-62. Calls from Endpoint to ISDN-PRI Trunks

Originating	Chosen BCC from the Routing Pattern				
BCC	**BCC 0**	**BCC 1**	**BCC 2**	**BCC 3**	**BCC 4**
BCC 0	P	PT	PT	PT	PT
BCC 1	B	P	B	B	B
BCC 2	PM	B	P	B	B
BCC 3	B	B	B	P	B
BCC 4	B	B	B	B	P

Legend

B	Block the call with intercept treatment
P	Allow the call and send the originating endpoint's BCC in the SETUP message. Use the ITC as shown in the following table
PT	Allow the call and send the BCC and ITC chosen from the routing pattern in the SETUP message
PM	Insert a pooled modem for the call and send the BCC and ITC chosen from the routing pattern in the SETUP message

If BCC 1, 2, 3, or 4 is chosen from the preceding table, the following table is used to determine the appropriate ITC.

Table 3-63. Calls from Endpoints to ISDN-PRI Trunks

Originating ITC	Chosen ITC from the Routing Pattern			
	unr	**rest**	**both endpt**	**both unr**
unr	P	B	P	PU
rest	B	P	P	PU

Legend

B	Block the call with intercept treatment
P	Allow the call and send the originating endpoint's ITC in the SETUP message
PU	Allow the call and send ''unrestricted'' in the SETUP message

BCC and ITC Determination on Calls from Trunks to ISDN-PRI

Table 3-64. Calls from Trunks to ISDN-PRI Trunks

Originating	Chosen BCC from the Routing Pattern				
BCC	**BCC 0**	**BCC 1**	**BCC 2**	**BCC 3**	**BCC 4**
BCC 0	P	PT	PT	PT	PT
BCC 1	B	P	B	B	B
BCC 2	PT	B	P	B	B
BCC 3	B	B	B	P	B
BCC 4	B	B	B	B	P

Legend

B	Block the call with intercept treatment
P	Allow the call and send the incoming trunk's BCC in the SETUP message. Use the ITC as shown in the following table
PT	Allow the call and send the BCC and ITC chosen from the routing pattern in the SETUP message

If BCC 1, 2, 3, or 4 is chosen from the preceding table, the following table is used to determine the appropriate ITC.

Table 3-65. Calls from Trunks to ISDN-PRI Trunks

Originating	Chosen ITC from the Routing Pattern			
ITC	**unr**	**rest**	**both endpt**	**both unr**
unr	P	B	P	PU
rest	B	P	P	PU

Legend

B	Block the call with intercept treatment
P	Allow the call and send the incoming trunk’s ITC in the SETUP message
PU	Allow the call and send ‘‘unrestricted’’ in the SETUP message

The system does not insert pooled modem for any interworking trunk-to-ISDN-PRI trunk calls. The BCC and ITC of an incoming trunk is determined as follows:

- ISDN-PRI Trunk BCC and ITC are in the received SETUP message
- AVD Trunk BCC and ITC are the BCC and ITC values administered on the trunk group form
- RBAVD Trunk BCC and ITC are the BCC and ITC values administered on the trunk group form
- Data Trunk BCC and ITC are the BCC and ITC values administered on the trunk group form
- Voice Trunk BCC is 0.

BCC and ITC Determination on Calls from ISDN-PRI Trunks to Endpoints (GRS not Involved)

Table 3-66. Calls from ISDN-PRI Trunks to Endpoints

Originating	Terminating Endpoint BCC				
BCC	**BCC 0**	**BCC 1**	**BCC 2**	**BCC 3**	**BCC 4**
BCC 0	P	P	PM	P	P
BCC 1	P	P	P	P	P
BCC 2	P	P	P	P	P
BCC 3	P	P	P	P	P
BCC 4	P	P	P	P	P

Legend

P	Allow the call, and (1) if it is a voice originated call, let the calling user decide whether the terminating endpoint is the correct endpoint or not based on audible feedback (for example, data tone), or (2) if it is a data call, the data handshake procedure will establish or drop the call based on the compatibility of the endpoints.
PM	Insert a pooled modem and terminate the call to the endpoint. The ITC defaults to restricted in this case.

NOTE:
The system does not use ITCs when terminating to an endpoint.

Considerations

- **ACCUNET and SDDN Services**

  The system will be able to use ARS tables to route calls to these networks. For example, BCC 1 is a 56 kbps service. If an ACCUNET 56 kbps Service trunk group is in a routing pattern that uses GRS, BCC 1 should be set to yes.

- **SDI Service**

  The system will be able to support the AT&T Switched Digital International (SDI) Service, which requires 64 kbps unrestricted service for Mode 0 calls or 64 kbps restricted or unrestricted rate adapted to 56 kbps for Mode 1 calls. SDI will reject calls encoded as 64 kbps restricted without rate adaptation.

- **Integrated Access on Non-ISDN-PRI Trunks**

  The T1 carrier access to the AT&T serving office will allow sharing of the same trunk group for voice and data calls. For example, the same trunk group may carry both voice calls (requires a BCC of 0) and Mode 1 data calls (requires a BCC of 1). This situation requires the trunk group preferences in the ARS routing pattern to be administered with a ‘‘yes’’ for both BCC 0 and BCC 1.

- **Voice and Voice-Grade Data Calls**

  Voice and voice-grade data calls cannot be routed separately if voice-grade data calls are assigned BCC 0. Voice-grade data calls could be assigned BCC 2.

- **Modem Pooling**

  A system originating a data call over a public or private network is responsible for inserting the pooled modem (when needed). Since the originating switch knows the speed of the call, it can insert the appropriate pooled modem. Since tandem switches do not know the speed of the call, they cannot make the decision on the type of pooled modem needed.

Interactions

The following features interact with the Generalized Route Selection feature.

- **Automatic Route Selection (ARS) and Automatic Alternate Routing (AAR)**

  In ARS/AAR, routing is based on the dialed number, the Facility Restriction Level of the call originator, the partitioning group number, and the time-of-day. In GRS, routing of calls is additionally based on the BCC and ITC to distinguish voice from data calls. For all trunks, ISDN-PRI as well as non-ISDN-PRI, the BCC and ITC are checked to see if the route selected is compatible.

- **AAR/ARS Partitioning**

  It is possible to perform GRS administration for each partition separately by using different routing patterns.

- **Interworking**

  Generalized Route Selection will support interworking; that is, the routing patterns may contain a combination of ISDN-PRI and non-ISDN-PRI trunking facilities. For non-ISDN-PRI trunking facilities, the BCC and ITC are determined by (default) administration. For ISDN-PRI trunking facilities, the BCC and ITC are determined by the information received on the signaling channel (D-channel) of the trunk.

- **Call by Call Service Selection**

  For each preference in a routing pattern, the customer may optionally administer an IXC code and a Network Specific Facility parameter to be used when an outgoing call is made using an ISDN-PRI facility. Call by Call (CBC) Service Selection allows the dynamic identification of a specific service type request on a per call basis. For CBC Service Selection feature, the trunk group is administered as CBC. This allows the customer to pool several types of services together and assigns the service type to them on a call basis.

- **Data Pre-Indication**

  When the Data button is pressed on DCP voice terminals, the switch uses the BCC and ITC of the associated data module.

- **Electronic Tandem Network (ETN) Services**

  To provide ETN services, the ISDN-PRI trunks can be used to interconnect any combination of the following switches: System 85 R2V4, DEFINITY system Generic 1, Generic 2, or Generic 3. (An ETN is a network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations.)

  The ETN Traveling Class Mark (TCM) will be passed between the tandem nodes by the TCM information element of the SETUP message on the ISDN-PRI facilities. (Traveling Class Marks represent the Facilities Restriction Level and are used by the distant tandem switch to determine the best available facility consistent with the user's calling privileges.) The Satellite Hop Control (Conditional Routing) Count and End-to-End Connectivity message, which is used in System 85 R2V2 or Generic G2, is tandem into the SETUP message without being analyzed. For non-ISDN-PRI tandem trunks, the TCM is outpulsed after the destination address.

Administration

The following additional items are administered by the System Manager for GRS.

- **Routing pattern BCCs** — For each trunk group in the Routing Pattern, there will be an indication of what BCCs can use that preference. The values are 0, 1, 2, 3, and 4. More than one BCC may be supported by one trunk group preference. The BCCs assigned to a trunk group in a Routing Pattern may or may not be the same value assigned to the same trunk group in another Routing Pattern.

  The "ITC (Information Transfer Capability)" field administers the type of traffic (restricted, unrestricted, or both) that this routing preference may carry. This field is displayed and administrable if at least one of the BCC data preferences is set to `y`. If only BCC 0 is set to `y`, then the "ITC" field is not displayed. The value of the "ITC" field may be set to `rest` (restricted), `unre` (unrestricted), or `both`. The default is `rest`.

The "BCIE (Bearer Capability Information Element)" field specifies how to create the Information Transfer Capability codepoint in the Bearer Capability IE of the SETUP message. This field only applies to ISDN-PRI trunks and is displayed and administrable if the "ITC" field is administered as `both`. The value `both` provides extra flexibility for managing facilities in a mixed unrestricted/restricted environment. The value of the "BCIE" field may be set to `ept` (endpoint), `unr` (unrestricted), or `both`. The default is `ept`.

If the "BCIE" field is set to `ept`, the ITC of the originating endpoint is used to encode the SETUP message. If the "BCIE" field is set to `unre`, then `unrestricted` is encoded in the SETUP message.

- **Trunk Group BCCs** — Each non-ISDN-PRI, data, AVD, or RBAVD trunk group is administered a BCC to identify the type of traffic on the trunks in that group.

  The "ITC (Information Transfer Capability)" field is provided for non-ISDN trunk groups only. This field is displayed and administrable only if the "Comm Type" field is administered as `data`, `avd`, or `rbavd` and the "BCC" field is not set to `0`. The value of the "ITC" field may be set to `rest` (restricted) or `unre` (unrestricted). The default is `rest`.

- **Access Endpoint BCCs** — For access endpoints where the "Communication Type" field is set to `56K-data` or `64K-data`, the "ITC (Information Transfer Capability)" field is displayed and administrable. The value of this field may be set to `restricted` or `unrestricted`. The default is `restricted`.

- **Station BCCs** — For all non-ISDN endpoints, the "ITC (Information Transfer Capability)" field is displayed and administrable on the Station Data Module Page. The value of this field may be set to `restricted` or `unrestricted`. The default is `restricted`. This field is not displayed for voice-only stations and BRI stations.

  For BRI stations, the Default ITC (Information Transfer Capability) is displayed and administrable on the Station Data Module Page. This field administers the ITC for BRI data modules that originate an administered connection (AC). The value of this field may be set to `restricted` or `unrestricted`. The default is `restricted`.

- **Data Module BCCs** — For all non-ISDN endpoints specified as ''`pdm`,'' `data-line`, `netcon`, or `tdm`, the "ITC (Information Transfer Capability)" field is displayed and administrable. The value of this field may be set to `restricted` or `unrestricted`. The default is `restricted`.

  For BRI data modules, the Default ITC (Information Transfer Capability) is displayed and administrable. This field administers the Information Transfer Capability for BRI data modules that originate an administered connection (AC). The value of this field may be set to `restricted` or `unrestricted`. The default is `restricted`.

For endpoints (DCP data modules, voice terminals, and so on) a read-only “BCC” field appears on the screen form for that endpoint. This field reflects the endpoint’s current BCC which is determined automatically by switch software.

Hardware and Software Requirements

No additional hardware is required.

Optional AAR, ARS, and ISDN-PRI services software is required.

Go to Cover

Feature Availability

Go to Cover is available with all Generic 3 releases.

Description

Allows users, when making a call to another internal extension, to send the call directly to coverage.

Go to Cover is activated by pressing a Go to Cover button. This button can be used only after the call is ringing.

Details of how Go to Cover is used in conjunction with Call Coverage are given in the "Call Coverage" feature description elsewhere in this chapter.

Consideration

Go to Cover gives the calling party the option to send calls directly to coverage.

Interactions

The following features do not redirect to coverage unless the caller presses the Go to Cover button:

- Intercom — Automatic
- Intercom — Dial
- Priority Calling

Go to Cover can only be used if the called party is assigned a call coverage path; that is, the called party must have alternate answering positions assigned.

Administration

Go to Cover is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Go to Cover button.

Hardware and Software Requirements

No additional hardware or software is required.

Hold

Feature Availability

Hold is available with all Generic 3 releases.

Description

Allows voice terminal users to disconnect from a call temporarily, use the voice terminal for other call purposes, and then return to the original call.

Multi-appearance Voice Terminal Hold

Multi-appearance voice terminals have a Hold button for activating the Hold feature. Multi-appearance voice terminal users can hold a call on each call appearance. To hold a call, a user, while active on a call, simply presses the hold button and the call is held at the call appearance being used for the call.

Single-line Voice Terminal Hold

Two types of Hold (soft hold and hard hold) are provided for single-line voice terminal users. With soft hold, the user can hold the current call, consult with another party or activate/deactivate a feature, and return to the soft held call. This type of hold is used to conference or transfer a call that *includes the held call.* Hard hold can be used to hold the current call and then perform operations that *do not include the held call.* These operations could include calling another party, answering a waiting call and transferring or conferencing the waiting call with another party, activating or deactivating features, and so on.

To activate soft hold, a user, while active on a call, presses the Recall button or flashes the switchhook. The user can then conference or transfer the call that is on hold. If the user dials another party and presses the Recall button or flashes the switchhook a second time, the held call is conferenced with the user and the other party. For a call transfer, the system ignores any subsequent presses of the Recall button or flashing of the switchhook. If the user is the controller of a conference call, and presses the Recall button or flashes the switchhook, the last party added to the conference is dropped. The user can transfer the call by hanging up after conferencing the call.

To activate hard hold, a user, while active on a call, presses the Recall button or flashes the switchhook. The user then dials the Answer Hold-Unhold access code. This puts the call on hard hold. The user can then perform any operation that does not involve the held call. When the user wants to return to the hard held call, the user should go on hook. The held call rings the voice terminal and can then be answered.

If a user has a call waiting and activates hard hold, the current call is placed on hard hold and the waiting call is answered automatically.

Considerations

With the Hold feature, voice terminal users can temporarily disconnect from one call and handle another call. For example, a busy voice terminal user who receives another call can place the first call on hold and answer the second call. This results in fewer missed calls. The Hold feature can also be used when a user receives a call and needs to make another call to obtain information for the calling party.

A call involving an attendant cannot be held by a single-line voice terminal user. A call involving an attendant can be held by a multiappearance voice terminal user, unless the user attempts a conference or transfer of the call.

A call dialed by a single-line voice terminal can be dropped within the first ten seconds (after dialing is completed) by flashing the switchhook.

One party on hold can hear music if the Music-on-Hold feature is provided. The music is removed when the voice terminal user reenters the call.

Interactions

The following features interact with the Hold feature.

- Automatic Callback

  A single-line voice terminal user cannot receive an Automatic Callback call while it has a call on Hold.

- Bridged Call Appearance

  Any user, active on a bridged call, can place the call on hold. If a call on a bridged call appearance is placed on hold and no other users with a bridged call appearance of the same extension number are connected to the call, the status lamp at the Bridged Appearance button indicates that the call is on hold. If the primary extension or another bridged appearance is connected to the call, the status lamp at all bridged appearances indicates an active status for the call.

- Interaction with Multi-Appearance Conference/Transfer

- Interaction with Priority Calling

  It is possible to receive priority ringing and have a call on soft hold.

- LWC

  A held multiappearance voice terminal user can activate LWC toward the holding user.

  A single-line voice terminal user cannot activate LWC toward another user while a call is on soft hold.

- Personal Central Office Line

  When a user, active on a PCOL call, puts the call on Hold, the lamp flutters or winks the same as a call appearance lamp for hold-button. When a user, active on a PCOL call, puts the call on Hold, the status lamp associated with the PCOL button does not track the busy/idle status of the PCOL.

Administration

The Hold feature is administered by the System Manager. The only administration required is to assign the Answer Hold-Unhold feature access code.

Hardware and Software Requirements

No additional hardware or software is required.

Hold - Automatic

Feature Availability

This feature is available with G3i-Global, G3rV1, all G3V2, and later releases.

Description

System-Wide Administrable Hold—Automatic (hereinafter referred to as Automatic Hold) is a station/attendant feature. The system comes with this feature turned off by default.

The feature is described for stations as well as the attendant because of the similarity of the operation. Automatic Hold allows attendants and Multi-Function station set users to alternate easily between two (or more) calls. For example, depression of a second call appearance automatically puts the active call (if any) on hold, and makes the second call appearance active. If the Automatic Hold feature is disabled (the default), the active call appearance is dropped.

The ability to administer the Automatic Hold feature allows the customer to enable and disable the Automatic Hold Feature on a system-wide basis. A user of a multibutton station may place on Automatic-Hold as many calls as the number of call-appearances minus one.

The controlling station can have only one ’soft’ auto-held call at a time. A ’soft’ hold is the state of a line after the conference or transfer button has been pressed but before either process is completed. The controlling station is guaranteed the ability to reenter any auto-held call later unless the auto-held party or parties disconnect or the auto-held tone times out.

Considerations

When Automatic Hold is disabled, if the Multi-Function Digital Telephone (MFDT) selects an inactive call appearance while there is still a call on an active call appearance, the active call is dropped. If Automatic Hold is enabled, when the MFDT selects an inactive call appearance while there is a call on an active call appearance, the active call is put on Hard-Hold. This is the same as if the MFDT had depressed the HOLD button and then selected a call appearance.

Whenever only the Automatic Hold feature is involved, for example, when the attendant on an active loop presses a second loop, the active call is placed in Hard-Hold. Soft-Hold allows the attendant to extend calls and conference calls together.

The Held Call Timed Reminder does not apply to conference calls and is not initiated when a conference is placed on hold. The net result is that the attendant

Automatic Hold call is treated the same as an attendant call placed in Hold by the depression of the HOLD button.

The Automatic Hold feature operates in conjunction with the START key or Automatic Start feature of an attendant console. The START key/Automatic Start operation has precedence over the Automatic Hold feature. Any feature that uses the Start key/Automatic Start operation places the call on the active loop on Soft-Hold.

Interactions

As mentioned earlier, the Automatic Hold feature is identical to the action generated when an MFDT places a call on hold using the HOLD button and then selecting the (another) inactive call appearance. The features, therefore, work identically, except for the saving of a button push. The Automatic Hold feature is implemented as an equivalent to pressing the HOLD button. A call placed in Automatic Hold is in Hard-Hold, as with the depression of the HOLD button.

Therefore, Automatic Hold, when invoked by the methods already mentioned, operates identically to the HOLD feature and its interactions. See the "Hold" feature for additional information.

Automatic Hold interacts with DCS and Centralized Attendant Service (CAS). It operates transparently with DCS and CAS. Note that the Auto-Hold feature is administered separately for each node in a DCS network.

Administration

Automatic Hold is administrable on a system-wide basis only by persons whose administration ID provides access to system parameters. Administration of the auto hold feature takes place on the System-Parameters Feature Form.

Hardware/Software Requirements

No special hardware is required.

Hot Line Service

Feature Availability

Hot Line Service is available with all Generic 3 releases.

Description

Allows single-line voice terminal users, by simply lifting the handset, to automatically place a call to a preassigned extension number, public or private network telephone number, or feature access code.

The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the Hot Line Service user lifts the handset, the system automatically routes the call to the stored number and the call completes as though it had been manually dialed. If the appropriate feature access code is prefixed to the stored number, AAR, ARS, Data Privacy, or Priority Calling can be used on the call. Also, if the Public or Private Network Access code is the stored number, the voice terminal user is connected to an outgoing trunk and can dial the outside number.

A Hot Line Service voice terminal receives calls allowed by its COR. Call reception is not affected by Hot Line Service. Likewise, the Hot Line Service destination is not affected by Hot Line Service.

A DDC, a UCD, a TEG extension number, or any individual extension number within any of the groups can be a Hot Line Service destination. Also, any extension number within a DDC group, UDC group, or TEG can have the Hot Line Service feature assigned.

Considerations

The Hot Line Service feature is useful in any application where very fast service is required. Also, if a voice terminal is used only for accessing a certain facility, it can be assigned to Hot Line Service. The Hot Line Service voice terminal user simply lifts the handset and is connected to that facility.

The number of voice terminals that can be assigned Hot Line Service is not limited, and the number of voice terminals that can be assigned the same destination is not limited. The limit, if any, would be on the number of entries that can be stored in the Abbreviated Dialing lists.

Interactions

A Hot Line Service user cannot activate any feature unless the access code is, or is part of, the destination number.

- Bridged Call Appearance — Single-Line Voice Terminal

  If a single-line voice terminal is administered for Hot Line Service, bridged appearances of that voice terminal's extension also places a hot line call automatically when a user goes off-hook on that bridged appearance.

- Loudspeaker Paging Access

  Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

- Ringback Queuing

  If a Hot Line Service call accesses a trunk group with Ringback Queuing assigned, the call can queue unless the voice terminal is termination restricted by its COR. Queuing, when applicable, is automatic on single-line voice terminals; dialing is not required.

Administration

Hot Line Service is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Abbreviated Dialing Lists
- Hot Line Destination Number

Hardware and Software Requirements

No additional hardware or software is required.

Hunting

Feature Availability

Hunting is available with all Generic 3 releases.

Description

Checks for the active or idle status of extension numbers in one or more ordered groups. If all members of a group are active, the call can route to another group through Call Coverage or can wait in a queue for an available group member, if a queue is provided.

Hunting is accomplished through the ACD, Call Coverage, DDC, and UCD features. The order of hunting is defined under each individual feature.

Considerations

Hunting is useful whenever a group of voice terminal users receives a high volume of calls. It minimizes call completion time and attendant assistance is not required.

Agents should not be used for hunt group calls when they are used for ACD split calls. This is because all ACD calls are answered before hunt group calls. That is, no hunt group calls are answered until all ACD calls are answered.

The oldest call waiting termination is only supported for agents who are servicing ACD calls only.

Interactions

Individual attendant extensions can be in hunt groups. However, attendant return call features do not work for these types of calls.

Administration

Hunting is administered through the "Automatic Call Distribution (ACD)", "Call Coverage", "Direct Department Calling (DDC) and Uniform Call Distribution (UCD)" features. Administration of each of these features is discussed under that feature elsewhere in this chapter.

Hardware and Software Requirements

No additional hardware or software is required for Call Coverage, DDC, and UCD. ACD requires ACD software. Call Vectoring is also required for vector-controlled splits.

Inbound Call Management (ICM)

Feature Availability

Applications running an Inbound Call Management (ICM) application can be connected to any DEFINITY switch that can be provisioned with the necessary CallVisor ASAI interface. The G3vs and G3s ABP models cannot be equipped with a CallVisor ASAI interface.

Planners must consider the ICM traffic, rated switch capacity, CallVisor ASAI interface traffic, and rated capacity of the adjunct application processor when planning an ICM installation. AT&T's Technical Design Center can provide consulting services to assist in such planning.

Description

ICM improves the handling of inbound calls in such applications as telemarketing, claims processing, etc. An application on a host processor is integrated with switch features such as Automatic Call Distribution (ACD), Call Vectoring, and Call Prompting to provide enhanced features and improve efficiency. The host process may be a CallVisor/PC, CONVERSANT voice system, Telephony Services Server (running Novell NetWare(R)) serving a local area network, or a processor from one of AT&T's CallVisor ASAI vendor partners. The CallVisor ASAI Planning Guide provides information on the various vendor partners.

The CallVisor ASAI link is a two-way interface that lets applications receive information about calling parties, prompted digits, called number, etc. The applications can request that the switch route calls, transfer calls, etc. A variety of queries and notification capabilities are also available.

The following are some typical ICM applications:

- The DEFINITY system passes calling party information (ISDN CPN/BN) and the call outcomes to a host application for screen pop, and possible (later) supervisory transfers (with screen duplication).
- The DEFINITY system provides the host application with caller and prompter information about all incoming calls to a selected number. The host application consults a database and then tells the switch where to route the calls (preferred agent, *best customer* treatment, accounts receivable, etc).
- The DEFINITY system uses the Call Prompting feature to obtain a customer account number and then passes this information to the host for call routing or screen pop.

- The DEFINITY system can send CPN/BN and DNIS to a voice response unit (VRU), connect the caller to the voice response unit, and let the VRU interact with the user to determine handling for the call. Such a system can verify callers' identity and provide access to database information (claims status, account balance, etc).
- A host application can transfer a call to an ACD and have the call treated (and tracked on CMS) as an ACD call. This is known as ''Direct Agent Calling.''
- A host application can attach application-specific information to an ICM call using "User-to-User Information" fields. If the host then transfers the call to another DEFINITY switch over primary rate interface (PRI) facilities, a CallVisor ASAI application at the receiving switch receives the application-specific information. Therefore, if an ICM application at one switch has determined a caller's account number, claim number, etc., the information could be passed to a special list (for example) on another switch where the application may have to transfer the call.

The CallVisor ASAI Technical Reference describes additional application scenarios in greater detail.

Data Screen Delivery

Passing incoming call information (CPN/BN, DNIS, Lookahead Interflow information, digits collected from Call Prompting, agent selected) to a CONVERSANT voice system can be used to deliver the appropriate data screen when the voice call is delivered to an agent. Data screens can also be transferred or duplicated by the CONVERSANT voice system for transferred or conferenced calls. A simplified configuration of this type of application is shown in Figure 3-17. CONVERSANT VIS is referenced for illustrative purposes only, indicating both switch and CONVERSANT capabilities; other adjunct processors have similar capabilities but should be verified for a particular application.

In this application, the CONVERSANT voice system or host requests notification for events (call offered, call ended, call connected, call dropped, call transfer, alerting, and so on). The switch notifies the CONVERSANT voice system using event reports when the call arrives, when the agent answers, when the call drops, and so on. The CONVERSANT then passes the appropriate information to the host so it can send that data screen to the agent's data terminal. Knowing a call drops prior to being answered, the CONVERSANT voice system can track abandoned calls or use CPN/BN information for call backs.

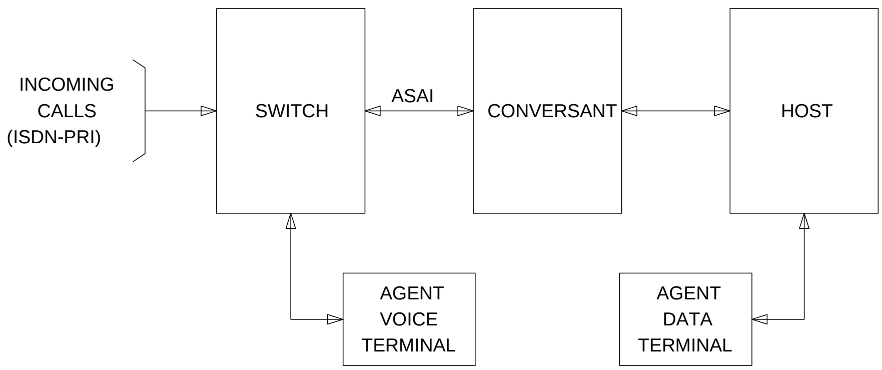


Figure 3-17. Simplified ICM Configuration for Data Screen Delivery

The ASAI interface could also be shown above direct to the host. CONVERSANT. does not have to be in the middle if the host supports ASAI.

Integration With Speech Processing Adjuncts

ICM can be used to provide integration with Voice Response Units (VRUs). The advantages of using ICM with the CallVisor ASAI in addition to tip/ring interfaces are as follows:

- Data screen integration is provided on transferred calls.
- Notification of answer is provided on internal calls (CallVisor ASAI capabilities let you know what happens with the call).
- Delivery of ISDN network information such as CPN/BN/DNIS is provided (instead of having to prompt for this information).

A simplified configuration of this application is shown in Figure 3-18. In this application, the CallVisor ASAI link is used by the switch to pass incoming call information to the CONVERSANT voice system. The call is distributed by the switch ACD to an available voice line. After collecting digits via a DTMF keypad, the CONVERSANT voice system transfers the call back to an ACD split or specific agent on the switch via CallVisor ASAI messages. If the call is transferred to a split agent, the CallVisor ASAI link is used by the switch to pass an event report containing which agent in the split receives the call. The CONVERSANT voice system forwards the agent identification to the host for delivery of the associated data screen to the agent selected to handle the call.

Digits collected by the CONVERSANT voice system are not passed to the switch to display on the agent's voice terminals but can be displayed on the agent's data terminals. If the digits collected by the CONVERSANT voice system are the extension to which the call is to be routed, these routing digits are passed to the switch as the destination in the CallVisor ASAI third party make call request. The third party make call request is used by the CONVERSANT voice system to set up various types of calls.

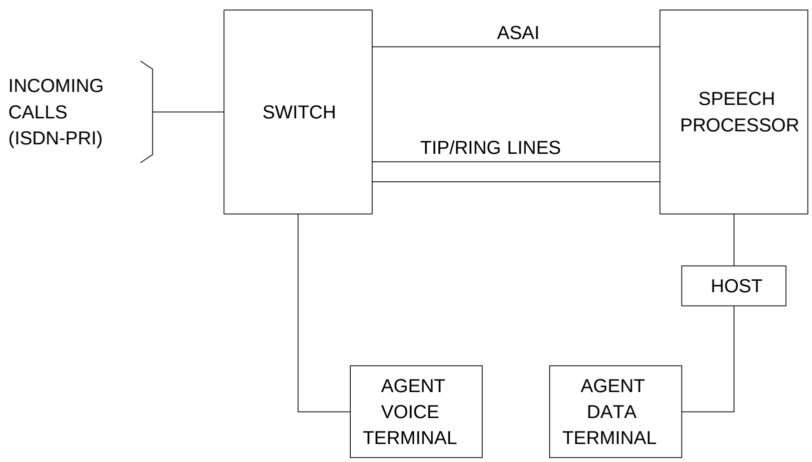


Figure 3-18. Simplified ICM Configuration for Speech Processor Integration

Host/Adjunct Call Routing

Incoming call information can be used by the host or CONVERSANT voice system adjunct to route the call to a split, vector, or particular agent (basically any valid extension number). The call could even be routed off of the switch if desired. The CONVERSANT voice system can also use the incoming call information to tell the switch that the call should be treated as a priority call. Routing can be based on the area code dialed from, the country code, digits collected from the Call Prompting feature, dialed number or service, agent availability, or information in a customer database.

To implement adjunct (CONVERSANT voice system) call routing, calls must come into a vector which contains an `adjunct routing` vector command. When the `adjunct routing` vector command is encountered, the switch initiates the route CallVisor ASAI capability. Vector processing proceeds with the next step (which could provide ringing, announcements, music, and so on) while the caller waits. A default split or answering position can also be specified in the vector, in case the CONVERSANT voice system does not respond in the administered amount of time (determined by the announcement/wait steps). Announcement and wait steps are needed to give the host time to respond.

Direct Agent Calling

DAC is a new function that allows an adjunct to initiate or transfer a call to a particular ACD agent and have the call treated as an ACD call.

Calls that originally enter the switch as ACD calls and are rerouted to a particular agent via adjunct routing, or transferred from a tip/ring agent to a live agent via a “third party make call” request, are treated as ACD calls for the duration of the call. This is important for a number of reasons:

- The call may queue for the agent.
- Agents need to receive zip tone when these calls are delivered.
- Agents may have After Call Work associated with these calls.

The CMS and BCMS correctly measure these calls as ACD calls.

Direct Agent Calls have the highest priority of any calls.

Adjunct Activation of Direct Agent Calling

CallVisor ASAI third party make calls and route select calls with the direct agent call option are treated as direct agent calls. The receiving agent’s extension appears as the destination and the split extension in the direct agent call option.

Delivery of DAC

If the agent receiving the direct agent call is available to answer an ACD call in the associated split, the direct agent call is delivered to the agent. Zip tone (480 Hz for a 1/2 second, not repeated) is applied if the agent is automatic answer.

If the receiving agent is not available to answer an ACD call (for example, the agent is busy on a call, in the After Call Work mode, or in the Auxiliary Work Mode), the receiving agent is notified with a ring-ping if the agent has a multi-function voice terminal or is on-hook. If the receiving agent has a single-line voice terminal and is not available, the receiving agent will hear call waiting tone (even when the Call Waiting feature is not assigned) if the agent is off-hook. The ring-ping or call waiting tone is given only once per call when the call is queued. The active work mode button lamps for the associated split on the receiving agent’s voice terminal will do a fast flutter, indicating a direct agent call is waiting. This starts when the first call queues and stops when all direct agent calls leave the queue (answered, abandoned, or sent to coverage).

The originating agent hears normal call progress tones and ringback. If the originating agent drops from the call, the caller hears call progress tones and ringback. A forced first announcements will not be heard by the originating agent or caller.

Direct agent calls are queued and served in a first-in first-out order, before any non-DAC. Therefore, when an agent becomes available, the switch first checks for any direct agent calls before serving normal ACD calls in queue.

The voice terminal display for the receiving agent, before a transfer is complete, shows the originating agent's name and number. The voice terminal display for the receiving agent, after the transfer is complete, shows the caller identification (CPN/BN or trunk group name for external calls, and name/number for internal calls) and the original split or VDN name.

Direct agent calls follow the receiving agent's coverage and call forwarding, if activated. Once the call goes to coverage or is forwarded, the call is no longer treated as a DAC. CMS is informed that the call has been forwarded.

Answering a Direct Agent Call

The receiving agent answers a direct agent call by becoming available in the split with which the direct agent call is associated. While on a direct agent call, the agent becomes unavailable to subsequent ACD calls.

If the receiving agent logs off by unplugging the headset, the agent may still answer a direct agent call in queue by logging back in and becoming available.

Considerations

A maximum of eight CallVisor ASAI links may be assigned.

Information from a CONVERSANT voice system (except for the dialed number) cannot be carried with the call and displayed on a voice terminal. For example, digits collected in a CONVERSANT voice system adjunct cannot be passed to the switch for display.

CallVisor ASAI and BX.25 CPN/BN-ANI are not supported simultaneously.

Interactions

The following features and functions interact with the ICM feature:

- Automatic Answer

  DACs to agents assigned automatic answer will receive single zip tone answer.

- Queue Status Indications

  DAC are not included in the number of calls waiting and oldest call waiting on the queue status indication for the split.

- Call Coverage

  If the split associated with a direct-agent call has call forwarding or night service activated for the split, the direct-agent call is forwarded. If the priority calling option is requested, the direct-agent call forwards with priority ringing at the night service destination.

After a DAC call successfully terminates to a split, if the destination agent has Call Forwarding or Send All Calls activated, then the DAC calls will be forwarded. If the priority calling option is requested, then the DAC call will forward with priority ringing at the call forwarded destination, but will not forward to the covering point in the case of Send All Calls when the priority calling option is requested.

Direct-agent calls follow the destination agent’s coverage path. If the priority calling option was requested, the DAC follows the standard priority call rules for coverage, meaning the call will not go to coverage. Calls (either regular ACD or direct-agent) in queue will remain in queue until the caller abandons or agent answers. Calls follow the agents internal or external coverage path based on the origin of the call, not the internal/external status of the adjunct providing routing instructions.

■ Call Forwarding

New DAC will forward if the agent activates Call Forwarding. Direct Agent Calls already in queue prior to Call Forwarding activation will remain in queue.

■ Call Vectoring

Call Vectoring is used in conjunction with CallVisor ASAI capabilities. The `adjunct routing` vector command is required. For adjunct routing, if the call queues to a split or the call leaves vector processing, a route end request is sent to the CONVERSANT voice system.

Direct agent calls cannot go through vectors.

■ Call Prompting

Digits collected by the Call Prompting feature become part of the current call information that is passed to a CONVERSANT voice system adjunct.

■ Call Waiting

Call waiting tone is used to notify single-line users that a DAC is waiting, whether the call waiting feature is optioned or not.

■ DCS

DAC cannot be made over a DCS link. If the receiving agent is not an internal extension, DAC is denied.

■ Night Service

DACs will go to Night Service if Night Service is activated for the receiving agent’s split.

■ Priority Calling

CallVisor ASAI capabilities permit both Priority Calling and DAC for the same call. Priority DAC will not go to coverage.

■ Send All Calls

If an agent activates send all calls, *new* DACs will immediately go to the agent's coverage. DACs already in queue prior to Send All Calls activation will remain in queue until the coverage ringing timeout occurs.

Administration

ICM is administered by the System Manager.

The ACD feature must be administered as described in the ACD feature description elsewhere in this manual.

If the Call Vectoring and/or Call Prompting features are to be used, these features must be administered as described in the Call Vectoring and Call Prompting feature descriptions elsewhere in this manual.

In order to make or receive direct agent calls, an agent must be assigned a COR that allows DAC.

Direct agent calls wait in split queues. Split queues must be properly sized.

Hardware and Software Requirements

The only additional hardware requirements for ICM are those for the CallVisor ASAI. These requirements are in the following paragraphs.

The system supports TN744C-Tone Detector/Call Classifier, TN748C-Tone Detector, TN420C-Tone Detector, or TN2182-Tone Clock/Detector/Generator for use as tone detectors. TN420C, TN744C, and TN2182 support A-law. With respect to CallVisor ASAI features, the TN744C is required for those customers who desire switch call classification.

Each port on the eight port TN744C acts as a touch-tone receiver or call classifier. Each call classifier port is capable of detecting tones as well as Special Information Tones.

This tone detection will work only if the public network provides similar tones to those used in the US.

Each CallVisor ASAI BRI Interface Link requires a port on a TN556 ISDN-BRI circuit pack.

CallVisor ASAI Interface software is required. A TN778 Packet Control board and, in a multiple port network design, TN570 expansion interface are required.

Individual Attendant Access (IAA)

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows users to access a specific attendant console. Each attendant console can be assigned an individual extension number.

A user can access an individual attendant by simply lifting the handset and dialing the extension number assigned to the desired attendant. An individual attendant extension number can also be assigned to users' abbreviated dialing button for fast access to the specific attendant.

Individual attendants can be accessed by voice terminal users, incoming trunks, Remote Access, and other attendants. A specific attendant, when called, can extend the call to another trunk or extension.

Each individual attendant has a queue that allows incoming calls to wait. For G3iV1, this individual attendant queue has priority over all other attendant seeking calls. For G3r and all of Version 2 and later releases, the Individual Attendant Access call is placed in the Attendant queue according to the priority assigned in Attendant Priority Queueing.

Whenever a call is in an individual attendant's queue, the top lamp of the Forced Release button (basic console) or the Personal lamp (enhanced console) lights to indicate this condition. Call Waiting tones are provided only on calls to the attendant group and are not provided for waiting individual attendant calls.

An individual attendant can be a part of a hunt group. The hunt group can be a DDC group or a UCD group. Calls to individual attendants and calls to the attendant group have priority over hunt group calls to an individual attendant.

Any call made from an attendant console which is assigned an individual extension is considered to be made from the individual attendant, not the attendant group.

Considerations

With Individual Attendant Access, attendant consoles can become more flexible by assigning each one an individual extension number. An individual attendant extension allows an attendant to use features that an attendant group cannot use; for example, individual attendant extensions can be a member of a DDC or

UCD group. An individual attendant can also be accessed when the Centralized Attendant Service feature is in effect. Another advantage is that each individual attendant extension can have its own Class of Restriction and Class of Service.

The Position Available lamp on the attendant console only indicates whether or not attendant group calls can be accepted. It does not indicate whether or not individual attendant calls can be accepted.

Each attendant console has one position busy button. When the lamp associated with this button is lighted, the attendant does not receive attendant group calls but can still receive individual attendant calls.

Since hunt groups have better queuing and make busy features than individual extensions, it may be desirable to assign an individual attendant as the only member of a hunt group. This way the individual attendant could receive calls as a hunt group member for more efficient handling of calls.

Interactions

The Individual Attendant Access call is placed in the Attendant Queue according to the priority assigned in Attendant Priority Queuing (for G3iV1 and all G3V2 and later releases).

The following features interact with the Individual Attendant Access feature.

- Abbreviated Dialing

  Individual attendant extensions can be in Abbreviated Dialing lists. Individual attendants, however, cannot have their own Abbreviated Dialing lists.

- Attendant Display

  For calls to or from individual attendants, individual attendant names (when specified) are displayed instead of the individual attendant extensions.

- Bridged Extension

  Individual attendant extensions cannot be assigned to a bridged call appearance.

- Busy Verification of Terminals and Trunks

  An individual attendant extension cannot be busy verified.

- Call Coverage

  Individual attendant extensions can be points in a coverage path but cannot be a member of a coverage answer group.

- Call Park

  Individual attendants can park calls on their own extension or another individual attendant extension.

- Call Pickup

  Individual attendant extensions cannot be in Call Pickup groups.

- CAS

  Individual attendants can be accessed when CAS is in effect.

- COR and COS

  Each individual attendant extension has its own COR and COS. However, it is recommended that an individual attendant and the group with which he or she is associated be assigned the same COR.

- DDC and UCD

  Individual attendant extensions can be assigned to DDC and UCD groups. Unlike voice terminal users, individual attendants can answer DDC and UCD calls as long as there is an idle call appearance and no other DDC or UCD call is on the console.

- Facility Busy Indication

  An individual attendant extension can be stored in a Facility Busy Indication button.

- Integrated Directory

  The names and extensions of the individual attendants are stored in the directory associated with this feature.

- LWC

  A message from an attendant indicates whether it is from the attendant group or whether it is from an attendant which has an individual extension.

- Night Service — Night Console Service

  Activation and deactivation of this feature affects only calls to the attendant group. Calls to individual attendant extensions are allowed when night service is active. A night-only attendant console with an individual extension can receive individual attendant calls when night service is not active.

- Privacy — Attendant Lockout

  This feature applies only to attendant group calls. Individual attendant calls are not affected.

- Voice Terminal Display

  For calls from individual attendants, individual attendant names (when specified) are displayed instead of the individual attendant extensions.

Administration

Individual Attendant Access is administered on a per-attendant console basis by the System Manager. The following items require administration for each attendant console:

- Extension number
- Name
- COR
- COS

Hardware and Software Requirements

No additional hardware or software is required.

Information System Network (ISN) Interface

Feature Availability

This feature is available with all Generic 3 releases.

Description

The AT&T ISN is a packet-switched local area network that links mainframe computers, minicomputers, word processors, storage devices, personal computers, printers, and communications processors into a single system.

The ISN is a packet-switched local area network. This local area network is made up of one or more modular data-only digital communications switches.

The interface between the system and the ISN is via a Data Line Port in conjunction with an Asynchronous Data Unit (ADU). Either a Modular Processor Data Module (MPDM), a 7400A data module may be used instead of the Data Line Port and ADU, although the latter is more economical. The MPDM or ADU connects to an AIM on the Packet Controller or Terminal Concentrator (see Figure 3-19). This interface allows the system and the ISN to share data capabilities. (The 7400A connections are the same as shown below for the MPDM.)

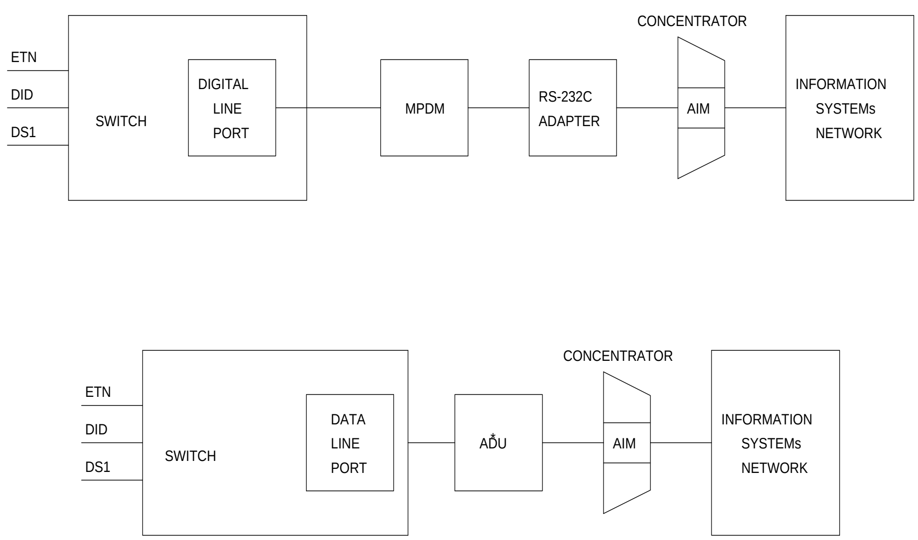


Figure 3-19. System-to-ISN Connectivity

Data is transferred between the system and the ISN on one-way trunks (either incoming or outgoing). Each data line port is administered for a specific data rate, which can be any of the common asynchronous data rates ranging from low to 19,200 bps.

Considerations

Connectivity between ISN and the system provides the following major benefits:

- Users on ISN may (in addition to having access to other endpoints directly connected to ISN) have access to any endpoint connected to the system or addressable from the system.
- Users who either connect to or have access to the system may also access endpoints connected to ISN.

Since the ISN switches are modular, the local area data communications network can be designed so that it is both versatile and cost-effective. A single packet controller can be configured to support from 40 to 1,920 data ports.

Interactions

The following features interact with the ISN Interface feature.

- Abbreviated Dialing

  Outgoing lines cannot use Abbreviated Dialing.

- Automatic Circuit Assurance

  Automatic Circuit Assurance is not provided for data line port links to or from the ISN.

- Data Call Setup

  Data Terminal (Keyboard) Dialing is used to access ISN endpoints. A data call to an ISN data endpoint from a system digital data endpoint requires two-stage dialing. A user must first dial the extension assigned to the outgoing ISN group, and then interact with ISN and enter the second address (data endpoint).

- Data Hotline

  Outgoing lines cannot use hot line calling.

- Modem Pooling

  If an analog data endpoint is used in an ISN connection, and a conversion resource is needed, the system will obtain a conversion resource from the appropriate pool.

- System Measurements

  No traffic measurements are made on data line port links to or from the ISN.

- Uniform Call Distribution

  Outgoing lines should be members of a UCD group. This way, the system automatically selects an idle port when a user tries to access the ISN.

Administration

Data module extensions used to access ISN must be administered by the System Manager as data lines connected to the ISN. The System Manager can then administer the other options required for each data line. These options include:

- Keyboard Dialing — If the line is incoming (to the system), Keyboard Dialing should be enabled so that the system can be accessed by the ISN. If the line is outgoing (to the ISN), Keyboard Dialing should be disabled.
- Configuration — This option should be disabled on both incoming and outgoing lines to prevent the ISN from changing the data line configuration.
- Busy Out — This option should be enabled for outgoing lines so that a member of the outgoing ISN group can be ‘‘busied-out’’ and let calls go through another member of the group.
- Speeds — Data speeds should be selected according to individual needs, and should be the same as those at connecting ISN ports. Only one speed should be assigned to each data line port.
- Autoadjust — This option is not needed with the ISN, and should be disabled on incoming lines. This option can only be set if Keyboard Dialing is enabled.
- Permit Mismatch — This option should be disabled on both incoming and outgoing data lines.
- Disconnect — The disconnect sequence should be administered according to the characteristics of the device. This option can only be set if Keyboard Dialing is enabled.
- Parity — This option should be administered as even. This option can only be set if Keyboard Dialing is enabled.
- Dial Echoing — This option should be disabled so that characters are not echoed back to the ISN. This option can only be set if Keyboard Dialing is enabled.
- Answer Text — This option should be disabled, and can only be set if Keyboard Dialing is enabled.

- Connected Indication — This option should be disabled, and can only be set if Keyboard Dialing is enabled.
- COR — Outgoing lines should be origination restricted. Incoming lines should be termination restricted.

Hardware and Software Requirements

One TN726 Data Line circuit pack is required for each eight ISN interfaces. No additional software is required.

Integrated Directory

Feature Availability

Integrated Directory is available with all Generic 3 releases.

Description

Allows internal system users with display-equipped terminals to access the system database, use the touch-tone buttons to key in a name, and retrieve an extension number from the system directory. The directory contains an alphanumeric listing of the names and extension numbers assigned to all voice terminals administered in the system.

The Integrated Directory feature can be accessed by display-equipped voice terminal users or Attendants with an assigned Integrated Directory button.

The names in the directory are those administered by the System Manager on the “Individual Voice Terminal” forms. Names cannot exceed 15 characters (including spaces and commas) and can be entered in one of the following three formats.

- Last name, comma, first name, space, then middle name or initial, if desired. For example, the following entries are acceptable:
  - — Jones,Betty Ann
  - — Smith,A E
  - — Thomas,John J
  - — Abbott,Lynn
- First name, space, second name or initial, and then last name. For example, the following entries are acceptable:
  - — Betty Ann Jones
  - — A E Smith
  - — John J Thomas
  - — Lynn Abbott
- A single entry is also acceptable:
  - — Cafeteria
  - — 1J409
  - — 2F816
  - — Purchasing

The following is an example of a typical Integrated Directory database:

— 1J409
— Abbott,Lynn A
— Brown,Kent J
— Cafeteria
— Carr,Danny
— Carter,Ann
— 2F816
— Purchasing
— Barbara Quincey
— Roberson,Don T
— William Ruoff
— Smith,A E
— Streck,R T

The touch-tone buttons are used to key in the numbers and letters labeled on them. The following exceptions apply:

- **7** (PRS) is also used for a Q.
- **9** (WXY) is also used for a Z.
- ***** is used for a space or comma.
- # is not used.

To activate the Integrated Directory feature, the user presses the Integrated Directory button. This puts the voice terminal in the Integrated Directory mode and turns off the tones normally generated when a touch-tone button is pressed. The touch-tone buttons are now used exclusively for keying in names and not for dialing.

After the Integrated Directory button is pressed, the alphanumeric display shows DIRECTORY — PLEASE ENTER NAME. Names are always keyed in the following order: last name, comma, and then first name or initial. When searching for a single entry, the letters or numbers are keyed in order. Several letters might be needed to get the correct entry.

When a button is pressed, the display shows the first name that matches the first letter on the button. For example, if a user is searching for the name Ann Carter and presses **2** to key in the letter C, the display might show Abbott,Lynn A and an extension number. (**2** matches A before it matches C.) If the user presses **2** again to key in the letter A, the display stays the same. (Again, AB is matched before CA.) If the user now presses **7** to key in an R, the display might show Carr,Danny and an extension number.

At this point, the user can press **8** to key in the letter T or can press the Next Message button on the alphanumeric display. Pressing Next Message displays the next name in the directory and, in this case, might be Ann Carter.

When the desired name and extension number are displayed, the user can automatically place a call to that person by pressing the Call Display button.

If a name is entered but not found in the directory, the display shows NO MATCH — TRY AGAIN. You can then enter another name. To search for another name, the user presses the Integrated Directory button again, and the feature is reactivated.

To exit the Integrated Directory mode, the user presses one of the other mode buttons assigned to the alphanumeric display module; for example, the Normal mode button.

Considerations

With Integrated Directory, users spend less time looking up names and extension numbers. Instead of searching through lists or directories, a user simply keys in the desired name and the display shows the name and extension number. Less dialing time is also required if a Call Display button is provided. When the desired extension is displayed, the user just presses the Call Display button to automatically place the call.

A maximum number of users can activate Integrated Directory at the same time. If more than this maximum number of users try to activate the feature at the same time, the Integrated Directory button lights and the display shows `Directory unavailable –– Try Later.`

The entire directory cannot be searched by pressing **2**. Pressing **2** and then continually pressing **Next Message** displays, one by one, all entries beginning with A, B, C, and 2. If all entries have been displayed and **Next Message** is pressed again, the display repeats from the first entry in the listing associated with **2**.

When the voice terminal is in the Integrated Directory mode, it cannot be used to make calls or access features by dial code. It can, however, still be used to activate other features or to place calls if dialing is not required. Also, a user can enter the Integrated Directory mode while active on a call, and calls can be received when the Integrated Directory mode is active.

The set of characters allowed in the Integrated Directory database are the alphanumeric characters (A-Za-z) and digits (0-9) as well as space ( ) and comma (,) which are used as delimiters. In addition, the following special characters are allowed: hyphen (-), apostrophe (‘), period (.), slash (/), and ampersand (&.) These special characters, though considered legal characters, are not entered into the Integrated Directory. Instead, a period (.) is replaced by a space ( ). Apostrophe (‘), hyphen (-), slash (/), and ampersand (&) are ignored.

For G3r, if a name of a station or data module begins or ends with a comma (,) or contains more than one comma (,,), the directory search for that name fails.

If a character outside the allowed set is entered as the name of a station or data module, the directory search for that name fails.

NOTE:
The “/” and “?” characters are punctuation marks and should *not* be used to represent information. The “+” or “%” characters may be used to represent information.

Interactions

The following features interact with the Integrated Directory feature.

- Attendant Display and Voice Terminal Display

  If prefixed extensions are used in the system’s dial plan, the prefix is not displayed when the extension is displayed. The Call Display button can be used to dial prefixed extensions because the system dials the prefix, even though it is not displayed.

- Merlin®/System 25 Voice Terminal Support — 731xH Series Support

  The Merlin/System 25 7309H, 7313H, 7314H, 7315H, 7316H, and 7317H voice terminals do not support Integrated Directory displays.

- Touch-Tone Dialing

  Call origination and feature access by dial code is not allowed when the Integrated Directory feature is active.

Administration

Integrated Directory is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Display Module
- Directory Button
- Call Display Button
- Messaging Cartridge (for 7404D)
- Next button

Hardware and Software Requirements

No additional hardware or software is required.

Integrated Services Digital Network (ISDN) — Basic Rate Interface (BRI)

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows connection of the system to equipment or endpoints that support an Integrated Services Digital Network (ISDN) by using a standard ISDN frame format called the Basic Rate Interface (BRI).

An ISDN provides end-to-end digital connectivity and uses a high-speed interface which provides service-independent access to switched services. Through internationally accepted standard interfaces, an ISDN provides circuit or packet-switched connectivity within a network and can link to other ISDN supported interfaces to provide national and international digital connectivity. Two types of ISDN interfaces are currently defined: the PRI and the BRI. This description focuses on ISDN-BRI.

The ISDN-BRI is a 192 kbps interface that carries two 64 kbps B-channels and one 16-kbps D-channel. Each B-channel supports voice and data, while the D-channel transports data, signaling, and other bits for framing.

NOTE:

- Although ISDN-BRI can support data transmission on the D-channel, DEFINITY system Generic 3 does not support this capability. DEFINITY system Generic 3 only provides signaling on the D-channel.
- ISDN-BRI is only available on DEFINITY switches that support 5-law companding. See the *DEFINITY Communications System Generic 3 System Description and Specifications*, 555-230-206, for further details.
- The word ‘‘endpoint’’ is used whenever statements apply to BRI voice terminals, BRI data modules, and integrated BRI voice/data terminals.

The DEFINITY switch supports the following across the ISDN-BRI interface to an endpoint:

— A single voice call

— A single data call

— A voice call and a data call simultaneously

— Two simultaneous data calls

ISDN-BRI Endpoint Configurations

There are two possible configurations with ISDN-BRI:

- Point-to-point — Only one endpoint connected to a BRI port
- Multipoint — Multiple endpoints connected to a BRI port. This configuration is also referred to as ‘‘passive bus configuration.’’

Because the DEFINITY system BRI provides non-blocking voice and data services, a maximum of two endpoints are supported on one BRI port in a multipoint configuration.

The DEFINITY switch dynamically allocates the two B-channels on a BRI interface to handle voice and data requests. Because there are two B-channels, only two simultaneous service requests can be granted at any time on a BRI port to provide non-blocking service (either through point-to-point or multipoint configurations).

When one endpoint is capable of providing two service requests (such as an integrated voice/data endpoint or an endpoint that supports two simultaneous %B-channel data calls), the endpoint must be configured as point-to-point because both B-channels can potentially be used simultaneously by the two services. You can have the following endpoint types in the point-to-point configuration:

- One voice-only endpoint
- One stand-alone data endpoint
- One integrated voice/data endpoint

NOTE:
Even though an integrated voice/data endpoint supports two service requests (that is, both voice and data), the integrated endpoint is not considered to be in multipoint configuration because it is only one endpoint.

If two endpoints are each capable of providing only one service request, then they can be administered on the same BRI port. Doing so provides a multipoint configuration. In this case, both B-channels can potentially be used simultaneously by two service requests and, since each BRI port provides two B-channels, no additional endpoints can be administered on this BRI port. You can have the following endpoint types in the multipoint configuration:

- Two voice-only endpoints
- Two stand-alone data endpoints
- One voice-only endpoint and one stand-alone data endpoint

NOTE:
An endpoint that supports two simultaneous B-channel data calls is not supported in a multipoint configuration.

Terminal Equipment Identifier (TEI)

The terminal equipment identifier (TEI) is used to set up communication between the switch and an endpoint. The DEFINITY system supports two types of TEIs: fixed and automatic. A fixed TEI endpoint supports one fixed TEI value (0 to 63), which is encoded into the terminal equipment, and the fixed TEI initialization procedure. When administering a fixed TEI endpoint, you must assign the endpoint's fixed TEI value to the "Station" or "Data Module" forms for that endpoint. If the endpoint's fixed TEI value differs from the TEI assigned to the "Station" or "Data Module" forms for that endpoint, no communication is established between the switch and the endpoint. As a result, the endpoint is incapable of providing services. Normally, the manufacturer specifies the fixed TEI value encoded into the terminal or provides procedures for modifying the fixed TEI value.

An automatic TEI endpoint supports automatic TEI initialization procedures and receives a TEI from the system during initialization. With automatic TEI endpoints, you are not entering any TEI values to the "Station" or "Data Module" forms. Only the automatic TEI endpoints are permitted to be used in multipoint configurations. Currently, all supported BRI endpoints are automatic TEI endpoints.

Service Profile Identifier (SPID)

When more than one endpoint is connected to a BRI port (for example, a multipoint configuration), the switch uses the Service Profile Identifier (SPID) to associate endpoints with the administered station or data module extensions. The SPID enables the switch to differentiate between the endpoints connected to the same BRI port.

You must administer the SPID on the "Station" or "Data Module" forms, and then program the SPID in the BRI endpoint using the procedure in the endpoint's users' manual. During initialization, the endpoint sends the SPID to the switch. The SPID administered on the "Station" or "Data Module Administration" forms must match the SPID which is programmed into the endpoint. If the SPID on the "Station" or "Data Module Administration" forms does not match the SPID programmed into the endpoint, the system restricts service to that endpoint.

SPID administration and programming are required for a multipoint configuration. However, SPID administration is optional in a point-to-point configuration because there is only one endpoint connected to the BRI port. If the SPID is administered in a point-to-point configuration, it must match the SPID programmed into the endpoint. If the SPID is not administered, the switch uses the port to associate the endpoint to the administered station or data module extension.

NOTE:
The BRI version of the PC Interface feature, the PC/ISDN Platform, is a non-initializing terminal and does not require a SPID. In fact, in the stand-alone configuration, this type of terminal cannot be assigned a SPID.

ISDN-BRI Voice/Data Terminal Equipment

Voice transmission on ISDN-BRI is provided by the 7505, 7506, 7507, 8503T, 8510, and 8520 voice terminals. All tests and services available to DCP users are also available to BRI users.

Data transmission on ISDN-BRI is provided by the 7500 Data Module (or a compatible stand-alone data module) and the Asynchronous Data Module (ADM). The 7500 Data Module is a stand-alone unit that supports asynchronous or synchronous DCE and asynchronous DTE:

- In asynchronous mode, the 7500 supports packet- or circuit-switched data communications, and can be controlled via the front panel or the keyboard of a connected terminal.
- In synchronous mode, the 7500 supports circuit-switched or nailed-up data communications, requires either the Multi-purpose Enhancement Board or the High-Speed Synchronous Enhancement Board, and can only be controlled via the front panel.

The ADM may be used with asynchronous DTE as a data stand for 7500-series BRI voice terminals. Consisting of a board located inside the BRI voice terminal, the ADM allows the transmission of integrated voice and data through one voice terminal. (Separate extension numbers are used for the voice and data calls.) The ADM supports the Hayes command set for compatibility with PC communications packages.

The PC/ISDN Platform provides both voice and data transmission and is part of the AT&T PC Interface package, Group 3. (Please see the "PC Interface" feature.) The PC/ISDN Platform can be used in a stand-alone (no associated voice terminal) configuration. When used this way or with a handset or headset, rather than one of the 7500 series voice terminals, the PC/ISDN Platform is a non-initializing terminal that does not support MIMs (Management Information Messages). This means that these ‘‘stations’’ do not go through an initialization procedure with the switch. It also means that they do not require a SPID, and that they do not initiate or respond to MIMs. Even when used with a 7500 series voice terminal, the endpoint (combined PC and phone) should still be administered as a non-initializing terminal that does not support MIMs.

NOTE:
When more than one PC/ISDN interface card is used on the same PC, each separate card provides a separate interface and is translated on the switch with its own ELL (Equipment Line Location) as a separate station.

Endpoint Initialization

To be fully operative, BRI endpoints must successfully complete endpoint initialization procedures. These procedures are usually done at installation time or as part of reconfiguration.

Multipoint Configurations on BRI Ports

In a passive bus multipoint configuration, the system supports two BRI endpoints per port, thus doubling the capacity of the BRI circuit pack. When changing the configuration of a BRI from point-to-point to multipoint, the original endpoint need not be reinitialized. However, only endpoints that support SPID initialization can be administered in a multipoint configuration.

Exchange of User Information

The BRI protocol provides the users with the capability of exchanging up to 128 octets of user information end-to-end. The information is passed in the

User-to-User Information IEs to the receiving endpoint without being interpreted by the switch. However, there are some limitations to the exchange of User Information IEs.

ISDN-BRI Data Service

Basic Digit Dialing

Regular digit dialing is provided through the ADM. Digits from 0 to 9, ''*'' and ''#'' can be entered. This feature can be used by the user either from the set keypad or from the EIA terminal interface.

Default Dialing

Default Dialing is also an enhancement to the user dialing capabilities of the Data Call Setup feature. By either typing a d followed by RETURN or pressing the data button twice, if a default address is administered, the switch terminates the call to the default address. If no default dialing has been administered, the call is disconnected in less than one second. This feature and the Data Hotline feature are mutually exclusive.

Data Hotline

Data Hotline is a security feature that allows a user to enter a `Dial` command, with no address specified, followed by a RETURN. The switch terminates the call to a preadministered hotline destination. If a user enters an address, either intentionally or unintentionally, the call processing discards the address string received for the hotline endpoint. The call processing automatically routes the call just as if the hotline destination address had been entered by the user. This service does not impose any restriction on incoming calls received at the endpoint.

NOTE:
This feature and the Default Dialing feature are mutually exclusive.

Administered Connections

An Administered Connection is an end-to-end connection between two access endpoints or data endpoints that is automatically established by the system whenever the system is restarted or the Administered Connection is administered and the Administered Connection is due to be active. The attributes of these connections are user-defined and administered on the “Administered Connection” form via the G3-MT.

Once the ADM has been administered as one endpoint of an administered connection, the system waits for the scheduled time to set up the connection. At the scheduled time, the system establishes the connection and maintains it for the specified length of time. Once the call is accepted, the set enters into the continuous mode for the specified length of time. If the switch is rebooted during the continuous connection, the connection reinitiates the call setup. At any time that the connection drops (for example, disconnected cabling), the switch re-initiates the call setup.

Call Request

The DEFINITY system Generic 3 call processing handles all various BRI Bearer data call requests that are presently defined. Some capabilities that are not supported by AT&T terminals may be provided by a non-AT&T terminal. The switch completes most call requests. For those capabilities the switch does not support, a proper cause value is sent back to the terminal.

Cause Value

BRI stations receive a cause or reason code that identifies why the call is being cleared. The BRI data modules convert certain cause values to text messages and displays them for the user.

Considerations

The system supports up to 60 simultaneous voice calls on a BRI since two 7507 terminals (that support 30 call appearances/bridged appearances each) can be administered on a BRI port.

The following features are not provided to BRI users:

- Since the ISDN-BRI protocol requires that the Bearer Capability must be specified at the time of sending the SETUP message (and cannot be changed during the call), the following data functions are not provided to the BRI voice users:
  - One button voice call setup transfer to data
  - One button data call setup transfer to voice
  - Preindication of a data call
  - Voice call transfer to data and data call transfer to voice

These functions require a change in the Bearer Capability after the establishment of the call, which is currently not allowed by the BRI protocol.

- Features that use the switchhook and Recall button (for example, Call Waiting and Analog Conference/Transfer/Hold/Drop) are applicable to analog voice terminals only

The DEFINITY system supports BRI data endpoints that perform layer 2 disconnect. BRI data endpoints that support layer 2 disconnect are administered as non-Management Information Message (MIM) supporting data only endpoints.

The DEFINITY system supports non-AT&T BRI voice and data terminals that perform enbloc sending using any one of the following formats:

- A SETUP message with all the digits in the Keypad IE and a Sending Complete IE.
- A SETUP message with all the digits in the Called Party Number IE and a Sending Complete IE.
- A SETUP message with all the digits in the Called Party Number IE and no Sending Complete IE.

If you are using a 7506D or 7507D to make calls that require additional digits, you can place a comma in the dial sequence after you receive a second dial tone or after the call has been set up. The comma is used to separate the called number from subsequent information.

Interactions

The following features interact with the ISDN-BRI services:

- Data Button

  Besides the call appearance and feature function buttons, BRI voice/data terminals have a fixed, dedicated data button (button 7 on the 7505D and 7506D voice terminals, and button 31 on the 7507D voice terminal) that is used for data call setup. In general, feature function buttons such as Call Forwarding or Send All Calls buttons are always associated with voice features, and cannot be used in conjunction with the data button. For example, the user cannot activate call forwarding for the associated data endpoint by using the data button followed by the Call Forwarding button and the designated extension.

- Interworking

  The same off-premises call types are permitted as for DCP, with the exception of voice to data and data to voice transfer.

- Modem Pooling

  The Modem Pooling feature provides the necessary protocol conversion between Mode 2 digital data endpoints and analog data facilities. A modem pool resource needs to be inserted by call processing during call setup for both call origination and call answering. This resource translates data between DMI Mode 2 protocol used by BRI data endpoints and the modulated signal used by the modem.

- Pre-Selection

  When an ISDN-BRI station assigned with the “Select Last Used Appearance?” field of the “Station” form set to yes completes a transfer while off-hook using the handset, the user will be left hearing a dial tone on the last-used appearance, rather than the silence heard in the same situation by an user of other station types.

- Voice Terminal Display

  BRI terminals take control of the display. For example, when the user is in dialing state (BRI terminal is in the Overlap Sending state) any display information sent to the terminal from the switch is buffered until the state changes and is displayed when the state changes.

  The 7506 BRI voice terminal, which has a 2-line 24-character display, splits a message when it recognizes a blank closest to the 24th character. This is left to the discretion of the terminal. As a result, the switch has no control over it.

- Busy Tone Forward Disconnect

  BRI terminals require that the Busy Tone Forward Disconnect (BTFD) feature be turned off. With BTFD off, an agent (with Malicious Call Trace) is able to change states after a call.

Administration

BRI Voice/Data

Administration of BRI voice terminals requires all the fields associated with the standard DCP station administration. Additional fields are used to enter the following information:

- TEI information: If the BRI terminal supports a fixed TEI value, it has to be entered at the time of station administration. The allowed values are 0 through 63. There are two fields: “Fixed TEI” and “TEI Value”. If the answer to the first field is yes, the second field shows up where the TEI value is administered. The TEI value on the “Administration” form must match the value supported by the terminal.
- MIM support: This is an “Administrable” field on the “BRI Station” form. If the answer to this field is ‘‘yes,’’ the following two fields need to be filled in:

1. Endpoint Initialization: If the BRI terminal supports endpoint initialization, the administrator has to enter the SPID value. The default value is the extension; however, the value can be changed at the time of administration. The SPID can be up to 10 digits, and uniquely identifies the terminal on the BRI. The SPID on the "Administration" form and the SPID programmed into the endpoint must be the same. (Refer to the terminal's user manual in order to change the terminal's SPID.) All SPIDs must be different for each endpoint on the same port. All SPIDs must be different from the service SPID, which is administered on the "System Maintenance" form.
2. MIM Maintenance / Management support: This is another BRI specific field that indicates if the terminal supports other maintenance and management messages.

For multipoint (passive bus) environment, the system administration checks the number of B-channels used for a port. Administration denies any attempt which might cause call blocking by restricting the number of endpoints on a port.

NOTE:
Separate extensions numbers are used for voice and data calls to the same endpoint.

Any PC/ISDN Platform configuration (stand-alone, with handset/headset, or with voice terminal) should be administered with MIM support set to ''no''. Other administrable features depend on the PC software application (if the application supports data transmission, the ''Data Module'' field must be set to ''yes'').

BRI Data

The 7500 Data Module is administered through the "Data Module Administration" form. The ADM is administered through the "Station Administration" form, using the data module administration page. In addition to the fields used by the DCP endpoints, the following new fields are used for BRI data module administration:

- Default Duplex: Full/Half (default Full)
- Default Mode: Synchronous/Asynchronous (default Asynchronous)
- Default Speed: 1200, 2400, 4800, 9600, 19200, 56000, and 64000 (default 1200)

  (In synchronous mode, the speed of the 7500 data module may be set to 56,000 or 64,000 bps.)
- Default Data Application: mode 0, mode 1, mode 2 sync, mode 2 async, and mode 3/2 adaptable

Default Duplex, Default Mode, and Default Speed values are used for initializing data module default attributes. The defaults are required for modem pooling conversion resource insertion when the endpoint does not support MIM query capability. If you are using supported endpoints (7500 and ADM), you should not change the default values. Changing the default values with the MIM Maintenance/Management support option as ''y'' has no effect on modem pooling. These endpoints support the MIM query capability, which enables the switch to query the endpoint when a call arrives.

Default Data Application specifies the default data protocol to be used for originating data calls if mode is not specified with the calling parameters. This mode is also used for Administered Connections and for terminating trunk calls that do not have bearer capability specified.

Data modules that support two simultaneous data calls should be administered as a 7500 and the MIM Maintenance/Management support option should be set to ''n.''

NOTE:
Two simultaneous data calls to a single endpoint use the same extension number.

Hardware Requirements

BRI services require the following hardware:

- The TN778 Packet Control circuit pack. This circuit pack provides the interface to the LAN (packet) bus on G3i (but not G3r) for establishing the signaling connectivity.
- The TN556 BRI port circuit pack, which is the Basic Rate Line circuit pack. Each BRI port board can support 12 line interfaces, each operating at 192 kbps.
- A TN2198 two-wire BRI port circuit pack can be used in place of the TN556. In this case an NT1 is also required.
- ISDN-BRI Type B and Type D Terminal Management S/T interface terminals.
- The AT&T ISDN 7505, 7506, 7507, 8503T, 8510, and 8520 voice terminals.
- The 7500 Data Module and the ADM. The ADM is supported by the AT&T ISDN 7505, 7506, and 7507 voice terminals with firmware version FP2.0 or later.
- The PC/ISDN Interface card used in conjunction with a standard PC and one to four voice terminals. See the *PC/ISDN Platform Installation and Reference* manual, 555-016-102, for additional information. Also see the PC Interface feature described in this book.

Integrated Services Digital Network (ISDN) — Primary Rate Interface

Feature Availability

ISDN-PRI is an optional feature that can be purchased with any Generic 3 releases except G3vs/G3s ABP.

Description

Allows connection of the system to an Integrated Services Digital Network (ISDN) by using a standard ISDN frame format called the Primary Rate Interface (PRI). The ISDN gives the system users access to a variety of public and private network services and facilities. The ISDN-PRI standard consists of layers 1, 2, and 3 of the Open System Interconnect (OSI) model. In ISDN-PRI, the transmission standard for layer 1 (the physical layer), is either DS1 (T1) or CEPT1 (E1).

The DS1 (T1) is a digital transmission standard that in North America carries traffic at the digital signal level-1 (DS1) rate of 1.544 Mbps. T1 facilities are also used in Japan and some Middle-Eastern countries. It consists of a 1.536 Mbps signal multiplexed with an 8 kbps framing channel. The 1.536 Mbps signal is divided into 24 channels (DS0s) of 64 kbps each, numbered 1 - 24. The ‘‘D’’ (data) channel multiplexes signaling messages for the ‘‘B’’ (bearer) channels carrying voice or data. When a D-channel is present, it occupies Channel 24.

The CEPT1 (E1) is a digital transmission standard that carries traffic at a rate of 2.048 Mbps. It is used in Europe and elsewhere. The E1 facility is divided into 32 channels (DS0s) of 64 kpbs information numbered 0-31. Channel 0 is reserved for framing and synchronization information. When a D-channel is present, it occupies channel 16.

The DEFINITY system offers several administrable protocols, each of which provides a different set of services that ISDN-PRI allows. For country 1 (U.S.A.), users can administer either the AT&T Switched Network or National ISDN-2. These protocols are discussed in detail later in this section. For other countries, the DEFINITY system provides a Global ISDN-PRI. This includes any combination of services, including (but not limited to) Basic Call, DDI, Display, and QSIG. The services available with Global ISDN-PRI are country-dependent. (Please refer to the "QSIG Global Networking" feature.

ISDN-PRI signaling in the system is supported by the TN767 (for 24 channels) or the TN464C or later version (for 24 or 32 channels) DS1/E1 Trunk circuit packs, coupled with (for all releases except G3r) the TN765 Processor Interface circuit pack. The D-channel (signaling channel) is switched through the TN765 circuit pack.

NOTE:
For G3r, the D-channel information can only use the TN464C (or later version) circuit pack; TN767 can be used for NFAS interfaces carrying only B-channels, but not for ISDN signaling information (D-channel).

With the ISDN-PRI, the system can interface with a wide range of other products including network switches, PBXs, and host computers. These products include the following:

- Public Network switches (for example, 4ESS, 5ESS, Northern Telecom DMS250, etc.)
- DEFINITY Communications System Generic 2 and System 85 R2V4.
- DEFINITY Communications System Generic 1
- DEFINITY Communications System Generic 3
- Some of the other products that adhere to the ISDN-PRI signaling protocol.

As an example of how the ISDN-PRI is used in private and public network configurations, see Figure 3-20 and Figure 3-21. As seen in these figures, the ISDN-PRI can be used to interface a PBX to a Public Switched Network, a PBX to a Host Computer, or a PBX to another PBX.

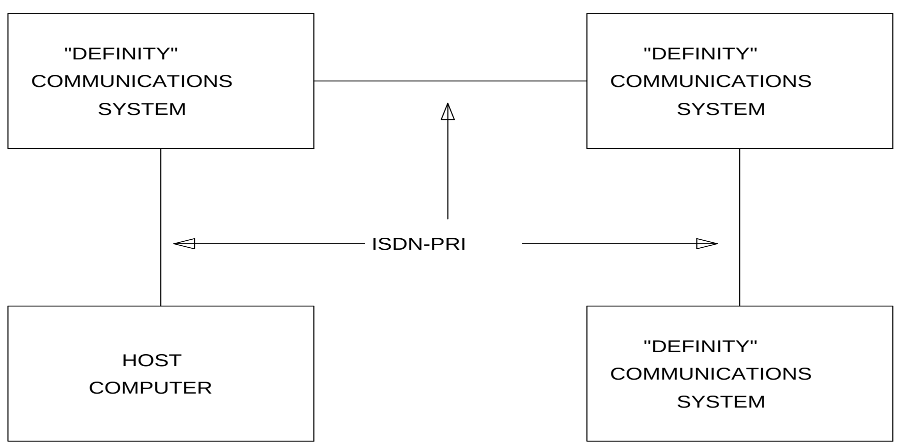


Figure 3-20. ISDN-PRI Private Network Configuration

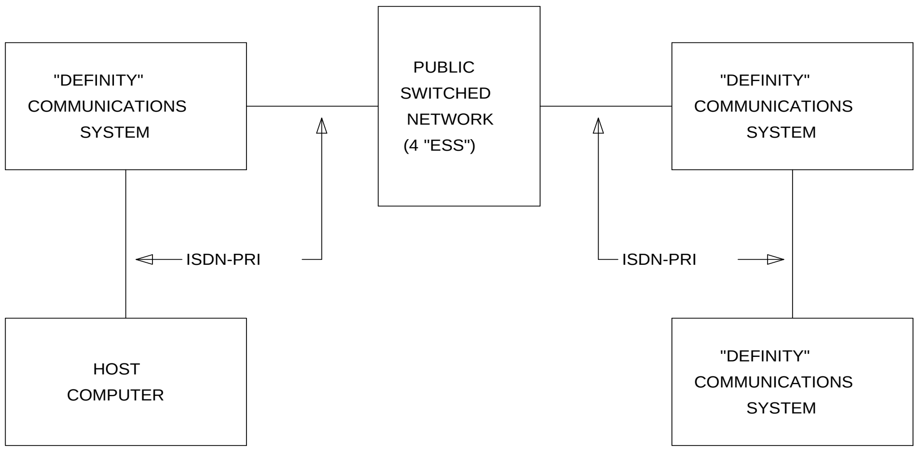


Figure 3-21. ISDN-PRI Public Network Configuration

AT&T Switched Network Protocol

The AT&T Switched Network Protocol is an ISND-PRI protocol that uses 4ESS implementation. The DEFINITY system ISDN-PRI implementation is consistent with the ITU-T Recommendation Q.931 and Q.921 for ISDN signaling. The AT&T Switched Network provides system users with the following services:

- Access to AT&T Switched Network Services (for example, SDN, MEGA, etc.)
- Call Identification Display
  - Calling Party Number (CPN)
  - Billing Number (BN)
  - Calling and Connected Number Display
  - Calling and Connected Party Name Display
- CPN/BN to Host Call Identification
- Private Network Services
- Wideband (NxDSO, H0, H11, H12) (both private and public networks)
- Call-by-Call Service Selection

These services are discussed in detail in the following subsections.

Access to AT&T Switched Network Services

ISDN-PRI provides access to AT&T Switched Network Services such as SDN (Software Defined Network), SDDN (Software Defined Data Network), MEGACOM, etc. An ISDN-PRI trunk group may be dedicated to a particular feature. Alternately, an ISDN-PRI call-by-call trunk group may provide access to several features. For a description of the ASN services accessible via ISDN-PRI (either via dedicated or call-by-call trunk groups) see the Call-by-Call Service Selection feature description elsewhere in this document.

Call Identification Display

ISDN-PRI Call Identification Display provides a transparent name/number display for all display-equipped voice terminals within an ISDN-PRI network. The feature is transparent in that the same information is provided at all ISDN-PRI facilities. Voice terminals using this feature should be digital voice terminals with a %40-character alphanumeric display. The Merlin hybrid sets with 32-character displays (7315H and 7317H) also support this feature.

ISDN-PRI Display Information is provided in addition to the normal Voice Terminal Display and Attendant Display features, when the network supports end-to-end ISDN-PRI connectivity. When both ISDN-PRI and DCS display information, or DCS display information only, are received, the switch displays the DCS display information in the DCS format. If ISDN display information is received, and no DCS display information is received, then the ISDN display information is displayed in the ISDN formats.

Two types of identification numbers are provided with the ISDN-PRI. These identification numbers may be used in the various types of displays used with the ISDN-PRI. The two types of identification numbers are as follows:

- Calling Party Number (CPN): A 0 to 15 digit DDD number associated with a specific station. When a system user makes a call that uses the ISDN, that user’s CPN is provided by the system for the ISDN. The “CPN Prefix” form is administered to create a 0-15 digit CPN from a local station number.
- Billing Number (BN): The calling party’s billing number that is provided to an inter-exchange network via Equal Access or Centralized Automatic Message Accounting (CAMA). This number is stored at either a local or network switch. If a customer is connected directly to the AT&T network, the BN is the customer’s billing number stored in that network. If the CPN is not provided on an incoming ISDN-PRI call, the system uses the BN for the station identification number.

The following types of display information are provided with the ISDN-PRI.

- Calling Party’s Number

  The calling party’s number is shown on the called party’s display. On calls generated from a DEFINITY system, the calling party’s number is a 0- to 15-digit DDD number. This number is provided only if the outgoing ISDN-PRI trunk group is administered to send the CPN and if the “CPN

Prefix” form is administered to create a CPN. For G3vs/G3sV1, G3iV1, and G3rV1, the Calling Party number, if sent, is always 10 digits long. On calls incoming to a system, the network may provide either the CPN or BN as the calling party’s number. Except for G3i-Global, dashes are inserted in the displayed number between the area code (if shown), the office code, and the local number. Extension numbers and 12-digit international numbers are shown without dashes. For G3V2 and later releases, dashes are only used for 7-digit and 10-digit numbers when the ‘‘North American Area Code’’ field is administered on the ‘‘Dial Plan” form.’

- Calling Party’s Name

  The calling party’s name is shown on the called party’s display. On calls generated from a DEFINITY system, the calling party’s name is provided if the ISDN-PRI trunk group is administered to send the name to the network. On calls incoming to a DEFINITY system, the (public or private) network may provide the calling party’s name. If the calling party’s name is not available, the called party’s display shows ‘‘CALL FROM’’ instead, followed by the calling party’s number (if available).

- Connected Party’s Number

  The connected party’s number is shown on the calling party’s display. On calls generated from a DEFINITY system, the called party’s number is shown on the calling party’s display as the calling party dials the number. If the (public or private) ISDN network provides the connected party’s number, the calling party’s display is updated to show the connected party’s number. The format of the connected party’s number is the same as that of the calling party’s number described previously on calls incoming to a DEFINITY system. The 0-15 digit number of the party who answers the call is provided to the ISDN network only if the incoming ISDN-PRI trunk group is administered to send the CPN to the network and the “CPN Prefix” form is administered to create a CPN.

  NOTE:
  The connected party may or may not be the party actually called.

  For G3vs/G3sv1, G3iV1, and G3rV1, the connected party number, if sent, is always 10 digits long.

- Connected Party’s Name

  The connected party’s name is shown on the calling party’s display. On calls generated from a DEFINITY system, the (public or private) ISDN network may provide the connected party’s name to the DEFINITY system, when the call is answered. If the connected party’s name is not available, the calling party’s display shows *ANSWERED BY*, followed by the connected party’s number (if available).

On calls incoming to a DEFINITY system, the connected party's name is provided if the incoming ISDN-PRI trunk group is administered to send the name to the network.

NOTE:
The connected party may or may not be the party actually called in the event that the call is transferred before the connected party answers the call.

The display fields that may be used for the ISDN-PRI are as follows:

- Name — Maximum of 15 characters
- Number — Maximum of 15 characters
- Miscellaneous Call Identification — Maximum of eight characters
- Reason for Call Redirection — Maximum of two characters

The display information varies, depending on the type of call, how the call is handled (for example, whether it is redirected or not), and what information is available on the call. The display information for basic calls (those with just a calling and called party) and for redirected calls is given in the following paragraphs.

ISDN-PRI Basic Call

A basic ISDN-PRI call has both a calling and a called party, and the called party answers the call. When the calling party goes off-hook, **a=** appears on the display. The digits then appear as they are dialed. These digits may be overwritten by the trunk group name if the "Outgoing Display" field of the "Trunk Group Administration" form is administered as yes. Once the call is answered by the called party (or by a user with a bridged appearance for the called party), the displays for the calling and called parties are described below.

- If *both the name and number information are available*, the displays are as follows. The "MISCID (Miscellaneous Identification)" field may be blank if that information is not provided.
  - — Calling Party Display

    a= CALLED NAME   CALLED NUMBER   MISCID
  - — Called Party Display

    a= CALLING NAME   CALLING NUMBER   MISCID
- If *only the name information is available,* the displays are as follows:
  - — Calling Party Display

    a= CALLED NAME   MISCID
  - — Called Party Display

    a= CALLING NAME   MISCID

- If *only the number information is available,* the displays are as follows:
  - — Calling Party Display

    a= ANSWERED BY CALLED NUMBER MISCID
  - — Called Party Display

    a= CALL FROM CALLING NUMBER MISCID
- If *neither the name nor the number information is available,* the displays are as follows:
  - — Calling Party Display (shows one of the following, depending on administration)

    a= DIALED NUMBER MISCID

    a= TRUNK NAME MISCID
  - — Called Party Display

    a= TRUNK NAME MISCID

Redirected ISDN-PRI Call

A redirected ISDN-PRI call is a call that has been redirected from the called party's extension by a feature such as Call Coverage, Call Forwarding All Calls, Bridged Call Appearance, or Call Pickup. Once the call is connected, the displays for the calling, called, and connected parties are as follows.

— Calling Party Display

a= CONNECTED NAME CONNECTED NUM MISCID

— Called Party Display

The following information appears on the display if the called party bridges onto the redirected call after it has been answered. In this situation, the connected party's display (given later) shows the same information. The calling party's display is also updated if the calling and called parties are on the same switch.

a= CONFERENCE 2

— Connected Party Display

The connected party is the party who answers the redirected call. The ''R'' indicates the reason for redirection.

a= CALLING ID to CALLED ID R

The CALLING ID and the CALLED ID may be the name or the number, depending on what has been received from the far end.

CPN/BN to Host Call Identification

The CPN/BN to Host feature enables CPN/BN information to be passed from the switch to the ISDN Gateway, so that the ISDN Gateway can forward the information to a host for data screen delivery to agents in an ACD split.

By delivering call identification information such as CPN/BN and switch information such as the answering agent's extension to an adjunct network (ISDN Gateway), the adjunct can automatically deliver data screens to agents for new call arrivals and call transfers.

Figure 3-22 shows a simplified diagram of a CPN/BN to host arrangement. The ISDN Gateway is a 3B2 or 6386 computer connected to the switch on one side and to a host computer on the other side. The connection to the switch is over a synchronous interface with BX.25 protocol.

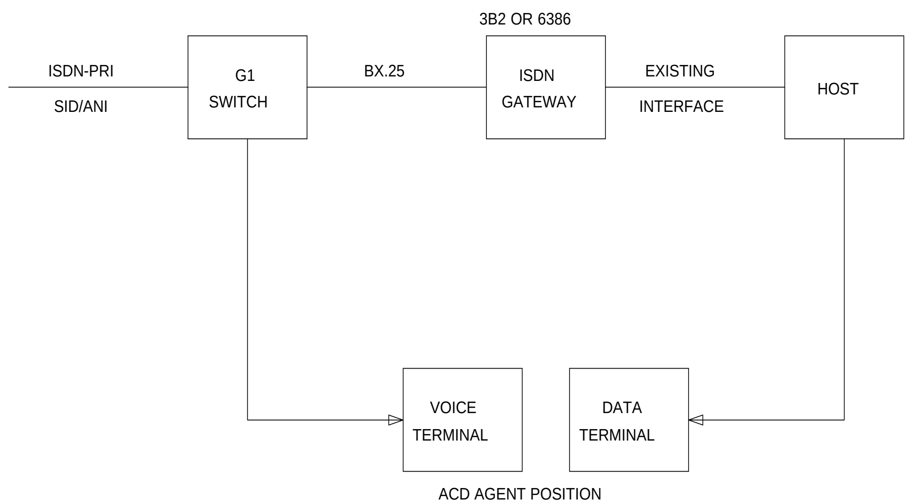


Figure 3-22. CPN/BN to Host Configuration

Private Network Services

In addition to providing access to switched public networks, the ISDN-PRI can provide private network services by connecting DEFINITY Generic 1, 2.1, 3, System 85 R2V4, or G2.2 systems in an Electronic Tandem Network (ETN) or Distributed Communications System (DCS) configuration. This gives customers more efficient private networks that support new integrated voice and data services. ETN and DCS services are provided as follows:

- ETN Services

  DEFINITY Communications Systems that function as tandem nodes in an ETN can be interconnected using DS1 trunking facilities and an ISDN-PRI. All signaling between the tandem switches is done with the ISDN-PRI D-channel and normal ISDN procedures. The ISDN-PRI can also be used to connect ETN tandem and main switches. In this case, the main switch collects all of the address digits from local users as well as users at other satellite and tributary switches, and originates a call over the ISDN-PRI to the tandem switch.

Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) are used with the ISDN-PRI and DS1 trunking facilities to access ETN facilities. The AAR and ARS features are used to collect the dialing information for the call that is originated from the main switch.

- DCS Services

  ISDN-PRI facilities can be used in a DCS arrangement whenever tie trunks are used to connect the DCS nodes.

  Most DCS features are not affected by the ISDN-PRI. However, the ISDN-PRI does have a minor impact on a few of the DCS features, as far as the functions that the local and remote switches must perform. Even though a DCS feature may be slightly affected in this manner, the use of the feature is still the same. If there is a conflict between a DCS message and an ISDN-PRI message on a call (for example, the calling extension number in the DCS message and the calling party's number in the ISDN-PRI message) the DCS message is used.

Wideband Switching

The primary function of the Wideband Switching feature is to provide support for services that require large bandwidth, such as high speed video conferencing. These services have traditionally been handled by dedicated facilities. With the Wideband Switching feature, dedicated facilities are no longer a requirement for these large bandwidth services.

The Wideband Switching feature supports end-to-end connectivity between customer endpoints at data rates ranging from 128 to 1536 kbps over T1 facilities and 128 to 1984 kbps over E1 facilities. Standard data services for use by Wideband Switching are HO (384 kbps), H11 (1536 kbps), H12 (1920 kbps), and NxDS0. See *DEFINITY Communications System Generic 3 Wideband Technical Reference,* 555-230-230, for more detail.

Call-by-Call Service Selection

Call-by-Call Service Selection allows the same ISDN-PRI trunk group to carry calls to a variety of services or facilities (such as a SDN, MEGACOM telecommunications service, MEGACOM 800 service, and so on) and/or carry calls using different inter-exchange carriers. This feature is described in detail under the Call-by-Call Service Selection feature description elsewhere in this manual.

Access to Software Defined Data Network (SDDN)

With ISDN-PRI, the Software Defined Data Network (SDDN) service may be accessed. SDDN provides virtual private line connectivity via the AT&T switched network (4ESS switch). The services provided by SDDN include voice, data, and video applications. SDDN services complement the Software Defined Network (SDN) voice services.

Access to Switched Digital International (SDI)

SDI provides 64 kbps unrestricted connectivity to international locations via the AT&T switched network. It is also the backbone for the AT&T International ISDN network. SDI complements the ACCUNET digital service already available to United States locations. This service can be accessed using the Call-by-Call Service Selection feature. SDI can provide economical high speed data transfer to international locations.

ISDN-PRI Interworking

ISDN-PRI Interworking is the combination of both ISDN-PRI trunking facilities and non-ISDN-PRI trunking facilities on a call. A non-ISDN-PRI trunking facility is any trunk facility supported by the system that does not use the ITU-T recommended Q.931 message set for signaling. Non-ISDN-PRI trunking facilities include facilities such as Analog trunks, AVD DS1 trunks, and DS1 trunks with bit-oriented signaling (robbed-bit or common channel).

The system supports the conversion of ISDN-PRI signaling to non-ISDN-PRI in-band signaling and the conversion of non-ISDN-PRI in-band signaling to ISDN-PRI signaling for Interworking purposes.

A mixture of ISDN-PRI and non-ISDN-PRI signaling is required in order to provide end-to-end signaling when different types of trunk facilities are used on a call. See Figure 3-23 for an example of Interworking. In this example, a call for someone at Switch B comes into Switch A. Interworking allows the ISDN-PRI signaling of the call to be converted at Switch A to non-ISDN-PRI in-band signaling before the call forwards to Switch B. Even though the call comes into Switch A on an ISDN-PRI trunk, Switch A can send the call to Switch B over a non-ISDN-PRI trunk by converting the signaling information.

The system provides accurate CDR billing information on calls that are not Interworked. Accuracy of CDR billing information on Interworked calls is equivalent to the accuracy provided by the public network.

The system does not support the conversion of DCS feature transparency messaging into ISDN-PRI messaging. Therefore, DCS-provided feature transparency is lost when the call leaves the DCS network. The basic call, however, still goes through.

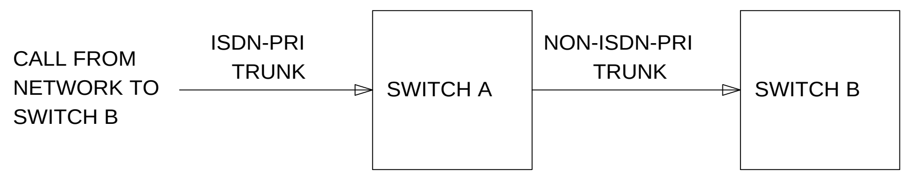


Figure 3-23. Interworking Example

National ISDN-2 Services

G3V3 and later releases support National ISDN-2 (NI-2), which is described in the TR1268 and TR1270 ISDN-PRI protocol standards defined by Bell Communications Research (Bellcore). Bellcore TR1268 and TR1270 are a superset of the NIU-302 standard and are part of the National ISDN-2 plan defined by the North American ISDN Users Forum.

NI-2 offers many of the same services as the AT&T 4ESS protocol, which was the default in G1, G3i, and G3r, and was the protocol for Country 1 (USA) in G3i-global and G3V2 and later releases. G3V3 and later releases support both the 4ESS and the NI-2 protocols and allows you to administer which one you wish to use. You must administer the protocol on the “DS1 Circuit Pack” form. Refer to the “DS1 Circuit Pack” form in the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for more information on administering ISDN-PRI protocols.

NI-2 provides users with the following services:

- Calling Line Identification
- Non-Facility Associated Signaling
- D-Channel Backup
- Wideband Switching
- Call-by-Call Service Selection

Calling Line Identification

Calling Line Identification for NI-2 is essentially Calling Party Number (CPN) Identification, as previously described in this section.

Non-Facility Associated Signaling

Non-Facility Associated Signaling (NFAS) allows an ISDN-PRI T1/E1 Interface D-channel (signaling channel) to convey signaling information for B-channels (voice and data channels) on ISDN-PRI T1/E1 facilities other than the one containing the D-channel. Please refer to the section on Facility and Non-Facility Associated Signaling elsewhere in this manual.

D-Channel Backup

D-Channel Backup is provided to improve reliability in the event of a signaling link failure. D-Channel Backup is discussed in detail in the section on Facility and Non-Facility Associated Signaling elsewhere in this manual.

Wideband Switching

Wideband Switching for NI-2 is essentially the same as that of the AT&T 4ESS ISDN-PRI protocol, previously discussed in this section.

Call-by-Call Service Selection

Call-by-Call Service Selection for NI-2 is essentially the same as that for the AT&T 4ESS ISDN-PRI protocol, previously discussed in this section.

Global ISDN-PRI

The DEFINITY system also provides ISDN-PRI for the global market, based on different countries' requirements. For example, the DEFINITY system G3V3 includes two separate protocols for connecting to the ISDN-PRI network in Germany:

- National ISDN-PRI Layer 3 protocol defined by FTZ 1 TR 6, based on the 1983 version of ITU-T Recommendations Q.930 and Q.931.
- ETSI ISDN (E-DSS1), compatible with the standards established by the European Telecommunications Standards Institute (ETSI).

Additionally, the DEFINITY system includes QSIG Global Networking — a platform that meets the requirements for the European Computer Manufacturers Association (ECMA) Private Network Generic Functional Procedures for Call-Related supplementary services. Please refer to the QSIG Networking section elsewhere in this manual.

Considerations

With the ISDN-PRI, system users have access to a variety of services that are only available through the ISDN.

ISDN-PRI Call Identification Display is provided on a call only if that call is routed through all ISDN-PRI facilities. Non-ISDN-PRI facilities do not carry the necessary display information. If the called party is at a non-ISDN-PRI facility, the system displays either the dialed digits or the trunk group name (depending on administration) on the calling party's display.

ISDN-PRI facilities support equal access to inter-exchange carriers.

Interactions

The following features interact with the Integrated Services Digital Network — Primary Rate Interface feature.

- Attendant Display

  The information provided by ISDN-PRI Call Identification Display is in addition to the display features already provided.

  When an ISDN-PRI call is redirected to the attendant, and both name and number display information is available, the name is displayed on the console for the calling and called party identification.

- Bridged Call Appearance

  ISDN-PRI Call Identification Display information is provided at both the primary extension number and the extension number with the bridged call appearance. Both displays show the same called party information, whether the call is made from the primary extension number or the bridged call appearance. On a call to a primary extension number, the calling party's display shows the identification of the called primary extension number, even if the call is answered by the bridged call appearance.

- Call Forwarding All Calls

  When an ISDN-PRI call is forwarded, no ISDN-PRI Call Identification Display information is shown on the display of the forwarding extension.

  The forwarded-to extension's display shows information on the calling party, called party (if the forwarded-to station is on the same switch), and the reason for redirection.

- Call Pickup

  When an ISDN-PRI call is answered via Call Pickup, the calling party's display identifies the answering party, the called party's display identifies the calling party, and the answering party's display identifies both the calling and called parties.

- Conference — Attendant

  A conference call is identified as a conference call with ‘‘n’’ number of conferees. This display information is generated locally and does not change the display of a user on another switch.

- Conference — Terminal

  A conference call is identified as a conference call with ‘‘n’’ number of conferees. This display information is generated locally and does not change the display of a user on another switch.

- DCS

  If both DCS and ISDN-PRI features are provided over the same facility with a DEFINITY system Generic 3, the ISDN-PRI display information is displayed in DCS format.

- FRL and TCM

  The TCM used to pass on the originating facility’s FRL is sent by ISDN facilities in the SETUP message.

- Hold

  When an ISDN-PRI call is placed on Hold, the display of the party who activates Hold goes blank and then identifies the newly calling or connected party if there is one. The held party’s display remains unchanged. When the held party is reconnected to the holding party, the holding party’s display is updated to indicate the current status of the call.

- Hunting

  On ISDN-PRI calls to a hunt group extension, the calling party’s display identifies either the group or the group member who answers the call, depending on administration.

- TEG

  On ISDN-PRI calls to a TEG, the calling party’s display identifies either the group or the group member who answers the call, depending on administration.

- Transfer

  When an ISDN call is transferred, the display of the party who transfers the call goes blank. The transferred party’s display does not change. The display of the transferred-to party identifies the party who transferred the call.

  When an ISDN call is transferred to a party on the same switch as the party who transfers the call, the information on the display of the party who transfers the call is shown on the transferred-to party’s display.

Administration

ISDN-PRI is administered on a per-system basis by the System Manager. The following items require administration.

- Communication — Interface Link (G3vs/G3s, G3i, and G3i-global)
- Communication — Interface Processor (G3vs/G3s, G3i, and G3i-global)

- DS1 Circuit Pack (for DS1 and E1)
- DS1 Synchronization Plan (for DS1 and E1)
- Processor Interface Circuit Pack (G3vsV1/G3sV1, and G3iV1)
- ISDN-PRI Trunk Group
- GRS Routing Patterns
- Signaling Group (see the Facility and Non-Facility Associated Signaling feature description elsewhere in this document).

The following administration items are required for ISDN-PRI Call Identification Display:

- A Direct Distance Dialing CPN Prefix Table which includes the following items:
  - — Extension Length — From 1 through 5
  - — Extension Code — Defines a set of extensions with the same leading digits as the extension code
  - — CPN Prefix — Used to create a 10 (for G3vsV1/G3sV1, G3iV1, and G3rV1) or 0 to 15 (for G3i-Global, G3V2, and later releases) digit DDD number for an ISDN-PRI CPN. The CPN prefix can be 10 (for G3vsV1/G3sV1, G3iV1, and G3rV1) or 0 to 15 (for G3i-Global and G3V2) digits in length.

    For G3vsV1/G3sV1, G3iV1, and G3rV1, the sum of the number of DDD prefix digits and extension digits must be greater than or equal to 10. If the sum is greater than 10, then the prefix digits takes precedence over the extension digits. The system does not send call identification information on calls from extensions that have an extension code with the DDD prefix left blank or that have no CPN prefix administered.

    For G3i-Global, G3V2, and later releases, the sum must be greater than or equal to the administered number of digits to send out.
  - — Whether to send the calling/connected party number and/or name “(Send Calling Number?”, “Send Connected Number?”, and “Send Name?” fields of the “ISDN-PRI Trunk Group” form). Setting these fields to y (yes), enables the sending of number and/or name, respectively, to the other side of the interface for display purposes.
- Whether the group name or member name is displayed on the calling party’s display (per Hunt Group and TEG).

The following administration is required for CPN/BN to Host Call Identification:

- Some administration may be required so that the switch receives CPN/BN from the originating interface to pass to the ISDN Gateway. If the originating switch is another DEFINITY system, the trunk group administration on that switch needs to be administered (Send Calling/Connected Number? is y) so that CPN info is sent. If the originating switch is a public network switch, it may be necessary to perform some administration on that switch, or to subscribe to a CPN/BN delivery feature. For example, CPN/BN delivery from the AT&T Switched Network is either on a subscription basis (that is, on all calls) or on a per call request basis. Per call CPN/BN requests at the AT&T Switched Network are administered on the “ISDN Trunk Group” form. In either case, four types of CPN/BN delivery may be requested (CPN only, BN only, CPN preferred, and BN preferred).

Hardware and Software Requirements

For G3vs/G3s PBP, G3i-global and G3i:

- A TN767 or TN464C/D/E DS1 circuit pack is required for assignment of a T1 signaling link and up to 24 ISDN-PRI Trunk Group members. The DS1 provides up to 24 ports.
- A TN464C/D/E DS1 circuit pack is required for assignment of an E1 signaling link and up to 31 ISDN-PRI Trunk Group members. The DS1 provides up to 31 ports.
- A TN768 or TN780 Tone Clock circuit pack is required to provide synchronization for the DS1 circuit pack.
- A TN765 Processor Interface circuit pack is required for use with all DS1 circuit packs.

For G3r, the following is required:

- A TN464C/D/E DS1 circuit pack for assignment of a T1 signaling link and up to 24 ISDN-PRI trunk group members. The DS1 provides up to 24 ports.
- For G3iV2 only, a TN464C/D/E DS1 circuit pack for assignment of an E1 signaling link and up to 31 ISDN-PRI trunk group members. The DS1 provides up to 31 ports.
- A TN767 is not needed, although it can be used for an additional B-channel in a signaling group as long as there is a TN464C/D/E in that group to carry the D-channel.
- A TN765 is not needed.

Display-equipped voice terminals are required for the display of ISDN-PRI Call Identification Display information.

One processor interface link is required per ISDN-PRI for G3i, and G3vs/G3s only.

If CPN/BN to Host Call Identification is desired, a computer, such as a 3B2, is required for use as the ISDN Gateway.

ISDN-PRI software is required.

Hardware and software requirements for countries other than the United States of America are detailed in the Application Notes.

Inter-PBX Attendant Calls

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows attendant positions for more than one branch location to be concentrated at one central, or main, location. Incoming trunk calls to the branch location, as well as attendant-seeking voice terminal calls, are routed over tie trunks to the attendants at the main location.

Inter-PBX Attendant Calls are incoming tie trunk calls from the branch location to the main location with the attendant group as the destination. If no attendant in the attendant group is available, these calls are queued. When an attendant becomes available, the call is routed to that attendant for handling. Calls can be extended the same as if the call were an incoming call to the main location. When the attendant releases the call, the tie trunk associated with the call is tied up with the call until the call is dropped.

A DEFINITY system Generic 3 can be a branch or a main location for this feature. A branch location can have local attendants. These local attendants can be accessed by the Individual Attendant Access feature. The attendants at the main location are the local attendants for the main location and also the attendant for the Inter-PBX Attendant Calls.

Considerations

With Inter-PBX Attendant Calls, the number of attendants required at each branch location is reduced. Also, users at each branch location can use the main location to access each of the other branch locations.

The Inter-PBX Attendant Calls feature can also be used within an ETN, where, for example, the attendant group for the network could be located at the main switch and serve other tandem switches connected by tie trunks.

Interactions

The following features interact with the Inter-PBX Attendant Calls feature.

- Attendant Control of Trunk Group Access

  If Inter-PBX Attendant Calls is enabled, and a call at a branch location attempts to access a controlled trunk group, the call is routed to the local attendant at the branch location, if there is one. If there is no local attendant, the call is routed to the attendant group at the main location.

- Attendant Display and DCS Attendant Display

  In a DCS environment, an incoming Inter-PBX Attendant Call from a branch location is displayed at the attendant console the same as a local call.

  In a non-DCS environment, an incoming Inter-PBX Attendant Call is displayed at the attendant console as an incoming tie trunk call.

- Attendant Recall

  If an attendant at the main location holds an Inter-PBX Attendant Call, the calling parties at the branch location cannot recall the attendant.

- Call Coverage

  At a branch location with Inter-PBX Attendant Calls enabled, a call redirected to a coverage path with the attendant group as a coverage point skips that coverage point. It goes to the next coverage point at the local switch, if administered, or continues to ring at the extension that is the previous coverage point. If the attendant group ‘‘0’’ is the only coverage point, it continues to ring at the principal’s extension.

- Centralized Attendant Service (CAS)

  CAS and Inter-PBX attendant calling cannot be enable at the same time.

- Dial Access to Attendant

  Dial Access to Attendant should be administered to the same digit on both the IAS main switch and the IAS branch switch. This is done via the dial platform. On the branch switch, the later PBX attendant access code (“Console Parameters” form) should be administered to match the main PBXs attendant group dial access code.

- Night Service

  The Inter-PBX Attendant Calls feature is deactivated when the branch location is put into night service, and reactivated when the branch location is taken out of night service.

Administration

Inter-PBX Attendant Calls is administered by the System Manager. The following items require administration:

- Branch location access to Inter-PBX Attendant Calls
- Inter-PBX Attendant Calls Trunk Group
- Inter-PBX Attendant Access Code

Hardware and Software Requirements

Requires a tie trunk group between the branch and main locations. No additional software is required.

Intercept Treatment

Feature Availability

Intercept Treatment is available with all Generic 3 releases.

Description

Provides an intercept tone or a recorded announcement or routes the call to an attendant for assistance when calls cannot be completed or when use of a feature is denied.

- Intercept Treatment — Tone

  Provides a siren-type tone to internal calls that cannot be completed as dialed.

  Intercept Tone is provided to voice terminals when users lift the handset and do not dial within 10 seconds, pause longer than 10 seconds between digits during the dialing process, or remain connected to Loudspeaker Paging for longer than an administered interval.

  When a single-line voice terminal user receives Intercept Tone for 30 seconds and does not hang up, then 10 seconds after the Intercept Tone stops, the user receives dial tone for a new call origination. Also, when the user does not hang up after all other parties on the call have hung up, then 10 seconds after all other parties disconnect, the user receives dial tone for a new call origination.

  When multiappearance voice terminal users receive Intercept Tone for 30 seconds and do not hang up, the call appearance returns to idle. If the multiappearance user is the last party left on a call, the call appearance immediately returns to idle.

  If a voice terminal extension is assigned a COS with Off-hook Alert, and the user of that voice terminal receives Intercept Tone for a specified period of time and does not hang up, an emergency call is placed to the attendant.

  On DID calls, if the MFC (Multifrequency Compelled Signaling) Intercept Treatment is set to provide a tone, this signal is sent to the CO to indicate that the called number is invalid. (For further details, see the "R2-MFC Signaling" feature description later in this chapter.)

  On DOD calls using MFC signaling, if the PBX receives the MFC Intercept Tone from the CO indicating the number called is invalid, the PBX sends an intercept signal to the calling station.

- Intercept Treatment — Recorded Announcement

  Provides a recorded announcement to DID and incoming Private Network Access calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations. The System Manager selects and records the message.

  Toll charges do not apply to DID and Private Network Access calls routed to Recorded Announcement.

- Intercept Treatment — Attendant

  Allows attendants to provide information and assistance to callers on all DID or incoming Private Network Access calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations. Normal toll charges apply to these calls.

- Intercept Treatment — Station

  Allows a specific voice terminal to receive certain calls that cannot be completed because of a controlled restriction (see Controlled Restrictions feature) or because the called party has activated Do Not Disturb. The controlled restrictions which can administered to send calls to station intercept are Outward, Termination, and Station-to-Station.

  The calling party hears audible ringing while the call is being routed to the voice terminal assigned for Intercept Treatment. The calling party receives no indication that the call is receiving Intercept Treatment.

Considerations

The Intercept Tone lets a user know when a call cannot be completed as dialed. The user can then hang up or try the call again. When DID and Private Network Access calls cannot be completed as dialed, a recorded announcement can be provided or, for more personal service, the calls can be routed to an attendant or voice terminal user.

Recorded announcements can be used with the system. None, some, or all of these announcements can be used for Intercept Treatment.

Only one person can be connected to an announcement at any given time. The caller is always connected to the beginning of the announcement.

Interactions

- R2-MFC

  For DID MF calls, whether the corresponding B.x signal or the Intercept Tone should be sent to the CO is administrable. The default is to send the standard DID/TIE/ISDN Intercept Treatment described in this section. If the option to send the B.x signal is set, then:

— For Group II type calls, the B.x signal for the intercept is sent to the CO.

— For non-Group II type calls, if the CO dials an invalid number, the trunk is locked (regardless of this option). If the CO dials the number which is valid but not assigned, the Intercept Treatment Tone is sent to the CO.

- Recorded Announcement

  Attendant Intercept and Recorded Announcement Intercept (both optional) cannot be used together. DID calls and Private Network calls cannot be assigned Intercept Treatment — Tone.

Administration

The Intercept Tone is standard and requires no administration. However, administration is required to determine whether DID and Private Network Access calls are routed to the attendant or to an announcement. Administration is required to determine whether calls sent to intercept because of controlled restrictions are routed to intercept tone, a voice terminal, an attendant, or an announcement. If an announcement is to be used, the announcement must be administered. Intervals are administrable with the “System Parameters” form.

Hardware and Software Requirements

Requires announcement equipment and one port on an Analog Line circuit pack for each announcement for each external analog announcement device or an aux trunk port for each external auxtrk device. The Multiple Integrated Announcements allows equipping additional TN750C circuit packs. See the "Recorded Announcement" feature for more details.

With DEFINITY system Generic 3i, a TN750 Announcement circuit pack can be used to provide up to 128 (for G3i), or up to 256 (for G3r) different announcements. The announcements can be recorded directly onto the TN750 circuit pack.

No announcement equipment is required when this circuit pack is used. No additional software is required.

Intercom — Automatic

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides a talking path between two voice terminal users. Calling users press the Automatic Intercom button and lift the handset, or vice versa. The called user receives a unique intercom alerting signal, and the status lamp associated with the Dial or Automatic Intercom button, if provided, flashes.

Considerations

With the Intercom—Automatic feature, users who frequently call each other can do so by pressing one button instead of dialing an extension number.

Single-line voice terminal users can receive Automatic Intercom calls, but cannot originate them.

A combination of the Intercom—Automatic and Intercom—Dial features can be used between terminals so that Intercom—Automatic applies in one direction and Intercom—Dial applies in the other.

Two terminals with Intercom—Automatic to and from each other, or terminals with combined Intercom—Automatic and Intercom—Dial to each other, must be in the same Intercom group.

Interactions

The following features interact with the Intercom—Automatic feature.

- **Call Coverage**

  Intercom calls are redirected only if the caller activates Go to Cover.

- **Data Privacy and Data Restriction**

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. Intercept tone is received when the ICOM button is pressed under this condition.

- **Intercom—Dial**

  This feature must be provided. Users assigned an Automatic Intercom button must be a member of the same Dial Intercom group as the destination extension number.

- **Single-Digit Dialing and Mixed Station Numbering**

  Prefixed extensions greater than five digits (including the prefix) in length cannot be assigned to intercom lists.

- **Class of Restrictions (COR)**

  COR may be administered such that station A cannot dial Station B. The user at Station A would receive intercept tone. However, Station A can be administered such that it has Intercom—Automatic capability to Station B. Thus A could call B. The System Manager has control of station restrictions and which stations are in the same Intercom group.

Administration

Intercom—Automatic is assigned on a per-voice terminal basis by the system administrator. Before Intercom—Automatic can be assigned, the associated Intercom group must be established. Each Intercom group requires the following administration:

- Intercom group number
- Length of dial code
- Extension number within the group
- Dial codes to access Intercom group members

Once the Intercom group is established, Automatic Intercom buttons can be assigned to members of the group. The following items must be administered for each button.

- Intercom group number to be accessed
- Dial code assigned to group member to be accessed

Hardware and Software Requirements

No additional hardware or software is required. Voice terminals equipped with a buttons and a status lamp are useful, but not required.

Intercom—Dial

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows multiappearance voice terminal users to gain rapid access to selected other voice terminal users within an administered group. Calling voice terminal users lift the handset, press the Dial Intercom button, and dial the 1- or 2-digit code assigned to the desired party. The called user receives alerting tone, and the status lamp associated with the Intercom button, if provided, flashes.

Considerations

With the Intercom—Dial feature, a group of users who frequently call each other can do so by pressing a Dial Intercom button and dialing a one- or two-digit code instead of dialing an extension number.

The maximum number of intercom groups that can be established varies according to the system. See Appendix A, "System Parameters".

Single-line voice terminals can receive Intercom—Dial calls, but cannot originate them.

A combination of Intercom—Dial and Intercom—Intercom can be used between terminals so that Intercom—Dial applies in one direction and Intercom—Intercom applies in the other.

A Intercom—Dial user can place an intercom call to all members in the group, including Intercom—Automatic members.

Two terminals with Intercom—Dial to and from each other, or two terminals with combined Intercom—Dial and Intercom—Automatic to and from each other, must be in the same Intercom group.

Interactions

The following features interact with the Intercom—Dial feature.

- **Intercom—Automatic**

  Users assigned this feature must be a member of a Dial Intercom group.

- **Call Coverage**

  Intercom calls are redirected to Call Coverage only if the caller activates Go To Cover.

- **Data Privacy and Data Restriction**

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. Intercept tone is received when the ICOM button is pressed under this condition.

- **Single-Digit Dialing and Mixed Station Numbering**

  Prefixed extensions greater than five digits (including the prefix) in length cannot be assigned to intercom lists.

Administration

Intercom—Dial is administered by the system administrator. The following items require administration:

- Intercom groups
  - Group number
  - Length of dial code
  - Extension numbers within the group
  - Dial codes to access Intercom group members
- Dial Intercom buttons.

Hardware and Software Requirements

No additional hardware or software is required. Voice terminals equipped with a buttons and a status lamp are useful, but not required.

Internal Automatic Answer (IAA)

Feature Availability

This feature is available with all Generic 3 releases except G3i-Global and G3rV1.

Description

Allows specific voice terminals to answer incoming internal calls automatically. Internal Automatic Answer (IAA) is intended for use with selected voice terminals with a built-in speakerphone, or stations with adjunct speakerphone or headset with the headset option enabled on the “Station” form’. For G3vsV1/G3sV1 and G3iV1, this feature is only available with the MERLIN/System 25 7313H, 7314H, 7315H, 7316H, and 7317H terminals. For G3V2 and later releases, this feature is available for most multi-function stations with a speakerphone or a headphone. A 602A terminal is off-hook when the headset/speakerphone is connected, therefore IAA answers a call if all other call-appearances are idle.

Internal calls eligible for automatic answer by the IAA feature are:

- Station-to-station voice calls, with both voice terminals on the same switch. (This includes redirected intra-switch calls.)
- An internal call over a DCS configuration to another internal extension. (This includes redirected inter-DCS calls.)
- External calls that are extended by an attendant console when the ‘‘Internal Automatic Answer for Attendant Extended Calls’’ option on the “System Parameters Feature’’ form is enabled.

The following types of calls are *not* eligible to be answered automatically via IAA:

- Calls from public network trunks (including PCOL)
- Calls arriving on non-DCS tie trunks
- Automatic Callback calls
- Automatic Circuit Assurance calls
- Data calls
- External calls extended by an attendant console when the ‘‘Internal Automatic Answer for Attendant Extended Calls’’ option on the “System Parameters Feature’’ form is disabled
- Calls redirected because of queue overflow of Emergency Access to the Attendant calls
- Terminating Extension Groups
- The receiving station’s “Active Station Ringing” option is set to “continuous”

An eligible call is answered automatically via the IAA feature if IAA is activated at the answering voice terminal and the answering voice terminal is idle (on-hook and able to accept an incoming call). A voice terminal that is off-hook, in the process of dialing digits, or has a call on hold is not considered idle.

IAA Feature Operations

With IAA, a single programmable feature button (IAA) can be assigned to voice terminals during voice terminal administration. When this feature button is pressed, the associated green status lamp lights and the IAA feature is activated. Pressing the same button again causes the status lamp to go dark and the IAA feature is deactivated. (Pressing the feature button has no effect on a currently active call or a ringing call.) As long as the IAA button's status lamp is on, the IAA feature remains activated. The IAA button may be toggled on or off at any time, regardless of the state of the voice terminal; the voice terminal's speakerphone is not affected by this action. Also, using the speakerphone to place calls does not affect the state of the IAA feature.

The calling voice terminal receives a tone when its call is answered automatically by a voice terminal with IAA activated. The called voice terminal receives a tone (a ''ring ping'') and then goes off-hook when automatically answering an IAA-eligible call. Both the answering voice terminal's speaker and microphone are turned on.

If the user of a voice terminal with IAA activated is currently busy on a call or is in the process of dialing digits, subsequent incoming calls are treated as if IAA is not activated. Thus, an already established call or call establishment activity temporarily disables the IAA feature for incoming calls.

Considerations

IAA provides convenient hands-free answering of internal calls.

The user of a voice terminal with IAA activated should always deactivate the feature when leaving the work area. Otherwise, incoming calls are unintentionally ''answered'' by the unattended station. This prevents calls from going to coverage.

For G3vsV1/G3sV1, and G3iV1, IAA is only available with the MERLIN/System 25 731xH series voice terminals and is not available with any other voice terminals.

There is no feature access code for the IAA feature; the station `int-aut-on` must be administered.

IAA is not available with ISDN BRI voice terminals or Attendant Console.

IAA and Automatic Answer are mutually exclusive.

IAA and the Continuous Ring options are mutually exclusive.

Interactions

The following features interact with the Internal Automatic Answer (IAA) feature:

- ACD Calls

  IAA applies to internal calls to an ACD agent.

- Automatic Answer

  IAA and Automatic Answer cannot be administered simultaneously on the same voice terminal.

- Automatic Callback

  Callback calls via Automatic Callback are not answered automatically by the IAA feature.

- Automatic Circuit Assurance (ACA)

  Calls generated by the ACA feature are not eligible for IAA.

- Bridged Call Appearance — Multi-Appearance Voice Terminal

  Calls terminating to a bridged appearance are not eligible for IAA at the bridged station, even if the bridged station has IAA enabled. (IAA can be used by the principal station to answer the call.)

- Bridged Call Appearance — Single-Line Voice Terminal

  Calls terminating to a bridged appearance are not eligible for IAA at the bridged station, even if the bridged station has IAA enabled.

- Call Coverage

  If an internal call is redirected to another voice terminal by a Call Coverage redirection criteria, then that call is eligible for IAA at the redirected voice terminal.

  IAA does not apply to calls to the original called extension when:

  — The called voice terminal has activated Send All Calls

  — The calling voice terminal has selected Go To Cover before placing the call

  Calls directed to a Coverage Answering Group are not eligible for IAA.

  NOTE:
  If the system administrator sets the coverage path for a station to ‘‘All-Calls’’ and that principal station activates IAA, the first coverage point hears a ring, the principal station automatically answers and the coverage-simulated bridge is dropped. The coverage station rings, but is not able to answer the call because the coverage simulated bridge has been dropped.

- Call Forwarding

  Calls to a station with IAA and Call Forwarding activated are forwarded and are not answered by the station dialed.

  NOTE:
  If the ‘‘forwarded to’’ station, is internal and has IAA active, the ‘‘forwarded to’’ station automatically answers the redirected call.

- DeluxePaging (Call Park)

  When using the DeluxePaging feature and call park times out, it returns to the originating station which parked the call and is eligible for IAA.

- Call Pickup

  Internal calls to a voice terminal in a Call Pickup group are eligible for IAA. If the called extension in a Call Pickup group is IAA-active, the call is automatically answered. An IAA-active voice terminal is not able to automatically answer calls to other voice terminals in it’s Call Pickup group.

- Conference

  Internal conference calls are eligible to be automatically answered via the IAA feature. If more than one conference party has joined a conference call through automatic answer, such parties remain connected until either they disconnect or the controlling party drops the call.

- Data Call Setup

  Data calls are not eligible for IAA.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  Internal calls to a DDC or UCD group member are eligible for IAA.

- Distributed Communications System (DCS)

  If a call is from another switch in a DCS configuration and the origin of the call is known to be an internal voice terminal on that switch, then that call is considered internal and is eligible for automatic answer.

- Do Not Disturb

  Activation of the Do Not Disturb feature preempts IAA at the called voice terminal.

- Go To Cover

  IAA does not apply to calls to the original called extension when the calling voice terminal has selected Go To Cover before placing the call.

- ISDN BRI

  IAA is not available with ISDN BRI stations.

- Ringback Queuing

  Automatic calls generated by the Ringback Queuing feature are not eligible for IAA.

- Send All Calls

  IAA does not apply to calls to the original called extension when the called voice terminal has selected Send All Calls.

- Terminating Extension Group

  Calls to a Terminating Extension Group extension are not eligible for IAA; however, calls placed to the individual extension are eligible.

- Transfer

  Transferred calls are eligible to be automatically answered via the IAA feature.

Administration

Internal Automatic Answer (IAA) is administered on a per-voice terminal basis by the System Manager. The only terminal administration required is an IAA button (int-aut-an), which must be assigned to each voice terminal desiring IAA, and for multifunction voice terminals that do not have a built-in speakerphone, the headset option must be set to ‘‘y’’ on the ‘‘Station’’ form.

If a voice terminal’s IAA button is removed via administration while the IAA feature is active, the IAA feature is deactivated and the associated status lamp goes dark. Current calls are not affected by this action.

A system-wide option determines whether IAA-administered voice terminals automatically answer external calls that are extended by an attendant console. Typically, to avoid redirecting an external call to an unattended IAA voice terminal, the IAA option on the “System Parameters’’ form is disabled. However, you can allow IAA to automatic answer extended external calls by enabling the System Parameters Form’s IAA option.

Hardware and Software Requirements

For G3V2 and later releases, all multi-function voice terminals equipped with a speakerphone or headset may use this feature, except ISDN BRI voice terminals.

For G3vsV1/G3sV1 and G3iV1, only the following voice terminal types support IAA:

- 7313H
- 7314H

- 7315H
- 7316H
- 7317H

Each of these hybrid voice terminals listed above requires a port on a TN762B Hybrid Line circuit pack.

No additional software is required.

Intraflow and Interflow

Feature Availability

This optional feature is available with all Generic 3 releases when ACD software is purchased.

NOTE:
For G3vs/G3s ABP, ACD is only available as part of the Basic Call Center Option.

Description

Allows ACD calls to be redirected from one split to another split under busy or unanswered conditions. Intraflow provides redirection of ACD calls to other splits within the system and may be activated using Call Coverage or Call Forwarding All Calls. Interflow uses the Call Forwarding All Calls feature to redirect ACD calls to an external location.

Intraflow allows splits to be assigned coverage paths or forwarded. Also, a split can be a point in a coverage path. Thus, Intraflow uses the Call Coverage feature to redirect ACD calls from one split to another split according to the coverage path's redirection criteria. For instance, a split's coverage path can be administered so incoming ACD calls are automatically redirected to another split during busy or unanswered conditions. Additionally, Call Forwarding can be used to unconditionally intraflow a split's calls.

An ACD call is intraflowed to another split whenever it is forwarded or the assigned Call Coverage redirection criteria is met. For a detailed description of Call Coverage redirection criteria, see the "Call Coverage" feature description in this chapter. For a detailed description of Call Forwarding, see the "Call Forwarding All Calls" feature description in this chapter.

If an ACD call is intraflowed to another split, the system attempts to terminate the call to an available agent. If an agent is not available, the system tries to place the call in queue at the covering split. The call enters the covering split queue, unless one of the following conditions exists:

- The destination split's ‘‘inflow threshold’’ is met
- The queue is full
- There are no agents logged in
- All the logged-in agents are in the AUX work mode

The inflow threshold is a parameter that is assigned to each split. If the oldest call in the split queue has remained in that queue for a length of time greater than the inflow threshold (0 to 999 seconds), then ACD calls cannot be intraflowed into that split. If an ACD call is forwarded or meets the Call Coverage redirection criteria, but cannot be intraflowed to another split or point in the coverage path, it will remain in queue at the original split even though coverage tone may be heard.

A split can be administered such that ACD calls intraflowed via Call Coverage from that split to another split have priority over other calls in queue at that split. If an ACD call intraflows from a split with ‘‘priority on Intraflow’’ to a covering split, and enters the queue at the covering split, that call is positioned in the queue ahead of any nonpriority calls but behind other priority calls already in the queue. In other words, all priority calls are answered before any nonpriority calls.

If the covering split is assigned a second delay announcement, an ACD call intraflowed to that split will receive the announcement, if the call remains in queue for a length of time equal to the second delay announcement interval. After the announcement is heard, the caller hears either music-on-hold or silence until the call goes to an agent. An ACD call intraflowed (via Call Coverage) to a covering split is never connected to the first delay announcement assigned to the covering split.

Calls redirected via Call Forwarding will receive a delay first announcement at the forwarded-to split. (A forced first announcement is never delivered at the destination split.)

As an illustration of how Intraflow works, assume the following:

- A call is intraflowed from split 1 to split 2 via Call Coverage.
- Split 1 is assigned priority on intraflow.
- Split 2 has a queue with three priority calls and four nonpriority calls.
- Split 2 has an inflow threshold of 90 seconds and the oldest call in queue at split 2 has been in queue for 60 seconds.
- Split 2 has been assigned a second delay announcement and has a second delay announcement interval of 45 seconds.
- Music-on-Hold is provided.

When the call is intraflowed from split 1 to split 2, the call enters the split 2 queue and is positioned in the queue ahead of the four nonpriority calls. The intraflowed call is then the fourth call in queue. Assume the call stays in the queue for 45 seconds and is still not answered. The call, at the end of 45 seconds, is connected to the second delay announcement for split 2. When the announcement is complete, the caller hears music-on-hold until the call is connected to an available agent. If the second delay announcement is administered to repeat, the system will attempt to connect the call to the second delay announcement again after the delay interval has expired (45 seconds.)

The Interflow feature allows ACD calls to be redirected from one split to a split on another switch or to another external location. This is accomplished by forwarding calls that are directed to the split extension to an off-premises location via the Call Forwarding All Calls feature. Calls can be forwarded to destinations off the PBX (that is, phone numbers on the public-switched telephone network). Calls cannot conditionally interflow. If a coverage point station or split is forwarded/interflowed, it is taken out of the coverage path. For details on how calls are forwarded to an off-premises extension, see the "Call Forwarding All Calls" feature description in this chapter.

A Coverage ICI button can be assigned to an agent’s multi-appearance voice terminal. The Coverage ICI button allows the agent to identify a call that is intraflowed from another split. When an agent receives a call that has intraflowed from the split assigned to that button, the button’s status lamp will light.

More advanced Interflow capabilities are supported by the "Call Vectoring" and "Look Ahead Interflow" features discussed in this chapter.

Considerations

Intraflow and Interflow provide the means to redirect ACD calls to alternate splits. Intraflow via Call Coverage provides for conditional redirection of ACD calls when certain conditions are met (such as Busy or Don’t Answer); Call Forwarding All Calls provides unconditional redirection. Therefore, calls can be directed to less busy splits, resulting in more efficient call handling. Interflow provides for all ACD calls to a specific split to be redirected to a split at the same location or another location.

Interactions

The following features interact with the Intraflow and Interflow features.

- Attendant Display and Voice Terminal Display

  These features provide call and queue identification for the covering split agents.

- Call Coverage

  When Intraflow via Coverage is provided, the Coverage Don’t Answer Interval associated with Call Coverage begins when the call enters the split queue. If the Coverage Don’t Answer Interval expires before either of the two delay announcement intervals expires, the call is redirected to coverage. If no coverage point is available to handle the call, the call remains in queue and may then be connected to a delay announcement. If either of the delay announcement intervals expires before the Coverage Don’t Answer Interval, the call is connected to a delay recorded announcement, if available.

- Temporary Bridged Appearance

  If an ACD call terminates to a split agent, but is intraflowed to another split before being answered, the Temporary Bridged Appearance at the split agent’s terminal or console is no longer maintained.

Administration

Intraflow and Interflow are administered by the System Manager. The following items require administration:

- Coverage Paths

  The same coverage path can be used for as many splits as desired. For efficient operation of the Intraflow feature, it is recommended that the redirection criteria for a split's coverage path be administered so that calls are redirected under busy or don't answer conditions (do not use ''all'' or ''send all calls'' as the redirection criterion).

- Don't Answer Interval and Don't Answer Interval for Subsequent Redirection

  The Don't Answer Interval specifies the number of ringing cycles that occur while the call is in queue or, if the call has been sent to an agent, the number of ringing cycles heard at the agent's terminal before the call is redirected to the first coverage point. This interval can be administered. All splits with the same coverage path are assigned the same Don't Answer Interval.

  The Don't Answer Interval for Subsequent Redirection specifies the number of rings at a covering split before the call attempts to redirect to the next coverage point. This interval is recommended to be two rings but can be administered from 1 to 99 rings. This interval is administered as a system parameter.

  NOTE:
  The ringing cycle is 5.2 seconds is not administrable; however, the number of ringing bursts (one, two, or three) is administrable by call type (internal, external, or priority).

- Whether or not each split has priority on Intraflow
- Inflow threshold
- Coverage ICI buttons as required
- All other items listed under administration of the ACD feature

Hardware and Software Requirements

No additional hardware is required. ACD software is required.

Last Number Dialed

Feature Availability

This feature is available with all Generic 3 releases.

Description

Automatically redials the last number dialed when users press the Last Number Dialed button or dials the Last Number Dialed feature access code.

The system saves the first 24 digits of the last number dialed whether the call attempt was manually dialed or an Abbreviated Dialing button was pressed.

Considerations

Last Number Dialed prevents the user from having to redial a busy number. If a user has dialed a busy number and that was the last number dialed, the user simply activates Last Number Dialed by button or dial access code. The system automatically dials the same number again.

Special characters (Pause, Wait, Mark, or Suppress) stored in an Abbreviated Dialing button are recognized by the system and are outpulsed when such a number is automatically redialed by the Last Number Dialed feature.

When a manually dialed number is redialed automatically, a delay in dialing is not recorded. The system outpulses the numbers as one continuous digit string. Thus, to accomplish automatic redialing, the distant end must accept the outpulsed digits without delay.

Last Number Dialed information is not saved on tape and can be used only for the next call origination. End-to-end signaling digits manually dialed are never saved.

Interactions

The following features interact with the Last Number Dialed feature.

- Abbreviated Dialing

  If the previously called number was in an Abbreviated Dialing privileged list, and if the user is not normally allowed to dial the number because of his or her COR, Intercept Treatment is given when using Last Number Dialed. To redial the number, the user must again use the Abbreviated Dialing privileged list.

- Automatic Callback

  Automatic Callback can be used after the Last Number Dialed feature is used on a call to an internal voice terminal.

- Bridged Call Appearance

  Activation of the Last Number Dialed feature causes the last number dialed from the voice terminal to be redialed, regardless of which extension number is used (primary or bridged call appearance).

Administration

Last number dialed is administered by the System Manager. The following items require administration:

- Feature Access Code for Last Number Dialed
- Last Number Dialed button

Hardware and Software Requirements

No additional hardware or software is required.

Leave Word Calling

Feature Availability

Leave Word Calling is standard with all Generic 3 releases except G3vs/G3s ABP. For G3vs/G3s ABP, Leave Word Calling is only available with the Voice Mail Application Software Option package, which is available when you purchase an AT&T voice processing adjunct (AUDIX, AUDIX Voice Power, AUDIX Voice Power Lodging, DEFINITY AUDIX).

Description

Allows internal system users to leave a short preprogrammed message for other internal users. For G3r, there can be multiple Message Server Adjuncts and AUDIX adjuncts.

The LWC feature electronically stores a standard message, for example, CARTER, ANN 2/7 10:45a 2 CALL 3124. This message means that Ann Carter called two times, the last time on the morning of February 7 and, wants a return call to extension 3124.

When a message is stored, the Message lamp on the called voice terminal automatically lights. This lamp is referred to as an Automatic Message Waiting lamp since the status of the lamp is controlled automatically by the system.

Another voice terminal may also receive an indication that an LWC message has been left for the called party. This is accomplished via a remote Automatic Message Waiting lamp at another voice terminal. The remote Automatic Message Waiting lamp is a status lamp associated with a button assigned for this purpose. The remote Automatic Message Waiting lamp lights at the same time that the Message lamp lights at the called voice terminal. A common use of a remote Automatic Message Waiting lamp is to provide an indication of an executive's message on a secretary's voice terminal. If the executive calls from outside to receive any messages, the secretary knows at a glance if any messages have been left. Remote Automatic Message Waiting lamps also allow an indication of LWC messages left for a DDC group, a UCD group, an ACD split, a TEG, and a PCOL group.

When identical messages are entered in the system, the date, time, and number of messages are updated. When nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

Messages can be stored by calling, called, and covering users. A covering user can be through the Call Coverage, Call Pickup, or Call Forwarding All Calls features. Messages are stored as follows:

- Storage by Calling User

  — Before dialing the desired extension number, the user presses the LWC button or dials the LWC access code and then dials the desired number.

  — After dialing the desired number but before the call is answered, a multiappearance voice terminal user presses the LWC button or a single-line voice terminal user presses the Recall button and dials the access code.

  — After the call has been answered by any user, the calling user presses the LWC button or the Recall button and dials the access code.

- Storage by Called User

  — After answering the call, the called user presses the LWC button. This leaves a message for the calling user to call back. (A called user can store an LWC message by dialing the LWC access code only if the called user has an analog voice terminal.)

- Storage by Covering User

  — After answering the call, the covering user presses the Coverage Callback button. This stores a message for the called user to call the calling user. After answering the call, the covering user presses the LWC button. This leaves a ‘‘call me’’ message for the originally called user.

In addition, a user placed on hold can activate LWC and leave a message for the holding user to place a return call.

LWC messages can be stored in the switch, in an AUDIX Adjunct or DEFINITY AUDIX System (but not AUDIX Voice Power), or in a Messaging Server Adjunct (MSA).

Messages are retrieved by users who have the Voice Terminal Display or Attendant Display feature. Users without the Voice Terminal Display feature have their messages retrieved by systemwide message retrievers or by covering users in their Call Coverage path.

Messages are retrievable with the Voice Message Retrieval feature.

If the following conditions are met, messages for users can be retrieved by selected voice terminal users or any attendant:

- The retriever must be in the called user’s Call Coverage path or must be administered as a systemwide retriever.
- Permission to retrieve messages must be administered to the called voice terminal.

Messages are protected by restricting unauthorized users from displaying, canceling, or deleting messages. A Lock function restricts a voice terminal, and an Unlock function releases the restriction. The Lock function is activated by dialing a systemwide access code. The Lock function is canceled by dialing a systemwide access code and then an Unlock security code unique to the voice terminal. These functions apply only to the voice terminal where the operation is performed. A status lamp can be assigned to show the locked or unlocked status of the voice terminal.

G3V2 and G3i-Global provide a choice of English or Italian for Voice Message Retrieval.

A calling user who left an LWC message can cancel that message if it has not already been accessed. The calling user lifts the handset, presses the LWC Cancel button or dials the access code, and dials the extension number of the called party. This deletes the message (even if the count was more than one) and causes all Message lamps associated with the called voice terminal to go dark (if the called user has no other messages).

Considerations

LWC lets users automatically leave short, simple messages for other users. When a voice terminal’s message lamp is lighted, the user simply has the message retrieved by an authorized user. This reduces the time spent making handwritten notes.

Ten terminals, or nine terminals and the attendant console group, can be administered as systemwide message retrievers.

If the stored message level reaches 95 percent of capacity, the status lamp associated with all Coverage Message Retrieval buttons in the system flashes. These lamps continues to flash until the stored message level drops below 85 percent capacity. Authorized retrievers can selectively delete messages to gain storage space. Old messages are not automatically purged by the system.

Interactions

The following features interact with the LWC feature.

- AUDIX Interface

  LWC Cancel cannot be used to cancel an AUDIX message.

- Bridged Call Appearance

  A LWC message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension number assigned to the bridged call appearance. When a user calls a primary extension, and activates LWC, the message is left for the primary extension, even if the call was answered at a bridged call appearance.

- Call Coverage

  The LWC feature can be used with or without Call Coverage. However, the two features complement each other. The Coverage Callback option of the Call Coverage feature is provided by the LWC feature. Also, a caller can activate LWC for the called party even if the call has been answered by a covering user.

- Centralized Attendant Service (CAS)

  LWC Message Retrieval does not work with CAS.

- Conference

  A member of a conference call cannot activate LWC because that user cannot be uniquely identified.

  After LWC has been activated for a party on a conference or transfer, the conference/transfer originator cannot press the Conference/Transfer button a second time to return to the original call. The conference/transfer originator must select the call appearance button to return to the previously held call.

- Expert Agent Selection (EAS)

  With G4V3 and later releases in an EAS environment, the Message Waiting Lamp by default tracks the status of messages waiting for the logged in EAS agent LoginID rather than messages for the physical terminal. With these releases, the operation of the Message Waiting Lamp can be changed so that it tracks the status of messages waiting for the physical terminal where the agent is logged in. See the 'Feature-Related System-Parameters' form in the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for more information.

- MERLIN/System 25 Voice Terminal — 731xH Series Support

  The MERLIN/System 25 7309H, 7313H, 7314H, 7315H, 7316H, and 7317H voice terminals support Leave Word Calling but do not support Message Retrieval via the display.

Administration

LWC is administered by the System Manager. The following items require administration:

- Identities of authorized systemwide LWC retrievers
- Locking and unlocking message retrieval and cancellation (per voice terminal)
- Lock dial access code (systemwide)
- Lock status lamp (per voice terminal)

- LWC activation (per voice terminal and the attendant group)
- LWC activation dial access code (systemwide)
- LWC button (per voice terminal)
- LWC Cancel button (per voice terminal)
- LWC cancellation dial access code (systemwide)
- LWC reception (per voice terminal and per Hunt group, that is, DDC group, UCD group, TEG, and PCOL group)

  Several options are available for G3r: SPE (switch processor element), msa-spe, AUDIX and none. If a customer administers for spe, then the message never goes to the MSA or AUDIX, it stays in spe only. If this is administered for msa-spe and the msa link goes down, the message is stored in spe until the link is restored and then it is downloaded to the MSA. If during administration, a station is changed from msa-spe to spe, all messages must be retrieved from the MSA first. If the opposite change is performed this is not necessary because, the message is downloaded to the MSA automatically.
- Maximum number of messages.
- Remote Automatic Message Waiting lamp on another voice terminal 80 allowed per extension, including an extension number for a DDC group, UCD group, TEG, and PCOL group; 80 allowed per system)
- Retrieval permission for covering users (per voice terminal)
- Unlock dial access code (systemwide)
- Unlock security code (per voice terminal)

All buttons associated with the display modes are administered through the Attendant Display and Voice Terminal Display features.

Hardware and Software Requirements

The appropriate voice synthesis board — TN725 (US English), TN433 (Italian), and TN457 (British English) — is required for Voice Message Retrieval.

If Leave Word Calling is used with AUDIX and/or MSA, additional hardware is required.

Line Lockout

Feature Availability

This feature is available with all Generic 3 releases.

Description

Removes single-line voice terminal extension numbers from service when users fail to hang up after receiving dial tone for 10 seconds and then intercept tone for 30 seconds. These intervals are administrable.

Line Lockout occurs as follows:

- A user does not hang up after the other party on a call is disconnected.

  In this case, the user receives the dial tone for 10 seconds and then receives the intercept tone for 30 seconds. The voice terminal is then taken out of service, if the handset is still lifted.

- A user pauses for 10 seconds between digits while dialing.

  In this case, the user receives intercept tone for 30 seconds. The voice terminal is then taken out of service, if the handset is still lifted.

The out-of-service condition remains in effect until the voice terminal user hangs up.

Considerations

The out-of-service condition provided by Line Lockout does not tie up switching facilities or call processing time. The facilities are then available for other users.

This feature does not apply to multiappearance voice terminals.

Interactions

Call intercept is provided by the Intercept Treatment feature.

Administration

The timeout lengths are administrable on the “System Parameters — Features’’ form.

Hardware and Software Requirements

No additional hardware or software is required.

Look Ahead Interflow

Feature Availability

This optional feature is available with all Generic 3 releases except G3s ABP.

Description

Provides flexible and intelligent ACD load balancing based on the Call Vectoring feature. Look Ahead Interflow enhances Call Vectoring interflow by ensuring that calls do not interflow to a backup switch that cannot accept the calls.

A Look Ahead Interflow call is attempted when a `route-to digits` or `route-to number` successfully seizes an ISDN-PRI trunk. A vector on the receiving switch then either accepts or denies the Look Ahead Interflow call attempt. The sending switch does not relinquish control of the call until it is accepted by the receiving switch. Until the call is accepted, the caller continues to hear any tones applied by the sending switch vector and the call remains in any switch queues.

If the call is accepted, the call is removed from any queues at the sending switch and control of the call is passed to the receiving switch.

If the call is denied, vector processing simply continues at the sending switch. Audible feedback and the call's position in any queues at the sending switch remains unaltered so the caller is unaware that a Look Ahead Interflow call attempt has been made. The call vector may then apply alternate treatment, which may include placing another Look Ahead Interflow call to an alternate backup switch.

Since Look Ahead Interflow uses the Call Vectoring feature, it is recommended that the "Call Vectoring" feature description, discussed elsewhere in this chapter, be read and understood before continuing with this description. This should make the "Look Ahead Interflow" description easier to comprehend.

Look Ahead Interflow is provided by using private network ISDN-PRI connections. Look ahead interflow calling SDN service is also provided via PRI. The sending switch generates an ISDN-PRI Look Ahead Interflow Information Element that is included with the ISDN SETUP message when the call routes on an ISDN facility. This information element contains interflow-related data, including DNIS information. The DNIS name may then be displayed if the call goes to an agent on the receiving switch.

Look Ahead Interflow Basics

When one switch has an overload of incoming calls, it may become necessary to route some of the incoming calls to another switch so they can be handled more efficiently and will not be lost. Look Ahead Interflow is simply the means used to determine whether the other switch is able to handle the calls. When preset thresholds at one switch are reached (for example, the number of calls in queue), and another call comes in, that switch checks to see if another switch can handle the call. The other switch then checks to make sure it can handle the call. If it can, the call is sent to that switch. If it cannot, the sending switch must try to process the call in another way, such as intraflowing to a backup split or placing a second Look Ahead Interflow call attempt to an alternate backup switch.

Look Ahead Interflow is accomplished through the use of call vectors and their associated commands. These call vectors are administered for both the sending and receiving switches:

- Sending Switch

  Vectors at the sending switch use conditional `goto` vector commands to test outflow threshold conditions, and `route-to` commands to send the call to another switch. The sending switch may provide alternate treatment if the call is denied at the receiving switch.

- Receiving Switch

  Vectors at the receiving switch use conditional `goto` vector commands to do inflow checking and decide whether the call should be accepted or denied. Call acceptance is accomplished when commands such as `queue-to main`, `check-backup`, `announcement`, `collect`, and `wait-time` are reached in the call vector at the receiving switch. (See Table 3-67 for more information on acceptance conditions.) Call denial is accomplished when commands such as `busy` and `disconnect after announcement none` are reached in the call vector at the receiving switch.

Two-Switch Look Ahead Interflow Configuration

An example of a Two-Switch Look Ahead Interflow configuration is shown in Figure 3-24. The operation of the sending switch and the receiving switch are described in the following paragraphs.

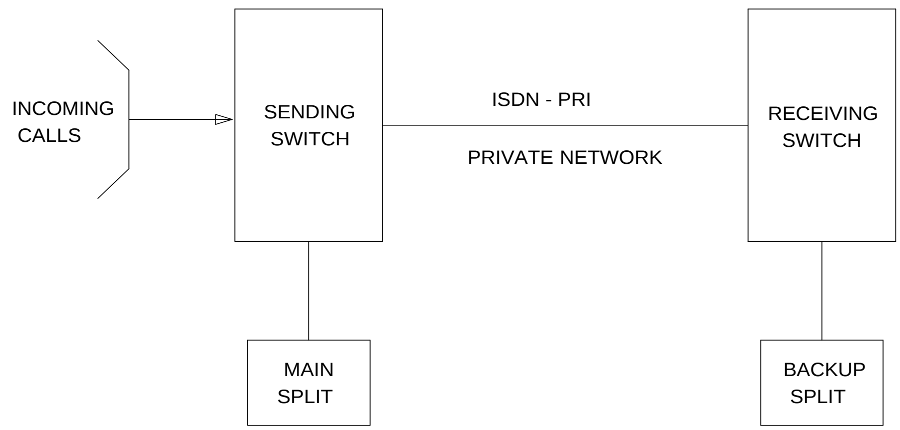


Figure 3-24. Two-Switch Look Ahead Interflow Connections

Sending Switch Operation

As with standard vectoring interflow, outflow checking and outflow is accomplished by means of conditional `goto` and `route-to` vector commands. There are no unique vector commands for this feature. If the Look Ahead Interflow option is enabled, and the call is being routed over an ISDN-PRI facility, interflow will automatically be carried out on a look ahead basis. (There is one exception to this rule. A `route-to with cov y` vector command will never result in a Look Ahead Interflow call.)

For Look Ahead Interflow calls, vector processing does not immediately terminate when ISDN-PRI facilities are successfully seized for a route-to operation. Instead, the call remains in any hunt group or split queues at the sending switch until it is accepted at the receiving switch. Any audible feedback initiated by the vector continues. If an agent becomes available at the sending switch during a Look Ahead Interflow call attempt, the caller is immediately connected to the available agent, and the Look Ahead Interflow call attempt is `dropped`.

The sending switch attempts to interflow the call to the receiving switch. We will assume a successful ISDN-PRI connection. One of the following then occurs.

- Call Acceptance

  If a call acceptance message is returned by the receiving switch before any call denial message, the sending switch will terminate vector processing, disconnect any tones applied by the sending switch vector and remove the call from all queues at the sending switch. Control of the call is now passed to the receiving switch.

- Call Denial

  If a call denial message is returned by the receiving switch before any call acceptance message, the sending switch will drop the Look Ahead Interflow call attempt and continue vector processing at the next vector step.

- Timeout

  If a call acceptance or call denial message is not returned from the receiving switch within 10 seconds after the receiving switch receives the Look Ahead Interflow call request, the Look Ahead Interflow attempt is dropped and the sending switch continues vector processing at the next vector step.

An example of an outflow vector used by a sending switch is as follows:

1. queue-to main split 3 pri m
2. announcement 1001
3. goto step 5 if oldest-call-wait in split 3 pri l > 30
4. wait-time 20 secs hearing music
5. route-to <a ten-digit number> with cov n if unconditionally
6. announcement 1002
7. wait-time 120 secs hearing music
8. goto step 6 if unconditionally

If split 3 has no available agents, Step 1 will place the caller in split 3's queue at medium priority. In Step 2, an announcement is played apologizing for the delay. Step 3 does the outflow checking. If calls have been queued up for longer than 30 seconds, the vector goes to Step 5 and does Look Ahead Interflow. Otherwise, the vector proceeds to Step 4 and music is played for 20 seconds. If the call is still not answered after 20 seconds, then the vector goes to Step 5 and attempts Look Ahead Interflow. In Step 5, a Look Ahead Interflow call is placed to a remote switch. If the call is accepted, the incoming call is removed from split 3's queue and control of the call is passed to the receiving switch. If the call is denied, the caller remains in queue and hears announcement 1002 followed by music.

Receiving Switch Operation

When the receiving switch receives the Look Ahead Interflow request, the call routes to a VDN, the VDN maps the call to the receiving switch's inflow vector and vector processing begins, starting with inflow checking. Inflow checking is accomplished with conditional `goto` commands in the inflow vector. The decision to accept or reject a call can be based on checks of the following:

- Number of staffed agents
- Number of available agents

- Time of day/week
- Number of calls in split's queue
- Number of seconds that the oldest call has been waiting in the split's queue
- Rolling Average Speed of Answer (G3V4 and later releases)
- Active VDN calls (G3V4 and later releases)
- Expected Wait Time (G3V4 and later releases)
- ANI (G3V4 and later releases)
- II-digits (G3V4 and later releases)

Once inflow checking is complete, acceptance of the look ahead call can be accomplished by executing any of the vector commands shown in Table 3-67.

Table 3-67. Call Acceptance Vector Commands and Qualifications

Call Acceptance Vector Commands	Qualification
announcement	Announcement available or Queued for announcement or Retrying announcement
check-backup	Call terminates to agent or Call queued for split
collect	Always
converse-on	VRU answers the call or Call queued to converse split
disconnect	With announcement and announcement available or With announcement and queued for announcement or With announcement and retrying announcement
messaging	Successful or Queued

Continued on next page

Table 3-67. Call Acceptance Vector Commands and Qualifications — *Continued*

Call Acceptance Vector Commands	Qualification
queue-to main	Call terminates to agent or Call queued for split
route-to	Terminates to valid local destination or Successfully seizes a non-PRI trunk or Results in Look Ahead Interflow call attempt and the call is accepted by the far end switch
wait-time	Always (except *wait-time* hearing i-silent which is neutral)

If, during inflow checking, the receiving switch decides that it is unable to accept the look ahead call, call denial can be accomplished by executing any of the vector commands listed in Table 3-68. Use of the `busy` command is recommended over the `disconnect` command to allow for compatibility with network services.

Table 3-68. Call Denial Vector Commands and Qualifications

Call Denial Vector Commands	Qualification
busy	Always
disconnect	With no announcement or With announcement but announcement unavailable

The vector commands shown in Table 3-69 do not generate either call acceptance or denial messages and are considered neutral.

Table 3-69. Neutral Vector Commands and Qualifications

Neutral Vector Commands	Qualification
adjunct routing	Always
announcement	Announcement unavailable
check-backup	The call neither terminates nor queues
converse-on	Call neither terminates nor queues
goto step	Always
goto vector	Always
messaging	Failure
queue-to main	The call neither terminates nor queues
route-to	Unsuccessful termination or Trunk not seized or Look Ahead Interflow call denied by far end switch
stop	Always
wait-time hearing i-silent	Always (New with G3V4)

The following is an example of an inflow vector used by a receiving switch.

1. goto step 6 if staffed-agents in split 12 is < 3
2. goto step 6 if queued-calls in split 12 pri l is > 4
3. queue-to main split 12 pri h
4. wait-time 30 secs hearing music
5. stop
6. busy

Steps 1 and 2 do inflow checking. For split 12, if the number of staffed agents is less than three or the number of calls in queue is greater than 4, call control is transferred to Step 6 which denies the look ahead call. Otherwise, Step 3 will queue the call for split 12 and return a call acceptance message to the sending switch.

Tandem Switch Configuration

Tandem Look Ahead Interflow can be accomplished with Call Vectoring and Look Ahead Interflow active at the receiving switch by using `route-to` commands that contain external destinations which use ISDN-PRI facilities.

An example of a tandem Look Ahead Interflow configuration is shown in Figure 3-25. The operation of the sending switch and the receiving switch are described in the following paragraphs.

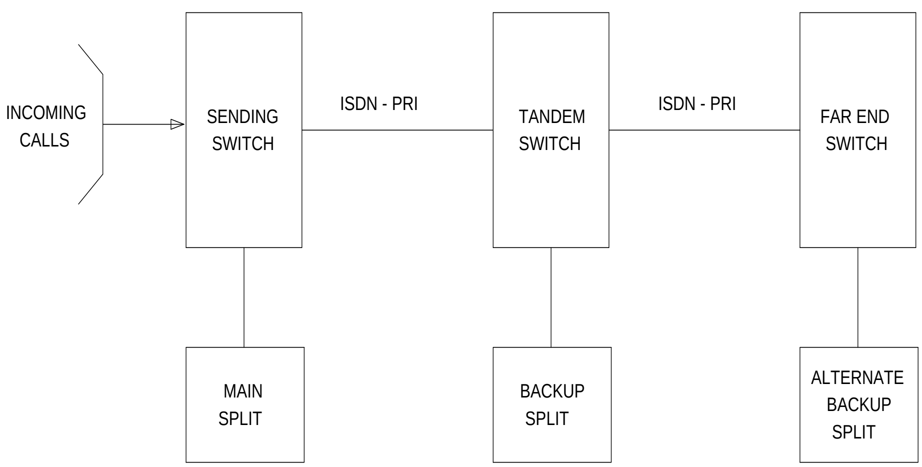


Figure 3-25. Look Ahead Interflow Using a Tandem Switch

Sending Switch Operation

The sending switch is unaware of the fact that its Look Ahead Interflow call is being tandemed to an alternate switch. The operation of the sending switch in the tandeming configuration is exactly as for the two switch configuration.

Tandem Switch Operation

If the receiving switch executes a `route-to` command that routes the call over an ISDN facility before call acceptance, the `route-to` command is performed on a ‘‘look ahead’’ basis in the same manner as a sending switch. If the call is accepted at the far end switch, acceptance is passed to the sending switch, and control of the call is passed to the far end switch with tandeming of the original calling party information and the original DNIS name. If the call is denied, the next step of the tandem switch vector is executed. If a denial is retained by the Tandem Switch to the originating switch, the next vector step on the originating switch is executed.

Great care should be taken by the vector programmer to ensure that the sending switch is not used as a backup location for the tandem switch or any of the far end switches. If administered in this manner, all trunk facilities could be tied up by a single call.

Example of Tandem Switch Vector

An example of a tandem switch vector follows:

1. goto step 4if staffed-agents in split 30 > 5
2. route-to <a 10-digit number> if unconditionally
3. busy
4. queue-to main split 30 pri m
5. announcement 2300
6. wait-time 60 secs hearing music
7. goto step 5 if unconditionally

Step 1 checks the inflow thresholds. If the inflow criteria is acceptable, Step 4(queue-to main) will provide acceptance to the sending switch. Therefore, Steps 5-7 provide a typical queuing-wait scheme. If the call cannot be handled at split 30, the `route-to` command checks another Look Ahead Interflow equipped switch on a "look ahead" basis. If the far end switch rejects the call, the `busy` command causes a denial to be sent to the sending switch. If the far-end switch accepts the call, then the acceptance is relayed back to the sending switch.

Far End Switch Operation

The far end switch is also unaware that tandeming has taken place. The far end switch will operate the same as in the case of the receiving switch in the two-switch configuration.

Display Information

Answering Agent's Display

The DNIS information in the Look Ahead Interflow information element provided by ISDN-PRI is presented on the answering agent's display on the receiving switch if the Look Ahead Interflow option is enabled, the call routes to a VDN, and the DNIS name is not blank.

The DNIS name included in the Look Ahead Interflow information element is as follows:

- If this is a tandemed look ahead call, then the Look Ahead Interflow DNIS information from the original look ahead call is used.
- If no redirection has taken place at the sending switch, the VDN name according to display override rules are used.

- If redirection has occurred, display override rules do not take effect. The Look Ahead Interflow DNIS will contain the original VDN name, or, if multiple VDNs are accessed, the name of the VDN last accessed by means of a `route-to` vector command.

VDNs that map to vectors which place Look Ahead Interflow calls must have their ISDN CPN prefixes administered. Failure to administer the ISDN CPN prefix results in a Look Ahead Interflow DNIS of all spaces being sent and displayed on the answering agent's terminal.

Originator's Display

On internal calls, the originator's display is the same as for normal Call Vectoring. However, the following precaution should be taken to prevent undesirable display updates from being received during Look Ahead Interflow call attempts.

Since most customers would not like the originator's display to be updated on each Look Ahead Interflow call attempt, Look Ahead Interflow calls should normally go out over trunk groups with the "Outgoing Display" field set to no.

Audible Feedback

Audible ringback is provided to the caller when a `wait-time hearing ringback` vector command is executed or when the call is successfully routed to a local destination.

Care must be taken by the vector programmer not to confuse callers by providing incompatible audible feedback at both the outflow and inflow vectors. For example, providing ringback at the receiving switch might be confusing to the caller if ringback and an announcement had previously been supplied by the sending switch.

Normally a vector for human callers should initially return audible feedback such as ringback, music, or an announcement so that the caller knows that the call got through. However, vectors which do not return any audible feedback may be used in situations where VDNs are accessed by machines and audible feedback may be inappropriate and/or undesirable.

The receiving switch vector does not initially give Look Ahead Interflow calls any locally denied termination treatment such as busy tone, reorder tone, or intercept tone. The caller does not hear these because feedback is still being provided by the sending switch vector.

Considerations

The receiving vector should not be administered to have calls route back to the vector for the outflowing switch. If administered as such, all trunk facilities may eventually be tied up with the same call.

All calls routed over ISDN-PRI facilities by means of `route-to number with cov n` and `route-to digits with cov n` vector commands on a switch where the Look Ahead Interflow option is enabled are treated as Look Ahead Interflow call attempts.

Initial audible feedback may be provided to the caller before interflow is attempted. Therefore, another audible feedback from the receiving switch may not be appropriate. For example, a caller hearing ringback on the sending switch may be confused if music is suddenly applied when the call interflows to the receiving switch.

Delay in interflowing should be minimized. An acceptance or denial response should be provided to the sending switch as quickly as possible by the vectors on the receiving switch.

If, during Look Ahead Interflow, the call terminates to an agent on the sending switch or the call is abandoned by the originator, the Look Ahead Interflow call is dropped, vector processing terminates, and the original call is removed from all split queues.

It is possible during a Look Ahead Interflow call attempt for a call to be accepted at a receiving switch, by means of a `queue-to main` or `check-backup` command, an instant before the call is answered at the sending switch. If the acceptance message is delayed due to signal propagation delay, there could be a short interval when the caller and the receiving switch agent are connected. An agent at the sending switch may then answer the call before the acceptance message arrives at the sending switch. The caller would then be disconnected from the receiving switch agent and connected to the sending switch agent. These phantom calls can be eliminated by appropriate programming of the inflow vector. If calls are accepted by `wait-time` or `announcement` vector commands before the call is queued to a split, there is no possibility of a phantom call occurring.

It is perfectly acceptable for a vector to route a call over an ISDN-PRI facility to a destination which is not a VDN. As far as the sending switch is concerned, this call will be treated as a Look Ahead Interflow call even though this is not in fact the case. Generic ISDN processing at the receiving switch will cause the call to be accepted. The DNIS name and any other information in the Look Ahead Interflow information element is ignored.

If a Look Ahead Interflow call terminates to a VDN on a receiving switch where the Look Ahead Interflow option is not enabled, intelligent interflow will result. However, the DNIS information in the Look Ahead Interflow information element is ignored and no intelligent interflow to far-end switches is possible.

Interactions

The following features interact with the Look Ahead Interflow feature:

- Attendant Control of Trunk Group Access

  Calls will not route over a trunk with Attendant Control of Trunk Group Access set.

- Authorization Codes

  Authorization Codes must not be required for interflow routing. The FRL assigned to the VDN should be high enough so that the route desired for routing interflow calls can be used without requiring an Authorization Code entry. If a route choice is encountered that requires a higher FRL, the interflow is considered an invalid destination (rejected for Look Ahead Interflow or not available for standard interflow) without the application of recall dial tone. Vector processing will continue.

- AAR

  ISDN-PRI facilities used to provide Look Ahead Interflow to a VDN on another switch in the customer's network can use the AAR feature if private facilities are to be used for call routing.

- ARS

  ISDN-PRI facilities used to provide Look Ahead Interflow to a VDN on another switch in the customer's network can use the ARS feature for call routing.

- ACD

  Look Ahead Interflow calls can be interflowed directly to ACD splits on remote switches which do not have Call Vectoring or Look Ahead Interflow enabled. In this case, the calls will be accepted or denied by generic call processing, even though full look ahead functionality will be lost. For example, an ACD split with a full queue would result in the call being rejected.

- Call Vectoring

  Call vectoring is required at the sending switch and the receiving switch.

  Call vectoring operates differently if Look Ahead Interflow is enabled at the sending switch. If ISDN-PRI facilities are successfully seized for a route-to operation, vector processing does not terminate until the call is accepted at the receiving switch.

  A forced disconnect at the receiving switch will not cause the originator's call to be dropped, but will only result in the failure of the Look Ahead Interflow attempt.

- CAS

  A centralized attendant can be a Look Ahead Interflow destination.

- DID

  The Look Ahead Interflow routing over ISDN-PRI facilities can enter the receiving switch on a ‘‘DID’’ basis. That is, a destination extension address (a VDN) is included in the ISDN SETUP message in the Called Number information element.

- Look Ahead EAS Interflow operates as described in this section for EAS and non-EAS environments.

- FRL and TCMs

  The FRL for interflow over ARS/AAR route choices is assigned to the original VDN used for the incoming call.

- Incoming Call Management

  The adjunct routing capabilities of vectoring can be used at the sending switch to determine if a call should be interflowed. Adjunct routing at the receiving switch can be used to tandem the call to a far-end switch.

- ISDN-PRI

  ISDN-PRI connectivity end-to-end over a private network is required for Look Ahead Interflow.

- Intercept Treatment

  No intercept treatment should be applied toward the caller as part of Look Ahead Interflow operation unless it results from a tandem connection on a non-Look Ahead Interflow basis. If the Look Ahead Interflow results in an intercept condition, the interflow should be rejected as part of the interflow vector programming, with alternate treatment provided at the sending switch.

- CDR (Sending Switch)

  No Ineffective Call Attempt or Outgoing Call CDR records will be generated for Call Vectoring `route-to` commands that are unsuccessful including denied Look Ahead Interflow attempts.

  If a local (on-switch) call to a VDN generates a Look Ahead Interflow call attempt that is accepted, and answer supervision is returned from the receiving switch, then one Outgoing Call CDR record is generated with the originating extension as the calling number.

  If an incoming (off-switch) call to a VDN generates a Look Ahead Interflow call attempt that is accepted, and no answer supervision is returned from the receiving switch, then one incoming CDR record will be generated. The VDN is the called number, and the duration is from the time answer supervision was provided to the incoming trunk.

  If an incoming (off-switch) call to a VDN generates a Look Ahead Interflow call attempt that is accepted, and answer supervision is returned from the receiving switch, then two incoming CDR records will be generated:

1. An incoming record with the VDN as the called number and the duration as the time since answer supervision was provided to the incoming trunk.
2. An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.

The sending of a Look Ahead Interflow information element is counted toward Message Associated User-to-User Information (MA-UUI) counts.

- CDR (Receiving Switch)

  On the receiving switch, an incoming Look Ahead Interflow call is treated like any other incoming vector call as far as CDR is concerned. An incoming CDR record is recorded on all calls which return answer supervision. The duration of the call is the time since answer supervision was returned.

  If answer supervision is returned by the vector (via an `announcement`, `collect`, `disconnect`, or `wait with music` command), and the call is never terminated to another destination, then the VDN extension is recorded as the called number in the CDR record.

  If the call terminates to a hunt group, then the VDN, hunt group, or agent extension is recorded as the called number. If the ‘‘Record VDN in Record’’ option of the Feature Related System Parameters is administered as ‘‘y,’’ then the VDN extension will override the ‘‘Call to Hunt Group - Record’’ administration option for Vector Calls.

  If the call terminates to a trunk (tandem), then two CDR records will be generated:

  1. An incoming record with the VDN as the called number and the duration as the time since answer supervision was provided to the incoming trunk.
  2. An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.

- Trunk-to-Trunk Transfer

  Interflowed calls may be transferred by a receiving switch agent to another trunk connection.

Administration

For full functionality, this feature requires the ISDN-PRI, Call Vectoring and Look Ahead Interflow options to be enabled at both the sending and receiving switches.

Private network ISDN implementations require Private Network access software in the sending and receiving switches. If the destination number is specified as a 10-digit public number, then ARS software could be used; otherwise, Private Network access software would be required.

"Call Vectoring" and ISDN-PRI ("Integrated Services Digital Network (ISDN) — Primary Rate Interface") must be administered as described in their separate feature descriptions discussed elsewhere in this chapter.

Hardware and Software Requirements

Existing ISDN-PRI hardware can be used for ISDN connectivity to the receiving switch. No new hardware is required. Interconnecting facilities must be ISDN-PRI with no interworking for the full capabilities of the feature to be operational. Look Ahead Interflow calls which interwork may interflow successfully but the ability to do so on an intelligent basis will be lost as will the Look Ahead Interflow DNIS information.

The ISDN-PRI, ETN (PNA), DCS or UDP, Call Vectoring, and Look Ahead Interflow features must be activated on both the sending and receiving switches. As indicated above, PNA or ARS will be required for call routing.

Loudspeaker Paging Access

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides attendants and voice terminal users dial access to voice paging equipment.

As many as nine individual paging zones can be provided by the system. (A zone is the location of the loudspeakers, for example, conference rooms, warehouses, or storerooms.) In addition, one zone can be provided by the system to activate all zones simultaneously.

Each of the 10 zones provided by the system is assigned an individual trunk access code. The trunk access codes are used to activate Loudspeaker Paging Access. A user can activate Loudspeaker Paging Access by dialing the trunk access code of the desired paging zone. In addition, the trunk access codes can be stored in Abbreviated Dialing lists. This allows multiappearance voice terminals to activate the feature via Abbreviated Dialing buttons. Attendants can use a Direct Trunk Group select button to activate Loudspeaker Paging Access, if the desired paging zone's trunk access code is assigned to one of the buttons.

Once a user has activated Loudspeaker Paging Access for the desired zone, the user can speak into the handset and make the announcement.

In addition to, or instead of the system loudspeaker paging equipment, a PagePac paging system can be used. A PagePac paging system has a distinct advantage over the switch paging system in that a PagePac system requires only one port on one circuit pack to provide as many as 39 paging zones. The switch paging system requires a separate port for each paging zone with a maximum of nine zones. Three different PagePac paging systems are available for use:

- PagePac 20

  This is the smallest PagePac system. The basic system provides a single paging zone with an input source for music over the paging system. The music can also serve as the music for the Music-on-Hold Access feature. Additional add-on hardware is available to provide multizone paging for 3, 9, or 39 paging zones.

- PagePac VS

  This system provides 1 to 3 paging zones. It also permits the paging of all zones simultaneously. Additional hardware is available to provide music and/or talkback over the paging system.

- PagePac 50/100/200

  This system provides up to 24 paging zones. Additional hardware is available to provide music and/or talkback over the paging system.

PagePac equipment is also easy to use. A user simply dials the extension number (PagePac 50/100/200 only) or trunk access code assigned to the PagePac system. This connects the user to the PagePac equipment. If there is only one paging zone, the user then uses the handset of the voice terminal to page someone. If there are multiple zones, the user, after hearing a steady tone, dials a one- or two-digit code to access the desired zone(s) before paging.

Considerations

With Loudspeaker Paging Access, a user can be paged at any location with loudspeaker paging equipment. This feature is particularly useful when used in conjunction with the Call Park feature. When a user is away from his or her location and receives a call, an incoming call can be answered and parked by another user. The called party can then be paged and told what extension number the call is parked on. The called party can then answer the parked call from a nearby voice terminal.

The system can have up to nine individual zones plus one zone to activate all zones simultaneously. A PagePac paging system can be used to provide up to 39 paging zones.

An LDN or DID call cannot be connected to the paging facility. However, the attendant can make the page and park the incoming call using the Call Park feature.

Interactions

The following features cannot be used with Loudspeaker Paging:

- Attendant Conference
- Terminal Conference
- Data Call Setup
- Hold
- Ringback Queuing
- Transfer

Normally, a call to a busy single-line voice terminal results in a call waiting tone being heard by the called voice terminal user. If that user is in the process of paging, the call waiting tone is not heard.

It is not possible to use a PagePac paging system for Code Calling Access when multizone paging is desired. The PagePac systems expect a two-digit code to access a particular zone. The system, however, immediately plays the chime code once a connection is established.

Administration

Loudspeaker Paging Access is administered by the System Manager. The following items require administration:

- The "Deluxe Paging" and "Call Park Timeout to Originator" field on the Feature Related System Parameters screen must be administered as no.
- Loudspeaker Paging Access buttons are assigned through the Attendant Direct Trunk Group Selection, Abbreviated Dialing, and Facility Busy Indication features.
- Trunk access codes and Class of Restriction (per zone provided).
- Paging expiration interval.
- CDR activation.

If a PagePac paging system is to be used, it must be assigned a trunk access code or extension number (PagePac 50/100/200 only).

If a PagePac paging system is accessed through a CO or analog trunk, administration is not done through the "Loudspeaker Paging Access" form. Instead, the line is accessed as a standard trunk (trunk access code of a CO) or a standard extension (dialed extension that connects to PagePac).

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) for each individual zone. Paging interface equipment, consisting of a 278A adapter and a 24 volt power supply, is also required for each individual zone. (This hardware can be shared with the Code Calling Access feature. Each feature is activated by the assigned trunk access code.) A TN417 is also required in Italy, the United Kingdom, and Australia.

If PagePac equipment is used, one port on a TN747 CO Trunk circuit pack, TN742 (TN746B supports A-law), TN769, or TN746B, Analog Line circuit pack, or TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) is required (depending on which PagePac system is used).

No additional software is required.

Loudspeaker Paging Access — Deluxe

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides attendants and voice terminal users with integrated dial access to voice paging equipment and Call Park capabilities.

When Loudspeaker Paging Access — Deluxe (also called Deluxe Paging in the remainder of this description) is activated, the call is automatically parked. The Call Park feature does not have to be activated separately. This is consistent with the Code Calling Access feature activation. In addition to the automatic Call Park capability, Deluxe Paging also lets parked calls return to the parking user with special distinctive alerting upon expiration of the Call Park Time-out interval.

Deluxe Paging also provides the ‘‘Meet-Me Paging’’ and ‘‘Meet-Me Conferencing’’ functions. With Meet-Me Paging, a user can simply activate Deluxe Paging, make the announcement for someone else to call him or her back and hang up. When the paged party answers, he or she is connected to the paging party. With Meet-Me Conferencing, another party can easily be paged and added onto a conference call.

The customer has the option of having either normal Loudspeaker Paging Access (discussed elsewhere in this document) or Loudspeaker Paging Access — Deluxe (Deluxe Paging). This description discusses only Deluxe Paging.

Paging Zones

As many as nine individual paging zones can be provided by the system.

(A zone is the location of the loudspeakers, for example, conference rooms, warehouses, or storerooms.) In addition, one zone can be provided by the system to activate all zones simultaneously.

Each of the 10 zones provided by the system is assigned an individual trunk access code. The trunk access codes are used to activate Deluxe Paging. A user can activate Deluxe Paging by dialing the trunk access code of the desired paging zone. In addition, the trunk access codes can be stored in Abbreviated Dialing lists. This allows multiappearance voice terminals to activate the feature via Abbreviated Dialing buttons. Attendants can use a Direct Trunk Group select button to activate Deluxe Paging, if the desired paging zone’s trunk access code is assigned to one of the buttons.

PagePac Paging

In addition to, or instead of the system loudspeaker paging equipment, a PagePac paging system can be used. A PagePac paging system has a distinct advantage over the switch paging system in that a PagePac system requires only one port on one circuit pack to provide as many as 39 paging zones. The switch paging system requires a separate port for each paging zone with a maximum of nine zones. Three different PagePac paging systems are available for use:

- PagePac 20

  This is the smallest PagePac system. The basic system provides a single paging zone with an input source for music over the paging system. The music can also serve as the music for the Music-on-Hold Access feature. Additional add-on hardware is available to provide multizone paging for 3, 9, or 39 paging zones.

- PagePac VS

  This system provides 1 to 3 paging zones. It also permits the paging of all zones simultaneously. Additional hardware is available to provide music and/or talkback over the paging system.

- PagePac 50/100/200

  This system provides up to 24 paging zones. Additional hardware is available to provide music and/or talkback over the paging system.

PagePac equipment is also easy to use. A user simply dials the extension number (PagePac 50/100/200 only) or trunk access code assigned to the PagePac system followed by the extension number where the call is to be parked. This connects the user to the PagePac equipment. If there is only one paging zone, the user then uses the handset of the voice terminal to page someone. If there are multiple zones, the user, after hearing a steady tone, dials a one- or two-digit code to access the desired zone(s) before paging.

Operations

User operations vary depending on the type of voice terminal the user has and whether or not the user is an attendant. Therefore, the various user operations are described separately for single-line voice terminals, multiappearance voice terminals, and attendants.

Activation of Deluxe Paging by Single-Line Voice Terminal Users

1. Go off-hook to get dial tone.

   If already on a call with another party, press the Recall button or flash the switchhook. The other party is placed on hold and recall dial tone is heard.

2. Dial the trunk access code for the desired paging zone. (Dial tone is heard.)

3. Dial the extension number where the call is to be parked. (Confirmation tone is heard and the call is temporarily parked.)

   To park the call on your own extension, dial a “#” instead of the extension number.

4. Make the announcement. (The loudspeaker paging timer starts.)

5. Press the Recall button before the administered loudspeaker paging timeout interval expires and go on-hook. (The paging equipment is released, the parked call is now waiting to be answered, and the timer for the Call Park Time-out interval starts.)

   If another party was on the call and was placed on hold in Step 1, that party and the paging party are in conference, parked on the call, and connected to the paged party when he or she answers the call. (This is known as Meet-Me Conferencing.)

   If the loudspeaker paging time-out interval expires before the Recall button is pressed, the paging user receives confirmation tone, the paging equipment is released, the call is automatically parked on your extension, and the calling party hears music (if provided). When the paged party answers the call, he or she is connected to the paging party. The paging party can then transfer the call to the calling party.

   If the Call Park Time-out interval expires, the call returns to the paging user with the proper distinctive alerting (One-burst for internal calls and conference calls with both internal and external parties; Two-burst for external calls). If unanswered, the call follows the coverage path of the paging user.

   If no answer-back is required on the call, hang up instead of pressing the Recall button. The parked call is dropped and the paging equipment is released.

Activation of Deluxe Paging by Multi-Appearance Voice Terminal Users

1. Go off-hook to get dial tone.

   If already on a call with another party, press the Transfer button. The other party is placed on hold and dial tone is heard.

2. Dial the trunk access code for the desired paging zone. (Dial tone is heard.)

3. Dial the extension number where the call is to be parked. (Confirmation tone is heard and the call is temporarily parked.)

   To park the call on your own extension, dial a “#” instead of the extension number.

4. Make the announcement. (The loudspeaker paging timer starts.)

5. Press the Transfer button before the administered loudspeaker paging timeout interval expires and go on-hook. (The paging equipment is released, the call is parked and is now waiting to be answered, and the timer for the Call Park Time-out interval starts.)

   If another party was on the call and was placed on hold in Step 1, that party is parked on the call and is connected to the paged party when he or she answers the call (answer-back). The Conference button can be pressed instead of the Transfer button to allow both the paging and held parties to be connected to the paged party on answer-back. (This is known as Meet-Me Conferencing.)

   If the loudspeaker paging time-out interval expires before the Transfer button is pressed, the paging user receives confirmation tone, the paging equipment is released, the call is automatically parked on your extension, and the calling party hears music (if provided). When the paged party answers the call, he or she is connected to the paging party. The paging party can then transfer the call to the calling party.

   If the Call Park Time-out interval expires, the call returns to the paging user with the proper distinctive alerting (One-burst for internal calls and conference calls with both internal and external parties; Two-burst for external calls). If unanswered, the call follows the coverage path of the paging user.

   If no answer-back is required on the call, hang up instead of pressing the Transfer or Conference button. The parked call is dropped and the paging equipment is released.

Activation of Deluxe Paging by an Attendant for Another Party

1. Press the Start button. The other party is placed on hold and the attendant gets dial tone.
2. Dial the trunk access code for the desired paging zone. (The attendant gets dial tone.)
3. Dial the extension number where the call is to be parked. (Confirmation tone is heard and the call is temporarily parked.) The attendant can also dial a ‘‘#’’ instead of the extension number to park the call on his or her individual attendant extension (if assigned).
4. Make the announcement. (The loudspeaker paging timer starts.)
5. Press the Release button before the administered loudspeaker paging timeout interval expires. (The paging equipment is released, the parked call is now waiting to be answered, and the timer for the Call Park Time-out interval starts. If the Split button is pressed, the timer for the Call Park Time-out interval starts, and both the held party and the attendant are parked and is connected to the call upon answer-back.)

   If the loudspeaker paging time-out interval expires before the Release button is pressed, the attendant receives confirmation tone, the paging equipment is released, and the call is automatically parked on the console.

If the loudspeaker paging timeout interval expires before the Release button is pressed, the attendant receives confirmation tone, the paging equipment is released, the call is automatically parked on the console, and the calling party hears music (if provided). When the paged party answers the call, he or she is connected to the paging party. The paging party can then transfer the call to the calling party.

Activation of Deluxe Paging Answer-Back by the Paged Party

1. Go off-hook to get dial tone.
2. Dial the answer-back feature access code. (Dial tone is heard.)
3. Dial the extension number where the call is parked, or dial ‘‘#’’ if the call is parked on the extension you are using. (Music-on-Hold, if provided, is removed from the parked call, all parties associated with the parked call receive confirmation tone, and the answer-back and parked parties are connected.)

Unparking a Loudspeaker Paging Call

If a user wishes to unpark a loudspeaker paging call that is parked on his or her extension, this can be accomplished by pressing the lighted Call Park button.

Considerations

With Loudspeaker Paging Access — Deluxe, a user can be paged at any location with loudspeaker paging equipment. Integrated Call Park capabilities allow the paging party to park the call without dialing a separate Call Park feature access code. When a user is away from his or her location and receives a call, an incoming call can be answered by another user. The called party can then be paged and told what extension number the call is parked on. The called party can then answer the parked call from a nearby voice terminal.

The system can have up to nine individual zones plus one zone to activate all zones simultaneously. A PagePac paging system can be used to provide up to 39 paging zones.

An LDN or DID call cannot be connected to the paging facility. However, the attendant can make the page and park the incoming call.

For non-local access of Deluxe Paging (such as Remote Access users, tie trunk users, and so on) the ‘‘#’’ cannot be used to park the call on your own extension.

Interactions

The following features interact with the Loudspeaker Paging Access — Deluxe feature.

- Bridged Call Appearance

  If the parked call includes a shared TEG, a shared PCOL, and/or a redirected call with a Temporary Bridged Appearance the maximum number of off-hook parties on the call is five, instead of six. The sixth position is reserved for the answer-back call.

- Call Coverage

  If a coverage call is parked by Deluxe Paging, the Temporary Bridged Appearance at the principal extension is maintained as long as the covering user remains off-hook or places the call on hold.

- Call Park

  A call cannot be parked on more than one extension at the same time.

  More than one call cannot be parked at the same extension at any given time. If a user tries to park a Deluxe Paging call on an extension that already has a parked call, that user receives intercept treatment.

  The Call Park feature provides up to 10 (40 for G3r) common shared extensions for use by attendants or by voice terminal extensions with console permissions. These extension numbers are not assigned to a voice terminal, but are stored in system translations and used to park a call. These extension numbers are particularly useful when one party is paged at the request of another party. The calling party is parked by Deluxe paging and the extension number is announced. Common shared extensions should be assigned to the optional selector console in the 00 through 09 (00 through 39 for G3r) block (bottom row) in any hundreds group that the attendant can easily identify. The lamp associated with the extension number identifies call parked or no call parked (instead of active or idle status).

  If the Call Park Time-out interval expires during Deluxe Paging, the call normally returns to the originator of the Deluxe Paging call. However, with Remote Access and Tie Trunk Access, the call goes to the attendant. If unanswered, the call follows the coverage path of the paging user.

- Call Pickup

  If a Call Pickup call is parked by Deluxe Paging, the Temporary Bridged Appearance at the principal extension is maintained as long as the answering pickup group member remains off-hook or places the call on hold.

- Call Waiting Termination

  Normally, a call to a busy single-line voice terminal results in a call waiting tone being heard by the called voice terminal user. If that user is in the process of paging, the call waiting tone is not heard.

- Code Calling Access

  It is not possible to use a PagePac paging system for Code Calling Access when multizone paging is desired. The PagePac systems expect a two-digit code to access a particular zone. The system, however, immediately plays the chime code once a connection is established.

- Conference — Attendant

  The maximum number of conferees on a parked Deluxe Paging call is five. The sixth conferee position is reserved for the answer-back call.

  A Deluxe paging call cannot be conferenced unless a party was placed on hold and parked with the call. The reason for this is that paging equipment cannot be placed on hold.

- Conference — Terminal

  For multiappearance voice terminals, the maximum number of conferees on a parked Deluxe Paging call is five. The sixth conferee position is reserved for the answer-back call.

  Single-line voice terminals can have a maximum of two conferees on a parked Deluxe Paging call.

  A Deluxe paging call cannot be conferenced unless a party was placed on hold and parked with the call. The reason for this is that paging equipment cannot be placed on hold.

- Data Call Setup

  If the Data button has been pressed as a pre-indication for modem pooling, access to Deluxe paging is denied.

- Data Privacy

  If a call, with Data Privacy activated, is parked by Deluxe Paging, Data Privacy for that call is automatically deactivated.

- DID

  A DID call cannot be connected to a Deluxe Paging facility.

- Hold

  Deluxe Paging facilities cannot be placed on hold.

- Hunt Groups

  If a hunt group member parks a call using Deluxe Paging, the call is parked on the member's own individual extension, not the hunt group extension.

- LWC

  If a user parks a call for his or her extension with the Conference button, any parking or parked parties cannot activate LWC because that party cannot be uniquely identified.

- Manual Originating Line Service

  Users assigned Manual Originating Line Service cannot access Deluxe Paging.

- Music-On-Hold Access

  Music-On-Hold, if provided, is connected to the parked party when there is only one conferee left on the parked call. Music-On-Hold is not connected to a parked call with more than one conferee.

- Multiple LDNs

  An LDN call cannot be connected to a Deluxe Paging facility.

- Night Service

  If a night station user parks a Night Service call with Deluxe Paging, the call is parked on the night station's primary extension.

- PCOL

  If a PCOL call is parked by Deluxe Paging, the Temporary Bridged Appearance of the call is maintained at the PCOL extension until the call is disconnected.

- Remote Access

  Remote Access users can access Deluxe Paging unless they are restricted by COR from doing so.

- Ringback Queuing

  Ringback Queuing is not provided for Deluxe Paging.

- TEG

  If a TEG member parks a call using Deluxe Paging, the call is parked on the member's own individual extension not the TEG extension.

- Transfer

  A Deluxe paging call cannot be transferred unless a party was placed on hold and parked with the call. The reason for this is that paging equipment cannot be placed on hold.

Administration

Deluxe Paging is administered by the System Manager. The following items require administration:

- The "Deluxe Paging" and "Call Park Timeout to Originator" field on the "Feature Related System Parameters" form must be administered as yes.
- Up to 10 (one per zone) Deluxe Paging buttons (per multiappearance voice terminal and attendant console). Buttons are assigned through the Attendant Direct Trunk Group Selection, Abbreviated Dialing, and Facility Busy Indication features.

- Trunk access codes and Class of Restriction (per zone provided).
- Answer-back access code.
- Paging expiration interval (from 10 seconds to 10 minutes).
- Call Park expiration interval (from 1 to 90 minutes in intervals of one minute).
- CDR activation.
- Console permissions to allow voice terminal extensions to park calls on common shared extensions (assigned via COS).
- Up to 10 common shared extension numbers.

If a PagePac paging system is to be used, it must be assigned a trunk access code or extension number (PagePac 50/100/200 only).

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) for each individual zone. Paging interface equipment, consisting of a 278A adapter and a 24-volt power supply, is also required for each individual zone. (This hardware can be shared with the Code Calling Access feature. Each feature is activated by the assigned trunk access code.) A TN417 is also required in Italy, the United Kingdom, and Australia.

No additional software is required.

Malicious Call Trace (MCT)

Feature Availability

This feature is available with Generic 3r V1 and optional with Generic 3 V2 and later releases.

Description

MCT provides a way for terminal users to notify a predefined set of users that they may be party to a malicious call. This set of users may then retrieve certain information related to the call. Based on this information, it is possible to identify the source of the malicious call or be capable of providing useful information to personnel at an adjacent switch to aid in the tracking down of the call's source. This feature also provides a method of generating an audio recording of the call.

Airlines, utility companies, government offices and major corporations are a few examples of customers that may experience malicious calls. Access to a feature such as MCT may allow them to deter or respond appropriately to bomb threats, product tampering, threats or other possible malicious calls.

MCT enables a user to collect information which could be used to identify the calling party. There are three different sets of MCT information that are collected (and subsequently displayed) depending on whether the malicious call originated within the system or outside of it.

- If the call originates within the system (the same node or DCS subnetwork), the calling number is displayed on the controlling terminal.
- If an ISDN calling number identification is available on the incoming trunk, then the calling number is displayed.
- Otherwise, the incoming trunk equipment location is displayed.

In the latter case it will be necessary for the customer to call the appropriate connecting switch. For all three types of calls, the controlling terminal will also display other information including the called number, activating number, whether the call is active or not, and, if appropriate, identification of any additional parties that may be on the call.

During the alerting of a MCT-Controller, that controller's display will show the message ‘‘MALICIOUS CALL TRACE REQUEST.’’ Incoming calls to the station while this message is being displayed will not provide the incoming caller's display information as it normally would.

Any action from the set itself that normally would impact the station's display will still be valid. For example, when a user goes off-hook to dial a number, the display may read ‘‘a=’’ then it will display the digits the person enters. This still operates in this manner. Provided there is a call to trace (or that is actively being traced), when the user's display is in NORMAL mode, pushing the MCT-Control button will display the next page of MCT-information for the controller regardless of the previous state of the display.

A station becomes the official controller of a particular trace by being the first person to push the MCT-Control button when its display reads ''MALICIOUS CALL TRACE REQUEST''. Once this has happened, the subsequent depressions of this button will cause the following pages of MCT-information to be displayed:

Page 1 ''MALICIOUS CALL TRACE REQUEST''

Page 2 ''MCT activated by: 1002 for: 51001''

Page 3 ''original call redirected from: 52001''

Page 4 ''voice recorder port: 01C0104:

Page 5 ''party2: 01C1505 <PORT ID>'' or ''party2: 01C1505 <ISDN PORT ID>''

Page 6 ''party3: 52001 <EXTENSION>:'' or ''party3: 52001 <ISDN PORT ID>''

Page 7 ''END OF TRACE INFORMATION''

The complete trace activity for MCT can be divided into three distinct phases:

- Feature activation
- Feature control
- Feature deactivation

Feature Activation

While the recipient of the malicious call is actively talking on a suspected malicious call, there are four ways that the MCT feature can be activated:

1. The Recipient pushes a MCT-Activate button previously administered on his or her set.
2. The MC-Recipient places the call on HOLD, gets a second call appearance, then dials the MCT-Activate feature access code. Upon hearing second dialtone, the MC-Recipient dials the extension number assigned to their set, a # sign, or they wait for timeout (10 seconds). For obvious reasons, this is not the recommended activation approach.
3. The Recipient signals a coworker to activate the feature for them on another set. The coworker would push the MCT-Activate button to dial the feature access code. Upon hearing dialtone, the coworker dials the extension on which the Recipient is talking.
4. Under certain circumstances, a MCT-Controller on switch A may request a controller on switch B to continue tracing a call that was tandemed through switch B. In this case the controller on switch A would supply the

controller on switch B with the trunk member port id they wish to have traced. The controller on switch B would then activate the feature by pushing a MCT-Activate button or dialing the FAC and upon hearing second dialtone, enter a ‘‘*’’ character followed by the trunk port id. For example, trunk port id 01C0401 would be entered as 0130401. The letters A through E of a port id are entered as 1 through 5 on the station keypad.

Once the MCT feature has been activated, information associated with the call is stored. Also, a group of previously administered extensions will begin alerting. Each should have been previously administered with an MCT-Control button. In addition, if a MCT-Recorder has been administered for your system and is available, it will be attached to the suspected malicious call.

Feature Control

After being alerted of an MCT activation, the first controlling terminal to respond by pushing an administered MCT-Control button will be considered the controlling party for the call. Alerting on the rest of the set of controlling terminals, if any, will stop.

The initial MCT-Control button push generates a display identifying the called party because both the called extension is displayed. Subsequent button pushes will cycle through the MCT information. The complete set of MCT information may be displayed repeatedly by continued pushes of the button.

In the case of a malicious call involving a non-ISDN trunk, the controlling party would call the personnel at the connecting switch during the feature control stage to request that a trace be performed. For this to be feasible, the controlling party should have the telephone number for the connecting switch available as well as a cross-reference of the system’s trunk-port numbers (including DS1 channel number, if appropriate) versus the trunk equipment locations at the connecting switch.

Feature Deactivation

The controlling party may at any time (but usually after completing the trace) complete the MCT feature activity by dialing the MCT-Deactivate FAC. This deactivation frees any resources involved in the current trace activity that were blocked by the MCT feature and, if appropriate, disconnects the MCT Voice Recorder from the call.

The following are the steps in the MCT sequence:

- MCT-Activate FAC — When followed by the extension receiving the malicious call, this initiates the MCT sequence. When followed by pressing the asterisk button on the dial pad followed by the port id, this makes the user the MCT Controller for a call that may involve this particular trunk member.

- MCT-Activate Button Push — This initiates an MCT sequence with the assumption that a malicious call is currently active on the terminal generating the MCT-Activate button push.
- MCT-Control Button Push — When generated from a terminal with a display, this tracks the progress of the trace information during an MCT sequence.
- MCT-Deactivate FAC — When entered from a controlling terminal, this ends an MCT sequence. This FAC will be accepted only when entered from a currently active controlling terminal.

Considerations

An MCT feature activation does not generate a call within the system to alert the controlling terminals. Since the alerting is not a call, it is not impacted by the existence or length queues associated with the controlling terminals.

There are, however, still interactions with real calls at the alerted terminals and these are noted in the next section. If a trunk is involved in the MCT, this resource will be blocked from being dropped from the switch-side to facilitate the tracing activity. Any terminal involved in the call will not be blocked from dropping.

Display Information

Although this is not a feature in and of itself, it is included here to point out that display information may be affected by MCT. With the exception of Emergency Access to the Attendant, any feature that would normally provide display information will not do so on a terminal acting as the controlling terminal. The remainder of that feature's functionality would remain the same only its ability to display information will be blocked. This remains in effect until the terminal performs an MCT-Deactivate.

Switch Failure

MCT information relating to a currently active malicious call is lost during a switch failure.

Interactions

The following is a description of how certain features behave differently when used in conjunction with an MCT activity.

- Attendant Interposition Calling and Transfer

  Provided that the console is not busy, an attendant who is acting as a MCT controller may receive alerting call, but that attendant will not receive the display information.

- Bridged Appearance — Temporary

  If a covering user has activated a trace, the original called party may bridge on to the call provided the normal rules governing such.

- Bridged Call Appearance

  If the user of a primary extension is the Recipient of the indication call, then a station with a bridged appearance of this extension may bridge onto the call according to the normal rules governing such. For an MCT-Activate button push, if the currently active extension is a bridged appearance, the system will record the primary extension as the MCT recipient.

  For an MCT-Activate FAC, the user should dial the number of the terminal with the bridged appearance that is actually on the call, instead of the bridged number. However, the system will log the primary extension as the Recipient. Likewise, for self-originated MCT activations via the FAC, the system will log the primary extension as the MCT Recipient provided that the recently ‘‘held’’ appearance is a bridged appearance. (When performing an MCT-Activate operation for one’s self, the ability to dial ’#’ or wait for interdigit timeout is still valid.)

- Busy Verification of Terminals and Trunks

  Sometimes while tracing a malicious call, the MCT Controller may need to contact a potential MCT Controller on an adjacent switch through which the call may have tandemed. That person, once involved in the call, would then perform an MCT activation.

  This introduces potential problems:

  — The malicious caller would hear a warning tone as a result of the intrusion.

  — The person performing the MCT Activate is not guaranteed to be the same person that will perform the associated MCT Control operation thereby, losing the ‘‘continuity’’ of tracing this call because of a trace on another switch.

  For G3r, this will be handled by allowing a user to enter an asterisk on the dial pad followed by the port id, during the second dialtone after an MCT Activate. In such a case, in order for the controller on the adjacent switch to get the appropriate MCT information, that person will have to enter onto the call via the Busy Verification feature. Furthermore, such an intrusion via busy verification results in a warning tone being applied which alerts all members on the call of the intrusion. The intrusion tone will not be applied if the MCT Controller on the adjacent switch activates busy verification immediately after generating a MCT Activate but before the routing timeout occurs.

- Call Waiting

  A call may not wait on the line of a Recipient indication call. Instead, the MCT Recipient’s line will appear busy.

- Conference

  The conference operation can be used by an MC-recipient to temporarily place a malicious caller on hold. The MC-recipient may choose to do this in order for MCT-Activate to generate a stimulus via a FAC.

  This is done by the user initiating the conference operation, receiving the second dialtone, entering the MCT-Activate FAC then halting the remainder of the conference operation and simply returning to the malicious caller's appearance.

  MCT-Activate can be generated for a member of a conference. As a point of clarification, the MCT-Activate is not affected by the number of parties involved in the conference.

- DCS

  If a voice terminal within the DCS network is involved in a malicious call, then the extension number will be recorded and subsequently displayed with the MCT information as appropriate. Beyond this functionality, DCS transparency is not provided for MCT. MCT-Activate, MCT-Control and MCT-Deactivate activities must be performed by terminals within the same DCS node.

- Emergency Access to the Attendant

  This feature continues to operate as expected however, it can affect how an MCT-Control operation works. Ordinarily, during an MCT-Control operation, no other feature may access the display of the terminal acting as the controlling terminal. However, if an Emergency Access call terminates at a console acting as the controlling set, MCT will relinquish control of the display until this call has been completed then MCT will again maintain exclusive control over the display.

- Individual Attendant Access

  An attendant acting as a controlling terminal (provided the attendant has unbusied the console) may receive alerting for a call to the individual attendant access extension but, the display information will not be displayed.

- Make-Busy

  Make-Busy activations are attempted by the switch for any station or console generating an MCT-Control stimulus. The attempt will be made for Make-Busy button(s) administered on the station or console. The success or failure of the attempt is determined by the normal rules governing Make-Busy activation.

  When this station generates an MCT-Deactivate stimulus, the state of the Make-Busy feature for this set will be returned to whatever the state was prior to the MCT-Control stimulus.

- Personal Central Office Line (PCOL)

  If a PCOL is involved in an MCT, then that trunk may be held up by the switch until the MCT-Deactivate has been processed.

- Position Busy

  A Position Busy activation will be attempted by the switch for any attendant in the role of MC-recipient activating PB. The success or failure of the attempt will be determined by the normal rules governing a Position Busy activation.

  When this station generates an MCT-Deactivate stimulus, the state of the Position Busy feature for this set will be returned to whatever the state was prior to the MCT-Control stimulus.

- Priority Calling

  A priority call attempt towards the MCT Recipient will be denied.

- Centralized Attendant Service

  An MCT-Activate processed on a branch PBX cannot be controlled by an attendant at the main PBX. The MCT controlling activity will have to be processed by a controlling terminal at the branch PBX. MCT-Activate, MCT-Control, and MCT-Deactivate activities must be performed by terminals within the same PBX.

- R2 Multi-Frequency Code

  MCT does not use R2 Multi-Frequency Code (MFC) signaling.

- Send All Calls

  If users' stations activating MCT-Control have a Send-All-Calls (SAC) button administered on their sets, then the SAC will be activated. This provides consistency with the automatic Position Busy activation that occurs when an attendant activates the MCT-Control button. Like the Position Busy, the SAC will be an activation attempt whose success or failure is dependent on the normal rules governing such.

  When this station-user generates an MCT-Deactivate stimulus, the state of the Send All Calls feature for this set will be returned to whatever the state was prior to the MCT-Control stimulus.

- Transfer

  If the MC-Recipient transfers a malicious call, the MCT information (displayed on the controlling terminal and saved in the MCT history) will still identify the transferring party as the MCT Recipient.

  Also, the Transfer operation can be used by an MC-Recipient to temporarily place a malicious caller on hold. The MCT Recipient may choose to do this in order to send an MCT-Activate stimulus via a FAC.

  This is done by the user initiating the Transfer operation, receiving the second dialtone, entering the MCT-Activate FAC, then halting the remainder of the Transfer operation and simply returning to the malicious caller's appearance.

Administration

The following describes administration.

- COR Administration

  This provides a user access to MCT features.

- FAC Administration

  This provides Feature Access Codes with which MCT-Activate and MCT-Deactivate may be performed.

- Station Administration

  This provides a user access to the MCT-Activate and/or MCT-Control buttons.

- System Parameters Administration

  This determines application of a tone audible to all parties during the recording of a malicious call and trunk group where the port id of the recorders are listed. Also, with this administration, the administrator is provided with the set of controlling terminals and the assignment of ports for MCT voice recorders. This is done on the “MCT Groups — Extension” form.

- MCT History Administration

  A user may see a history of the last 16 trace activities.

- System-initiated Operations

  For completeness, the following are the system initiated operations for MCT:

  — Conversation Recording — After processing a MCT-Activate button push or FAC, the system will attach a MCT Voice Recorder to record the conversation.

  — Historical Recording — After processing a MCT-Activate button push or FAC, the system will record the MCT-information to allow subsequent retrieval of the data via the “MCT History Administration” form.

- Capacities

  Number of Malicious Calls being Traced:

  — Up to 16 MCTs may be active in the system at any given time. This maximum includes currently controlled calls and calls in queue for being controlled.

  Number of Voice Recorders:

  — Only one MCT Voice Recorder may be used for each malicious call being controlled or queued for being controlled. (This also implies that the system maximum for MCT Voice Recorders is 16 since that is the maximum number of allowable active MCTs).

  Number of Controlling Terminals:

  — Up to100 terminals may be identified as controlling terminals.

Hardware/Software Requirements

Below is listed the set of hardware which may be used to support the MCT feature:

- Stations with displays
- Attendant consoles
- MCT voice recorders
- Auxiliary trunks (278A Adapter in conjunction with a TN763 or SN231)
- Trunks

Manual Message Waiting

Feature Availability

This feature is available with all Generic 3 releases.

Description

Enables multiappearance voice terminal users, by pressing a designated button on their own terminals, to light the status lamp associated with the Manual Message Waiting button at another multiappearance voice terminal. Activating the feature causes the lamp to light on both the originating and receiving voice terminals. Either terminal user can cause the lamp to go dark by pressing the button.

Considerations

This feature can be administered only to pairs of voice terminals such as a secretary and an executive. The secretary might press the designated button to signal the executive that a call needs answering. The executive might press the button to indicate `Do Not Disturb` or `Not Available` to the secretary. (The button can be marked to reflect the intended use.)

Interactions

None.

Administration

Manual Message Waiting is administered on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Manual Message Waiting buttons to the voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Manual Originating Line Service

Feature Availability

This feature is available with all Generic 3 releases.

Description

Connects single-line voice terminal users to the attendant automatically when the user lifts the handset. The attendant code is stored in an Abbreviated Dialing list. When the Manual Originating Line Service voice terminal user lifts the handset, the system automatically routes the call to the attendant using the Hot Line Service feature.

A Manual Originating Line Service user can receive calls allowed by the assigned COR. Call reception is not affected by Manual Originating Line Service.

Considerations

Manual Originating Line Service is useful in any application where all call originations are screened by the attendant. The user simply lifts the handset and is connected to the attendant.

The number of single-line voice terminals that can be assigned Manual Originating Line Service is not limited.

Interactions

A Manual Originating Line Service call is a Hot Line Service call to the attendant. A voice terminal user cannot activate features that require dialing. When a Night Service feature is activated, the Manual Originating Line Service call is redirected.

Administration

Manual Originating Line Service is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Abbreviated Dialing Lists (the attendant code must be a list entry)
- Hot Line Destination

Hardware and Software Requirements

No additional hardware or software is required.

Manual Signaling

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows a voice terminal user to signal another voice terminal user. The receiving voice terminal user hears a two-second burst of tone.

The signal is sent each time the button is pressed. If the receiving voice terminal is already being alerted with an incoming call, Manual Signaling is denied. The status lamp associated with the Manual Signaling button at the originating voice terminal flutters briefly to indicate the denial.

Considerations

With Manual Signaling, one voice terminal user can signal another voice terminal user. The meaning of the signal is prearranged between the sender and the receiver.

When a voice terminal user presses the Manual Signaling button, the associated status lamp lights for two seconds.

Interactions

None.

Administration

Manual Signaling is assigned on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Manual Signaling button to the originating voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

MERLIN®/System 25 Voice Terminal Support — 731xH Series

Feature Availability

This feature is available with all Generic 3 releases except G3i-Global and G3rV1.

Description

The feature allows MERLIN/System 25 customers to reuse the 7309H and the following 731xH series ATL (hybrid) voice terminals with the DEFINITY Communications System G3vsV1/G3sV1, G3iV1, G3V2, and later releases:

- 7313H
- 7314H
- 7315H (with built-in display)
- 7316H
- 7317H (with built-in display)

All of the above voice terminals connect to the DEFINITY switch via existing TN762B hybrid circuit pack and are supported in native mode. However, not all fixed-feature buttons on these terminals function completely when connected to the DEFINITY switch.

Both the 7315H and 7317H are equipped with integrated 16-character LCD displays

- Use the `disp-norm` button to toggle between `normal` call display mode (to display incoming and outgoing call data) and `local` voice terminal mode (to display the voice terminal supplied date and time).

  NOTE:
  When in local mode, the voice terminal can receive incoming calls or you can go ‘‘off hook’’ and dial an outgoing call. In these cases, the voice terminal automatically changes to normal call display mode.

- Displayed Messages consist of one or two 16-character lines. The ‘‘scroll’’ button is used to toggle between the two lines of a display. (For one-line displays, pressing the ‘‘scroll’’ button has no effect.)

NOTE:
For redirected calls sometimes the name is truncated at 14 characters and is followed by a space and ‘‘<’’ character. The ‘‘<’’ character indicates that the call was originally directed to the named party but has been redirected

to this station (for example, using call forwarding or call coverage). Scroll to the second line of the display to view the name of the calling party.

The 7315H and 7317H voice terminals display information differently from other voice terminal displays. For example, the 7315H and 7317H do not identify which call appearance is in use. Another example is that reason codes for calls direct or redirected to the station are restricted to one character codes. These codes may be different from what is seen on digital displays. Multiple character reason codes are changed to one character reason codes for call redirection.

Table 3-70 lists the redirection reason codes for the two different terminal types:

Any that is blank means that there is no flag that applies and/or the feature does not route the call to the respective voice terminal.

Table 3-70. Merlin Terminal Codes

Call Type	**Digital**	**Hybrid**
Send-All-Calls to Coverage	s	s
Coverage Path Criteria: Cover-All	c	C
Coverage Path Criteria: Don't-Answer	d	d
Coverage Path Criteria: Busy/Active	b	b
Return-Call to Attendant	rt	
Recall-Call	rc	
Redir Attendant Control Trunk Group Access	tc	
Redir by intercept treatment to Attendant	ic	
Priority Call	priority	!
Call Forward	f	f
Intercom Call	ICOM	i
Night Station Service Call	n	n
Call Pick-up	p	P
DID Trunk Calls to LDN	ld	#
Call-Park Time-Out	ic	p
Transfer Call Time-Out	rt	
Consult to Covered Principle-Station	priority	!

Continued on next page

Table 3-70. Merlin Terminal Codes — ***Continued***

Call Type	Digital	Hybrid
Control Outward Restriction	co	r
Control Station to Station Restriction	cs	r
Automatic Call-Back Call	callback	Q
ACD Hunt Group Supervisor Assist	sa	S
Control Termination Restriction	ct	r
Emergency Call to Attendant Queue Full Redirection	qf	E
All Call-Appearances on Station are Busy	B	B
CAS Main in Night Service	n	n
Call Forward to CAS-Main attendant	f	f
Service Observe agent & one other party		o
Held Call Timed Reminder	hc	
Call Redirected do to coverage criteria	cover	c
Terminate call via intercom	ICOM	i
Terminate call via ARS	ARS	A
Terminate call via Call-Forward	forward	f
Terminate call via Call-Park	park	p
Automatic Call Back Activate	ACB	Q
Queued at terminator	QTQ	q
Terminate via Call Waiting	wait	w
Terminate to Attendant via Trunk Group control	control	
Terminate to Attendant via Recall Call	recall	
Terminate to Attendant via Return Call	return	
Terminate to party with Do Not Disturb	DND	D

Considerations

The following features of the 731xH series voice terminals are affected when used with the DEFINITY switch:

- ACD

  The 731x hybrid terminals are not recommended for ACD agent applications because in heavy traffic conditions display information may be lost.

- Call Appearance Button LEDs

  Both the red and green LEDs associated with the first 10 dual-LED feature buttons are supported for the 7313H, 7314H, 7315H, 7316H, and 7317H voice terminals when these buttons are administered as call or bridged appearances:

  — The red ‘‘on’’ LED indicates the call appearance that is selected when the user goes off-hook.

  — The green ‘‘on’’ LED indicates the call at that appearance or at a bridged appearance.

- Feature Buttons

  Call appearances or bridged appearances can be assigned only to feature buttons 1 to 10 on the 731xH voice terminals.

- Feature Button LEDs

  The 731xH voice terminals have two LEDs (one red and one green) associated with each administrable feature button. The DEFINITY switch only supports the green LEDs associated with the feature buttons. Operation of the green LEDs is identical to operation of the 7400 series terminals.

- HFAI

  The DEFINITY switch replaces the HFAI capability on System 25 with the Internal Automatic Answer (IAA) feature. When an internal call arrives at an idle 731xH series terminal that has IAA activated, the speakerphone and microphone at that terminal activate and automatically answer the call. When this occurs, the calling voice terminal receives a distinctive tone burst, while the called voice terminal emits a distinctive tone burst via the speaker.

- Message Button

  The message button is not supported for the 731xH voice terminals. As a result, pressing the Message button on one of these voice terminals has no system or local effect.

- Message LED

  The Message LED is supported for the 731xH voice terminals. This LED is controlled by existing AUDIX and Leave Word Calling features on the DEFINITY switch. With these terminals, users cannot control the state of the Message LED by pressing the Message buttons. The Message LED operation is identical to that provided by the 7400 series voice terminals.

  The Message Retrieval and Retrieval Using Display modes are not supported for ATL terminals. As a result, these modes do not affect Message LEDs on ATL terminals.

- Microphone

  Microphone button operation is supported for the 731xH voice terminals. Each of these voice terminals support the following capabilities:

  — If the terminal is off-hook on the handset, pressing the Microphone button has no effect.

  — If the terminal is off-hook with the speakerphone active, pressing the Microphone button toggles the microphone between ‘‘on’’ and ‘‘off.’’

- Mode Switch

  The DEFINITY switch does not support the mode switch for testing or programming.

- Recall Button

  The Recall button is not supported for the 731xH voice terminals. As a result, the Recall button has no system or local effect when pressed by the user.

- Speakerphone

  Full speakerphone and headset-adjunct operation are supported for the 731xH voice terminals. For each of these voice terminals, the DEFINITY switch supports the following capabilities:

  — Activation/deactivation of the terminal’s microphone via the fixed MIC button or HFAI/MIC button.

  — Activation/deactivation of the speakerphone function via the Speakerphone button.

  — Control of the terminal’s microphone and speaker LEDs in accordance with the states of the microphone and speakerphone.

  If the speakerphone function is activated while the terminal is off-hook on the handset, the handset becomes inactive and the microphone (and speaker) become active.

- Stop and Pause (Drop and Hold) Sub-designated buttons

  The Stop and Pause functions are not supported for the 731xH voice terminals.

Misoperation Handling

Feature Availability

Misoperation Handling is available with all Generic 3 releases. G3V4 and later releases provide for modified misoperation alerting.

Description

Misoperation Handling defines the treatment of calls when a misoperation occurs. A misoperation occurs when calls are left on hold when the controlling station goes on hook.

For example, a misoperation can occur under either of the following conditions:

1. When a voice terminal with a call on hold goes on-hook during a feature operation prior to completion of that operation (in some cases going on-hook completes the operation, as in call transfer). For example, a misoperation occurs when a user places a call on hold, begins to transfer the call, dials an invalid extension number, and then hangs up.
2. When the system enters night service while attendant consoles have calls on hold.

With G3V4 and later releases, the system administrator can alter the standard Misoperation Handling to ensure that an external caller is not left on hold indefinitely, or dropped by the system after a misoperation with no way to reach someone for help.

Misoperation Handling varies considerably depending upon system administration, and the conditions that are in effect when the call is placed on hold. The following sections describe common Misoperation Handling scenarios.

Standard Operation

Standard Misoperation Handling is in effect when “Intercept Treatment on Failed Trunk Transfer” is “Y” and either:

- The “Misoperation Alerting” field on the “Feature-Related System Parameters” form is set to "n."
- The software release is earlier than Generic 3 Version 4.

Standard Misoperation Handling is dependent on the type of voice terminal in use when the misoperation occurs (analog single-line, or digital multi-appearance).

Analog Operation

The following scenario describes a typical misoperation on an analog terminal.

1. While connected to an incoming external call, an analog station user flashes with the intent of transferring the call to another terminal.
2. The user hears dial tone and then dials an invalid extension and hangs up.
3. A misoperation has occurred. The analog station user will receive priority ringback indefinitely

Digital Operation

The following scenario describes a typical misoperation on a digital terminal.

1. While connected to an incoming external call, a digital station user places the call on hold with the intent of transferring the call to another terminal by pressing Transfer.
2. The user hears dial tone and then dials an invalid extension and hangs up.
3. A misoperation has occurred. The held call remains on hold indefinitely with a flashing call appearance lamp.

G3V4 and Later Releases

G3V4 and later releases allow the system administrator to select "Misoperation Alerting" as an option for Misoperation Handling. With this option, calls receive different Misoperation Handling depending upon: the type of voice terminal in use when the misoperation occurs (digital multi-appearance, analog single-line, or attendant console); and the type of call on hold when the misoperation occurs. The following list describes the three call types used to determine Misoperation Handling. The scenarios following the Call Type list describe typical misoperations with "Misoperation Alerting" enabled.

- **Call Type 1:** An outgoing public network call is classified as Type 1 when it is ringing or answered. An incoming call is classified as Type 1 when it is answered.
- **Call Type 2:** An incoming external public network call that has not yet been answered is classified as Type 2. (A misoperation cannot occur with a Type 2 call because an unanswered incoming call cannot be placed on hold without being answered first.)
- **Call Type 3:** All other calls, that is, all internal calls, conference calls, and tie trunk calls of any type are classified as Type 3.

Analog Operation

The following two scenarios describe typical misoperations on an analog terminal.

Scenario 1

1. While connected to an incoming external call (Call Type 1), an analog station user flashes with the intent of transferring the call to another terminal.

2. The user hears dial tone and dials an invalid extension. The user then hears intercept tone.
3. At any point after this, when the user hangs up, the call re-alerts the user for 15 seconds, and is then routed to the attendant.

Scenario 2

1. While connected to an incoming external call (Call Type 1), an analog station user flashes with the intent of placing the current call on hold while calling another extension.
2. The user hears dial tone and dials the CAS Remote Hold/Answer Hold/unhold access code.
3. The user hears dial tone, dials and talks with the second call, and then hangs up.
4. A misoperation has occurred because the first call has been left on hold.
5. The terminal is alerted for 15 seconds with normal ringing and then the call is routed to an attendant.
6. If the first call has not been answered by the expiration of the timer, the call is dropped by the system.

Digital Operation

The following scenario describes a typical misoperation on a digital terminal.

1. While connected to an incoming external call (Call Type 1), a digital station user places the call on hold with the intent of transferring the call to another terminal.
2. The user hears dial tone and then dials an invalid extension and hangs up.
3. A misoperation has occurred.
4. The call on hold re-rings the terminal with a normal ring (not priority ringing) for the number of rings indicated by call coverage administration.
5. After ringing for the administered number of rings, the call is directed to the terminal's administered coverage path, which can direct the caller to an announcement and/or disconnect them.

Attendant Console

A misoperation only occurs on an attendant console when the system enters night service with calls on hold at an attendant console. The following scenario describes a typical misoperation on an attendant console.

1. The system enters night service with calls on hold at an attendant console.
2. All calls on hold on the console start re-alerting (as if the held call timed reminder had expired).

3. When the calls start re-alerting, a timer is started. The timer is set to the duration entered in the “Alerting (sec)” field on the “Attendant Console” form.
4. If the attendant has not answered the call by the time the timer has expired, the call is routed to the system night service destination. (If still not answered the call will be dropped by the "night service disconnect timer.")

Interactions

- Attendant Lockout

  The attendant lockout feature is temporarily disabled on calls that are re-alerting the attendant console following a misoperation. This allows the calls to be answered by the attendant.

Administration

To enable G3V4 and later release misoperation alerting, enter "y" in the "Misoperation Alerting" field on the “Feature-Related System Parameters” form. For complete instructions for administering Misoperation Handling, see "Misoperation Handling" in the *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

See the country applications notes for complete instructions for administering the Misoperation Handling feature to meet France Type Approval requirements.

Hardware and Software Requirements

No additional hardware or software is required.

Modem Pooling

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows switched connections between digital data endpoints (data modules) and analog data endpoints and acoustic coupled modems. The analog data endpoint can be either a trunk or line circuit.

Data transmission between a digital data endpoint and an analog endpoint requires a conversion resource since the DCP format used by the data module is not compatible with the modulated signals of an analog modem. The conversion resource translates the DCP format into modulated signals and vice versa.

The Modem Pooling feature provides pools of conversion resources.

Integrated conversion resources and combined conversion resources are available with the system. The integrated type has functionality integrated on the TN758 Pooled Modem circuit pack, which provides two conversion resources and each one emulates a Trunk Data Module cabled to a 212 Modem. This integrated type is not available for countries that use A law. The combined type is a Trunk Data Module cabled to any Trunk Data Module-compatible modem to provide a conversion resource. Combined type applies to all system independent of system companding.

When a conversion resource is required, the system queries the digital data module associated with the call to determine if its options are compatible with those supported by the modem pools. If the data module options are not compatible, the originating user receives intercept treatment. If the options are compatible, the system obtains a conversion resource from the appropriate pool. If a conversion resource is not available, the user receives reorder treatment. If all data calls, including analog, are not successfully established, the call will be disconnected within 15 seconds (handshake time-out).

In almost all cases, the system can detect the need for a conversion resource. Data calls originated from an analog data endpoint to a digital data endpoint require that the user indicate the need for a conversion resource, since the system considers an analog call origination as a voice call. This need is indicated by dialing the Data Origination Access Code before dialing the digital data endpoint. Use of Data Call Preindication before One-Button Transfer To Data is recommended when establishing data calls that use toll network facilities. Needed conversion resources are reserved before any toll charges are incurred.

The DEFINITY system provides a ''HOLD Time'' parameter to specify the maximum time any conversion resource may be held but not used (while a data call is in queue).

Combined conversion resources additionally supports the following configurations:

- IBM bisynchronous protocols typically used in 3270 and 2780/3780 applications. Both require 2400 bps or 4800 bps, half-duplex, synchronous transmission.
- Interactive IBM-TSO applications using 1200 bps, half-duplex, asynchronous transmissions.
- DATAPHONE II switched network modems supporting asynchronous and synchronous communications, and autobaud at 300, 1200, or 2400 bps.
- DEFINITY system Generic 3 can operate up to 19.2 kbps.
- Different pools can have different data transmission characteristics.

The following modem options are supported by the integrated (only) pool:

- Receiver Responds to Remote Loop
- Loss of Carrier Disconnect
- Send Space Disconnect
- Receive Space Disconnect
- CF-CB Common
- Speed, Duplex, and Synch (administered)

Considerations

Modem Pooling offers a pool of conversion resources which increase data call flexibility. Conversion resources allow analog data endpoints, using modems, to communicate with digital data endpoints (using data modules). Also, pooling of conversion resources allows maximum use of such facilities.

Data Call Preindication is recommended for off-premises data calls involving toll charges.

On data calls between a data module and an analog data endpoint, Return-to-Voice releases the conversion resource and returns it to the pool. The voice terminal user is then connected to the analog data endpoint.

For traffic purposes, DEFINITY system Generic 3 accumulates data on modem pooling calls separate from voice calls. Measurements on the pools are also accumulated.

When a voice terminal user places a data call to digital data endpoint and does not transfer the call to another digital data endpoint, but uses a modem or acoustically coupled modem, the user must dial the Data Origination access code assigned in the system before dialing the distant endpoint.

DEFINITY system Generic 3i can support up to five pools; Generic 3s can support up to two pools, and Generic 3r can support up to 63 pools. The pools may be combined, integrated, or any mix thereof. Each pool has a capacity of up to 32 conversion resources.

Use of Modem Pooling cannot be restricted. Also, queuing for conversion resources is not provided, although calls queued on a hunt group retain reserved conversion resources while queued.

Mixing of modems from different vendors within a combined pool should be avoided since a difference in transmission characteristics may exist. Mixing is possible, but satisfactory results are not guaranteed.

Data transmission characteristics (speed, duplex, and synchronization mode), as administered, must be identical to the Trunk Data Module and modem optioning by the customer.

Each data call that uses Modem Pooling uses four time slots (not just two). As a result, heavy usage of Modem Pooling could affect the Trunk Data Module bus blocking characteristics.

Tandem Switches will not insert a pooled modem. It is the responsibility of the originating switch to do so.

Interactions

The following features interact with the Modem Pooling feature.

- Data Call Setup

  Data calls to or from a Trunk Data Module cannot use Modem Pooling.

- Data-Only Off-Premises Extensions

  Modem Pooling is not possible on calls to or from a Data-Only Off-Premises Extension, when this type of digital data endpoint uses a Trunk Data Module.

- Data Privacy and Data Restriction

  The insertion of a modem pool does not turn off Data Privacy and/or Data Restriction.

- DMI

  Data calls originated from a local analog data endpoint to a DMI trunk must dial the Data Origination Access Code to obtain a conversion resource. Data calls on DMI trunks to local analog data endpoints automatically obtain conversion resources.

- DS1 Tie Trunk Service

  Conversion resources used for Modem Pooling can only be connected to AVD DS1 tie trunks via Data Terminal Dialing or by dialing the feature access code for data origination.

- CDR

  With G3i, Data Call CDR records the use of modem pools on trunk calls.

Administration

Modem Pooling is assigned on a per-pool basis by the System Manager. The following items require administration.

- Conversion Resources — For integrated conversion resources, assign Pooled Modem circuit packs. For combined conversion resources, assign Trunk Data Module and associated modems ports, speed (up to three speeds), and duplex and synchronization characteristics.
- HOLD Time (per pool basis) — Specify the maximum time any conversion resource may be held and not used (while a data call waits in a queue). Default value is five minutes.
- Data Origination Access Code — Allow users to indicate a need for a conversion resource on an analog data call origination.

G3i-Global can only use a modem compatible with the ITU-T 108.1 signaling procedures. Either a synchronous or asynchronous modem may be used.

Hardware and Software Requirements

One TN758 Pooled Modem circuit pack is required for each two integrated conversion resources provided. Combined conversion resource requires one port on the Digital Line circuit pack and one port on an Analog Line circuit pack.

No additional software is required.

Move Agents From CMS

Feature Availability

Move Agents From CMS is available from a CMS adjunct with all G3 releases. In addition, G3V4 and later releases provide the ability to move agents or change skills while the agent is staffed. With R3V2 and later CMS releases, a single agent's skill assignments can be changed while the agent is staffed. With CentreVu™ CMS R3V4 and later releases, up to 32 agents' split or skill assignments can be changed while the agents are staffed.

Description

Allows a user to move up to 32 agents from one split to another from the CMS terminal.

G3V3 and earlier releases restrict the move to agents who are not logged in. In addition, only one Expert Agent Selection (EAS) agent can have skill(s) changed on the CMS terminal at one time and this change does not take effect until they log out.

With G3V4 and later releases, users can change agents' split or skill assignments while the agents are logged in. In addition, with EAS one skill can be added, deleted or moved simultaneously for a group of up to 32 agents.

In either case non-EAS agent assignments can only be changed for splits that are measured by CMS. EAS skills do not need to be measured.

When agents are not logged in, the split/skill changes take effect immediately. Staffed agent changes are either immediate or pending. See "Move Agent While Staffed" later in this section for a description of when changes remain pending.

Even though the CMS screen can be used to move multiple agents, the system makes each move individually. If, for some reason, one of the requested agents cannot be moved, this does not affect the other move agent requests.

The CMS displays whether a move is successful, pending, or fails. It also gives the reason for any failed change.

In a non-EAS environment, when CMS is used to move agents from one split to another, all split-associated buttons assigned to an agent are automatically assigned to the new split. Split-associated buttons are those buttons assigned to an agent's voice terminal or console that are associated with a specific split. For example, the ACW button is a split-associated button. If an agent is assigned an ACW button for one split, and is moved to another split via the CMS screen, the ACW button, instead of being associated with the first split, is then associated with the second split. However, if a user already has a specific split-associated button for both the old split (split from which the agent is removed) and the new split (split to which the agent is added), the button assignments remain unchanged. This keeps duplicate split-associated buttons from being assigned to the same split. The following buttons are split-associated buttons:

- Manual-In
- Auto-In
- Auxiliary Work
- After Call Work
- Assist
- Oldest Queued Time (OQT)
- Number of Queued Calls (NQC)
- ICI

For complete instructions for moving agents, see: the *CentreVu™ Call Management System Release 3 V4 Administration Manual*, 585-215-800; or, the appropriate version Call Management System administration manual (585-215-521 or 585-215-511).

Move Agent While Staffed

With G3V4 and later releases, agent's can be moved between splits and EAS agent skills can be removed, replaced or added while the agents are staffed. In this way, Call Centers can be more responsive to changing requirements including peaks in Call Center activity.

In a non-EAS environment, an agent extension or group of up to 32 agents' extensions can be moved between splits. In an EAS environment, a skill can be added, deleted or moved at one time for a group of up to 32 agents.

The changes are effective immediately unless the agent is:

- Active on a call appearance
- Is in After Call Work (ACW)
- Has Direct Agent Calls (DACs) waiting (EAS only)
- Is the last agent in a non-vector controlled split with calls in queue for the split that is being moved (non-EAS only)

In these cases, the move is pending until the agent logs out or completes all calls and becomes available or enters AUX work. If a request is pending for a non-EAS agent, another request to move that agent will be denied.

When a staffed agent is moved between non-EAS splits, the agent work mode for the new split is the work mode of the removed split. With EAS, when a skill is added or changed, the agent's work mode remains the same as it was before the change.

G3V4 also allows for the administration of a new button used to notify agents of split/skill changes. If a change is made from CMS while the agent is staffed, the button (alrt-agchg) flashes at the agent's terminal. When the agent pushes the button, the flashing stops. The button also flashes when a user with console permissions adds or removes an agent's skill. If an agent changes his or her own skills from the voice terminal, the button lamp does not flash.

Considerations

Because changing an agent's split/skill assignments may affect what call is delivered next, agents need to have prior knowledge that a call of the new type could be delivered at any time. Supervisors may elect to call agents prior to making the move.

In a non-EAS environment, agents need to be aware of what splits are associated with a given set of work mode buttons. This may be difficult for agents in multiple splits, particularly when moves are common. In some applications, it may be necessary to limit agents in a single split or use EAS, which has a single set of work mode buttons.

In applications where agents are often moved between splits/skills, it is recommended that the VuStats button not have a particular split/skill administered to the button. The agent can enter the split/skill on the dial pad or the split can be referenced as the first, second, third, or fourth split logged into on the VuStats format.

Agents who have been moved from CMS in a non-EAS environment may not know which splits they are logged into when they need to log off. For this reason, it is recommended that a VuStats display be created for these agents showing the splits in which they are currently staffed.

Only agents with voice terminal extensions can be moved using the Move Agents From CMS feature. Individual attendants serving as agents cannot be moved using the Move Agents From CMS feature.

Agents cannot be added to or removed from splits via the Move Agents From CMS feature. They can only be moved from one split to another. Additions and removals must be done on the Management Terminal.

System administration (local or remote) that requires use of the ADD, CHANGE, or REMOVE commands cannot be done while the system is making agent moves as a result of a request from the CMS. After the moves are complete, this type of administration can be done.

Move agents, Move Vector Directory Numbers, or Change Vector requests from other CMS terminals will fail if one CMS terminal's Move Agents request is in progress.

Interactions

- Agent Call Handling

  If an agent is moved from one split to another via the Move Agent From CMS feature, the agent's split-associated buttons are reassigned to the new split. The agent should be provided with a set of button labels so the buttons can be updated accordingly.

- G3-MT/G3-MA

  Under normal circumstances when a skill is changed from G3-MT or G3-MA, the change does not take effect until the agent logs in. However, if a staffed agent is moved from CMS, the switch automatically logs the agent out and back in again. In this case, the G3-MT or G3-MA changes would take effect at the same time as the CMS changes.

Administration

None required.

Hardware and Software Requirements

A CMS adjunct and ACD software is required.

Multi-Appearance Preselection and Preference

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides multiappearance voice terminal users with options for placing or answering calls on selected appearances.

- Ringing Appearance Preference

  When a user lifts the handset to answer an incoming call, the system automatically connects the user to the ringing call appearance. If more than one call is incoming, the user is automatically connected to the eldest (first-in) ringing call appearance. The in-use (red) lamp tracks the ringing appearance and the answered appearance.

- Idle Appearance Preference

  When a user lifts the handset to place a call, the system automatically connects the user to an idle appearance even if an incoming call is ringing at another appearance. The in-use (red) lamp tracks an idle appearance when the handset is lifted.

- Preselection

  Before lifting the handset to place or answer a call, the user can manually select an appearance (press a call appearance button or a feature button) where the in-use lamp is dark. Preselection is used, for example, when the user wants to reenter a held call or activate a feature. Preselection also activates the speakerphone if the voice terminal is so equipped.

  The Preselection option overrides both Preference options. If the user does not lift the handset within five seconds after using Preselection, the selected appearance returns to idle.

  Preselection can be used with a feature button. For example, if an Abbreviated Dialing button is pressed, a call appearance is automatically selected and, if the user lifts the handset within five seconds, the call is automatically placed. Preference dictates whether the user is connected to the ringing call appearance or to an idle call appearance. If there is no incoming call, the user is automatically connected to an idle call appearance upon lifting the handset. This is true, regardless of the Preference option assigned.

Considerations

Multi-Appearance Preselection and Preference is used to select the call appearances to which users are connected when they lift the handset.

Multi-appearance voice terminals can have from 2 to 10 call appearances. One of these call appearances is reserved for placing calls or for receiving a Priority Calling call. If a voice terminal has two call appearances and one of them is active, a nonpriority call cannot access the other call appearance, even if the call appearance is idle. Also, the reserved call appearance is not a fixed-position button. It is simply the last idle call appearance. For example, assume a voice terminal has ten call appearances. Any nine can be in use, but the tenth (last) one is reserved. The restriction of the last call appearance is administrable on ‘‘Station’’ form.

This aspect of system operation should be considered when determining the number of call appearances for a voice terminal. The default value and recommended number of call appearances is three.

All incoming and outgoing calls require a call appearance. There are no hidden or free call appearances. For example, consider a member of a Call Pickup group with a Call Pickup button. When a call rings some other group member, it can normally be answered by pressing the Call Pickup button. However, pressing the button selects a call appearance for the call, if available. If a call appearance is not available, the call cannot be picked up. Similarly, calls originated using the Facility Busy Indication feature calls also require a call appearance. In this case, the call cannot be completed unless an idle call appearance is available. A Facility Busy Indication button on a called voice terminal provides a visual indication of the busy or idle status of another facility. It does not provide a talking path. These facts should be considered when determining the number of call appearances for a voice terminal.

Interactions

The following features interact with the Multi-Appearance Preselection and Preference feature.

- Call Coverage

  If Cover All Calls (part of the Call Coverage feature) is the redirection criteria to be used for a voice terminal, Idle Appearance Preference should also be assigned to the voice terminal. This allows the principal (called party) to lift the handset without being accidentally connected to a call which should be screened.

- Automatic Incoming Call Display

  Incoming calls are not displayed if Idle Appearance Preference is activated.

- Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI)

  When an ISDN-BRI station assigned with the “Select Last Used Appearance?” field of the “Station” form set to yes completes a transfer while off-hook using the handset, the user will be left hearing a dial tone on the last-used appearance, rather than the silence heard in the same situation by an user of other station types.

Administration

The Idle Appearance Preference option is administered on a per-terminal basis by the System Manager. If Idle Appearance Preference is not administered, the voice terminal has Ringing Appearance Preference. Both preference options cannot be used on the same voice terminal, and no preference is not an option. Administratively, Idle Appearance Preference (yes or no) is the only choice. No, which is the system default, selects Ringing Appearance Preference. No administration is required for preselection.

Hardware and Software Requirements

No additional hardware or software is required.

Multiple Call Handling

Feature Availability

Multiple Call Handling on request is available with G3V3 and later releases. Forced Multiple Call Handling is available with G3V4 and later releases. It is only available when Automatic Call Distribution (ACD) is optioned. There are also options for Multiple Call Handling on-Request and Multiple Call Handling Forced.

Description

Multiple Call Handling allows an agent to be interrupted with an additional ACD call. When MCH is optioned for a split or skill, the agent can receive an additional call when he or she requests one. Or, an additional call will automatically alert by ringing at the agent's terminal.

This feature is useful in applications in which it is important for an agent to take another call without dropping of the active call. For example:

- In some situations, an agent and a caller may need to wait on a call for information. MCH allows the agent to put the call on hold until the information is available and to handle other ACD calls in the meantime.
- In cases where ACD calls are considered to be more important to the business than non-ACD calls, MCH can be used to interrupt an agent who is on a non-ACD call with a call from an ACD split or skill.
- Similarly, in an EAS environment calls from one skill may be considered more important than calls from another skill. In this case, MCH can be used to interrupt an agent who has a call from the less important skill with a call from the more important skill.

The agent can only receive multiple calls when in the Auto In (AI) or Manual In (MI) mode. All forced MCH calls are delivered with ringing at the agent's station, not with zip tone. Requested MCH calls can be delivered with ringing or zip tone.

MCH is available in either an EAS or non-EAS environment. However, with EAS an agent may be administered with any combination of MCH and non-MCH skills. In a non-EAS environment, agents are limited to being logged in to only one split if that split is a MCH split. Therefore, a non-EAS agent who is logged in to a MCH split is not allowed to log in to another split. Similarly, a non-EAS agent logged in to a non-MCH split is not allowed to log in to an MCH split. If an EAS agent is a member of both MCH and non-MCH skills, the agent is able to process multiple simultaneous ACD or Direct Agent calls only in the MCH skills.

Forced MCH is not recommended with DDC distribution since the first agent would continue to receive calls until all of his or her line appearances were busy.

It is important with forced MCH to limit the number of unrestricted line appearances for an agent to the number of active calls the agent can handle.

Agents and supervisors in on-request splits or skills can use features such as Queue Status, VuStats and BCMS/CMS reports to determine if a call is waiting that must be answered immediately.

The following sections describe each of the MCH options. These options are administered on the “Hunt Group” form.

None

With this option, agents in the split or skill will not receive an ACD call from this split or skill if any call appearance has a call ringing, active or held.

On Request

With this option, agents in the split or skill can request additional calls by placing an active call on hold and then selecting the AI or MI work mode. A queued ACD split/skill or Direct Agent Call is then routed to the agent. The following conditions are true for agents on "on-request" splits or skills.

- If an agent goes into AI or MI work mode, but there are no calls in the queue, the agent is placed at the bottom of the MIA queue, at the bottom of their skill type in the EAD queue, or is made available in the DDC queue.
- Agents must select AI or MI work mode for each new ACD call the agent wants to take while a call is on hold.
- The agent can take additional ACD calls as long as there is an available line appearance.

On Request MCH is useful in situations where an agent may need to wait for information and so can elect to take another ACD call while the first call is placed on hold. It is also useful in conjunction with a feature such as VuStats, where the agent can determine that the queue for a split is too full and elect to take additional calls.

One Forced

With this option, an agent who is idle or is active on a non-ACD call will be automatically interrupted with a single ACD call from this split or skill when:

- The agent is in Manual-in or Auto-in work mode
- The agent is the most idle or next available
- No other ACD call for any of the agent’s splits or skills are alerting, active or held
- An unrestricted line appearance is available

- AUX work is not pending
- Move from CMS is not pending

As long as an ACD call is active or held, the agent will not automatically receive an additional call from the one-forced split or skill. However, an agent who has received the an ACD call can request another ACD call from a one-forced split or skill by placing the active call on hold and then selecting MI or AI work mode.

If an agent with multiple skills is active on an ACD call for a group with "one-forced" MCH, the agent could be forced an ACD call for one of the agent's other skills depending on the MCH type of their other skills.

An agent in a "one-forced" split/skill in the MI/AI work mode is unavailable for that split/skill from the time that any ACD call rings the agent until all ACD calls are abandoned, redirected or dropped by the agent.

Because One Forced MCH forces an ACD call to alert an agent who is not currently on an ACD call, it is useful in situation where ACD calls need to take precedence over other calls.

One Per Skill

One Per Skill MCH is only available with Expert Agent Selection (EAS). With this option, an agent who has no ACD call for this skill will be automatically interrupted with a single ACD call from this skill when:

- The agent is in Manual-in or Auto-in work mode
- The agent is the most idle or next available
- No other ACD calls are alerting
- No other ACD calls for this particular skill are active or held
- An unrestricted line appearance is available
- AUX work is not pending
- Move from CMS is not pending

As long as the one per skill call is active or held, the agent will not automatically receive an additional call from that skill. However, an agent who has received a forced ACD call can request another ACD call from a one-per-skill skill by placing the active call on hold and then selecting MI or AI work mode.

If an agent with multiple skills is active on an ACD call for a "one-per-skill" group, the agent could be forced an ACD call for one of the agent's other skills if that skill has a type of "many-forced" or "one-per-skill."

Because One Per Skill MCH allows a call from a split to be delivered even when the agent is already on an ACD call, it is useful when calls from one skill have higher priority than other ACD calls.

Many Forced

With this option, an agent will be automatically interrupted with an ACD call when:

- The agent is in Manual-in or Auto-in work mode
- The agent is the most idle or next available
- No other ACD calls are alerting
- An unrestricted line appearance is available
- AUX work is not pending
- Move from CMS is not pending

When an agent answers an ACD call, the agent becomes immediately available to receive another ACD call from a "many-forced" split or skill provided the above conditions are met.

Generally speaking, agents in "many-forced" groups in the MI/AI work mode are unavailable only when an ACD call is ringing at the agent's station.

Because Many Forced allows a call to be delivered even when the agent is already active on an ACD call, it is useful when very important or urgent calls must be answered, even when that requires putting an equally important call on hold. It can also be used to force Direct Agent Calls to an agent.

MCH Example

In this example, it is assumed that an agent is logged into four skills each with a different MCH option. The following table shows when a call would be delivered under the described conditions when an unrestricted line appearance is available and the agent is in the Auto-In or Manual-In work mode and when AUX work is not pending. The left column describes the existing conditions when the call arrives. Yes indicates that a call would be delivered for that split under that condition; "no" indicates that a call would not be delivered.

Table 3-71. MCH Call Delivery Example

Condition	Skill 1 (MCH=on-request)	Skill 2 (MCH= forced-one)	Skill 3 (MCH=one-per-skill)	Skill 4 (MCH=many)
no calls on set	yes	yes	yes	yes
one active extn call	no	yes	yes	yes
skill 1 call active	no	yes	yes	yes
skill 2 or 4 call active	no	no	yes	yes
skill 3 call active	no	no	no	yes
extn call held, no other action	no	yes	yes	yes
skill 1, 2, or 4 call held, no other action	no	no	yes	yes
skill 3 call held, no other action	no	no	no	yes
extn call held, then AI/MI selected	yes	yes	yes	yes
skill 1,2,3, or 4 call held, then AI/MI selected	yes	yes	yes	yes

Considerations

Work Modes

Activation of the AI or MI work mode with calls on hold will only succeed if the agent has an MCH split or skill and an idle unrestricted call appearance available. If this is not the case, the activation attempt is denied. If the activation attempt was via a work mode button, the button lamp flashes denial. If the activation attempt was via a feature access code, the agent receives intercept tone.

If an agent is in the AI work mode and activates MI work mode with calls ringing or active, the agent simply toggles to the MI work mode. Similarly, an agent in the MI work mode activating the AI work mode with calls ringing or active simply toggles to the AI work mode.

If an agent selects ACW with calls on hold, ACW is pending until all calls complete or until an MI call completes. If the agent selects AUX work with calls on hold, AUX work will be pending until all calls complete. Additional ACD calls will not be delivered when AUX work is pending. When a MI/AI ACD or Direct Agent call with pending ACW completes, the agent enters ACW. When an agent is active on a non-ACD call with ACW pending the agent can receive forced MCH calls.

If the agent is active on a MI ACD or Direct Agent call and some other MI ACD or Direct Agent Call abandons from hold, the agent remains in the MI work mode on the active call.

If the agent is active on an AI ACD or Direct Agent call or is active on a non-ACD call and in the AI or MI work mode and a MI ACD or Direct Agent call abandons from hold, the agent becomes pending for the ACW work mode and the after-call button lamp flashes pending.

If the agent reconnects to a held ACD or Direct Agent call, the agent work mode changes to the work mode associated with the held ACD or Direct Agent call (either Auto-In or Manual-fn). For example, if the agent is in the ACW mode and reconnects to a held MI work mode ACD call, the agent returns to the MI work mode. Similarly, if the agent is in the AI work mode and reconnects to a non-ACD call that was made or received while in the ACW mode, the agent returns to the ACW mode.

Interactions

The following interactions are true for all MCH options.

- Automatic Hold

  An agent in "many-forced," "one-forced," or "force-per-grp" split/skill may answer a ringing ACD call by pushing the line appearance. If the automatic hold feature is optioned, the active call will automatically be placed on hold.

- Call Work Codes (CWC) and Stroke Counts

  An agent processing multiple ACD calls simultaneously in a MCH environment can enter CWCs and Stroke Counts. When an agent enters a CWC or Stroke Count with multiple ACD calls on the station, the CWC or Stroke Count will be associated with the last ACD call the agent was processing. If a CWC or Stroke Count is entered during an active ACD call, possibly with calls on hold, the CWC or Stroke Count will be associated with the active ACD call. MCH splits/skills can have Forced Entry of Call Work Codes and Stroke Counts.

  If an "on-request" agent is active on a call that has forced entry of CWC or Stroke Counts and places that call on hold without entering a CWC or Stroke Count, the agent cannot request another call. The agent must first enter a CWC or Stroke Count for the held call.

  If a "many-forced" agent is in MI/AI work mode in a split/skill with forced entry of Stroke Counts and Call Work Codes, the agent will be forced an ACD call even if a Stroke Count or Call Work Code has not been entered.

  If an agent is active on a MI call and another MI call abandons with forced entry of Stroke Counts and Call Work Codes, the forced entry is ignored for the abandoned call.

- Direct Agent Calling

  Since ACD agents can service multiple calls from the MCH skills that the agent has logged in to, the agent can handle multiple Direct Agent calls from the MCH skills. When a Direct Agent call is queued in a split/skill, the queue status indicator for the split/skill will not be lit when a Direct Agent call is queued. ACD agents will be notified that Direct Agent calls are waiting by a ring ping and the current work mode lamp will flash.

- Move Agent While Staffed

  An agent with a move pending may hold a call and request another ACD call. All calls and ACW must be complete before the pending move will take place.

- Non-ACD Calls

  If an agent has activated AI or MI work mode with calls on hold, he or she can then answer or originate a non-ACD call. With "on-request" MCH, the agent is then temporarily unavailable for an ACD or Direct Agent call. With forced MCH if other conditions are met, a call may be delivered. If an agent in ACW reconnects to an AUXIN/AUXOUT call, the agent remains in ACW.

- Queueing

  When an agent becomes available, the system places the agent at the end of the queue for Uniform Call Distribution (UCD) hunt groups, at the bottom of the skill type for Expert Agent Distribution (EAD) hunt groups, or makes the agent available for Direct Department Calling (DDC) hunt groups. When this agent becomes the most available according to the group type (UCD, EAD, or DDC) a queued ACD or Direct Agent call is routed to the agent and rings at the agent’s station.

  If the last agent on a forced MCH split is pending for AUX work mode in a non-vector controlled split, the agent will be forced to drain the queue before the AUX work mode takes effect. This agent will continue to receive ACD calls based on the MCH type of their split and the AUX work mode will remain pending until the queue is drained.

- Redirection on No Answer

  If an MCH agent has a call active or on hold and the Redirection on No Answer time expires for another ringing ACD call, the call is redirected back to the split/skill. However, the agent is not taken out of service.

- Restricted Line Appearance

  If the last available line appearance is administered as "Restricted Last Appearance" on the “Agent’s Station” form, the agent does not receive additional ACD calls because the appearance is reserved for making conference or transfer calls.

- Voice Terminal Displays

The call prompting digits shown are those associated with the active ACD call. If the agent reconnects to a call on hold, the display will show the digits for the reconnected call.

Administration

Multiple Call Handling is enabled on the “System-Parameters Customer Options” form. It is administered on the “Hunt Group” form. See “Multiple Call Handling” in the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

Hardware and Software Requirements

With the exception of ACD no additional software or hardware is required. MCH is only available on multi-appearance terminals. If a CMS adjunct is used, On Request MCH requires R3V2 or a later release; Forced MCH requires CentreVu™ CMS (R3V4) or a later release.

Multiple Listed Directory Numbers

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows a publicly published number for each incoming and two-way (incoming side) FX and local CO trunk group assigned to the system. Also allows DID numbers to be treated as LDNs.

When a CO or FX LDN is called, a trunk group is accessed. The trunk group then routes the call to the incoming destination designated for that trunk group. The incoming destination for an FX or CO trunk group can be one of the following:

- Attendant group
- ACD split
- DDC group
- UCD group
- Remote Access

All DID LDN calls route directly to the attendant group.

Considerations

Multiple LDNs provide publicly published numbers for a business. These numbers allow public access to an attendant. LDNs are also useful when it is necessary that the public be able to contact a particular DDC or UCD group. The feature can also be used for Remote Access.

A unique display for incoming call identification can be provided for each LDN, including the DID numbers.

Interactions

If Night Service has been activated and a night console is not assigned or is not operational, incoming LDN calls route as follows:

- DID LDN calls route to a designated DID LDN night extension. If no DID LDN night extension is designated, DID LDN calls route to the attendant.
- Other incoming calls on trunk groups route to the night destination specified for the trunk group. If the night destination is the attendant, calls route to the DID LDN night extension, if specified. If no DID LDN night extension is specified, calls route to the attendant. If no night destination is

specified for the trunk group, the calls route to the normal incoming destination for that trunk group. If that destination is an attendant, calls route to the DID LDN night extension.

- Internal calls and coverage calls to the attendant route to the DID LDN night extension.

Administration

Multiple LDNs is administered by the System Manager. The following items require administration:

- Incoming destination for each CO trunk group and each FX trunk group used for LDNs
- DID LDNs
- DID LDN night extension
- A unique name for each LDN (optional, for display purposes)

Hardware and Software Requirements

No additional hardware or software is required.

Music-on-Hold Access

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides music to a party that is on hold, waiting in a queue, parked, or on a trunk call that is being transferred. The music lets the waiting party know that the connection is still in effect.

The system provides automatic access to the music source.

Considerations

The music provided by Music-on-Hold Access lets the waiting party know that he or she is still connected. Waiting parties are less likely to hang up. This results in a greater number of completed calls.

If a multiple-party connection is on hold, waiting in queue, or parked, music is not provided.

The number of calls that can be connected to Music-on-Hold Access simultaneously is not limited.

The treatment of transferred trunk calls can be controlled by the System Parameter field ‘‘Music (or Silence) on Transferred Trunk Calls.’’ Such calls can be administered to receive either music or silence.

If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from or pay fees to a third party such as the American Society of Composers, Artists, and Producers (ASCAP) or Broadcast Music Incorporated (BMI). You can purchase a Magic on Hold® system, which does not require such a license, from AT&T.

Interactions

When any one of the following features is activated, music is provided when one party is waiting or held:

- Hold
- Conference — Terminal
- Transfer (application of music or silence as opposed to ringback tone can be controlled for trunk calls)

- Call Park
- A call placed in queue for a DDC group, UCD group, or ACD split, can receive a delayed announcement followed by music.

If a call with either Data Privacy or Data Restriction activated is placed on hold, Music-on-Hold access is withheld to prevent the transmission of some musical tone which a connected data service might falsely interpret as a data transmission.

When Tenant Partitioning is in use, each tenant can be assigned a unique source for music to be heard when a caller is placed on hold. See the "Tenant Partitioning" feature for additional information.

Administration

Music-on-Hold Access is administered on a per-system basis by the System Manager. The only administration required is the assignment of the port number used to provide the feature, and the assignment of whether music, tone, or silence is heard on transferred trunk calls.

Hardware and Software Requirements

Requires the music source and one port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law), or one port on a supported Analog Line circuit pack such as the TN742. A KS-23395L4 coupler is required to connect the music source to the Analog Line port.

Also, if the music source is not FCC registered, a 36A or 909A voice coupler is required to provide an interface and system protection for the music source. See *DEFINITY Communications System Generic 3 Wiring*, 555-230-111, for a description of Music-on-Hold hardware requirements.

Multiple Music-on-Hold sources require Tenant Partitioning Software.

Names Registration

Feature Availability

This feature is available with all Generic 3 releases.

Description

Automatically sends a guest's name and room extension from the PMS to the switch at Check-In, and automatically removes this information at Check-Out. In addition, the guest's call coverage path (for example, a coverage path that terminates at a voice mail adjunct or a hotel operator) will also be sent from the PMS to the Switch during Check-In, and set to the ''Default Coverage Path for Client Rooms'' at Check-Out.

The information provided by the Names Registration feature may be displayed on any attendant console or display-equipped voice terminal located at various hotel personnel locations (for example, Room Service, Security, and so on). This allows personnel at these locations to provide personalized greetings to calling guests. For example, if John Smith called room service, the restaurant personnel with a display-equipped voice terminal, would see John's name and room extension and could answer with a personalized greeting.

Since the updates are sent automatically from the PMS to the switch, the System Manager does not have to manually add guest names into the switch via the Management Terminal. Normally, in a hotel environment where the daily turnover of guests is large, manual administration of the updates using a Management Terminal would be a full time administrative task, and would be duplicating the information already resident in the PMS. By linking the automatic updates to the check-in and check-out sequences, the hotel can provide personalized displays more efficiently.

Check-In and Check-Out

During the check-in procedure, information about the guest is obtained and stored in the hotel's PMS. At this time, the PMS sends a check-in message to the switch. When the check-in message is sent, the switch removes the outward restriction on the telephone in the guest room, changes the status of the room to occupied, clears any previous wake up calls and message waiting lamp indications, and deactivates Do Not Disturb. Guest Name Registration during checkin would add two more operations to those already being performed. These operations would be to update the PBX names internal table and the call coverage path for the guest station. Names Registration enhances the above list of operations by automatically sending a guest's name, extension (room) number, and preferred call coverage path upon check-in. Also, at check-out, Names Registration automatically changes the call coverage path to the administered ''Default Coverage Path for Client Rooms.''

The Check-In and Check-Out functions are discussed in the PMS description elsewhere in this document.

Guest Information Input/Change

Guest Information Input/Change, allows guest information (name or coverage path) to be entered or altered subsequent to the check-in message. Hotel personnel can change this information at the PMS and it is automatically sent to the switch.

The Guest Information Input/Change function is used in those situations when the guest's name associated with an extension must be changed, input of a guest's name must be made after the checkin sequence has taken place, or a change in call coverage arrangement must be made. For example, a hotel may check in airline personnel prior to their arrival at the hotel in order to guarantee their reservation. However, hotel personnel may be unaware of the guests' names until their actual arrival. The names of the airline personnel could be updated using the Guest Information Input/Change function upon actual arrival.

Names Registration Information Format

For both Names Registration and Guest Information Input/Change formats, the guest's name may consist of as many as 15 characters.

The format used by the PMS (last name first, plus first initial and title, and so on) will be sent to the switch and is displayed as it is stored within the switch. All spaces and commas within the name display must also be encoded within the 15 characters. In addition to the 15 character guest name, an extension number (normally up to five digits, but may be up to six digits with prefixed extensions), which corresponds to the guest's room number, will appear on display-equipped voice terminals at hotel service desks.

The guest's name may be in all upper-case letters, all lower-case letters or a mixture of upper- and lower-case. If a hotel would like to be able to use the Integrated Directory feature (described elsewhere in this document), the guest's name must be entered using one of the following methods:

- Last Name, comma, First Name (for example, Jones,Fred)
- Last Name, comma, First Name, space, Title/Middle Initial/Name (for example, Jones,Fred Mr)
- Last Name Only (for example, Jones)
- First Name, space, Middle Name, space, Last Name (Will appear as Jones,Fred A)

Only alphanumeric characters, commas and spaces may be used in the "Name" field when Integrated Directory is desired. When the feature is not in use, the guest name may be sent to the switch using the above methods and may use periods. However, the periods will not be displayed.

Call Coverage

Both Names Registration and Guest Information Input/Change messages contain call coverage path numbers. These numbers are not displayed but are used to configure the appropriate call coverage arrangements for guest phones. Path arrangements for voice mail, text messages, any available coverage point, or no coverage at all is sent by the PMS for automatic call coverage reconfiguration.

The Call Coverage paths are established at the switch and are then used by the PMS to alter the call coverage arrangement for a guest. If a customized arrangement is desired, the PMS must send a coverage path number (one through 600), and manual administration of the specific path can be performed through the Management Terminal.

Considerations

The guest information provided by the Names Registration feature allows hotel personnel to provide personalized greetings to calling guests. Since guest information updates are sent automatically from the PMS to the switch, the System Manager does not have to manually add guest names into the switch via the Management Terminal. By linking the automatic updates to the check-in and check-out sequences, the hotel can provide personalized displays more efficiently.

A maximum of 15 characters can be entered as a guest's name on the PMS.

The call coverage path numbers sent by the PMS to the switch for automatic reconfiguration should be limited to those administered in the switch and stored in the PMS.

The guest's room extension number can have a maximum of five digits.

The PMS controls the format of the name displayed on display-equipped voice terminals.

Interactions

The following features interact with the Names Registration feature.

- Call Coverage

  Establishing call coverage arrangements is not limited to the automatic update during checkin messages sent from the PMS. Hotel personnel require alternate coverage points other than those designated for guests. The switch can still be used to manually administer call coverage paths through the Management Terminal, while automatic updates can still be sent from the PMS for guests' extensions.

- COS

  If an extension has Client Room COS, the save translation operation clears the station name and sets the coverage path to the Default Coverage Path for Client Room when stored on tape. The existing information in memory is not affected. However, if the translations are read in, existing extensions will be affected until a database swap synchronizes the switch and PMS.

- Interface

  During a Room Change/Room Swap, the name originally associated with the first station number is changed/swapped to the second room station along with call coverage path, automatic wake-up entries, message waiting status and controlled restrictions.

Administration

PMS administration as described in the "Property Management System (PMS) Interface" description, elsewhere in this chapter, is required. In addition to this, the items in the following paragraphs should be taken into consideration.

To maintain necessary guest security, hotels do not divulge guests' room numbers to other guests or callers. For this reason, display-equipped voice terminals should not be assigned to guests' rooms. A guest with this capability would be able to dial another extension and view the guest's name at that extension.

Call Coverage paths must be administered on the switch, and the associated path numbers must be used by the PMS to establish coverage arrangements. If only one coverage arrangement is used by a hotel, this number must be used. For suite rooms, prearranged paths can be administered on the switch and the numbers stored within the PMS that would allow one room in the suite to be the coverage point for the other. Special customized arrangements at time of check in (coverage from one guest room to another) are performed by sending the coverage path number from the PMS then manually administering the attributes of the path at the switch.

Both the PMS and the switch are able to alter guests' names stored in the switch The last change that is made (by either system) is the change that is used.

The communication protocol used between the switch and the PMS must be administered as ''transparent.''

The Default Coverage Path for Client Rooms must be administered.

Hardware and Software Requirements

A PMS, if used, can be connected through an MPDM and port on a Digital Line circuit pack or through an ADU and a port on a Data Line circuit pack. A journal printer can be used and also requires an MPDM and a port on a Digital Line circuit pack or an ADU and a port on a Data Line circuit pack.

Optional Hospitality Services software is required to provide the PMS Interface feature.

Network Access — Private

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows calls to be connected to the following types of networks:

- Common Control Switching Arrangement (CCSA)
- Distributed Communications Systems (DCS)
- Electronic Tandem Network (ETN)
- Enhanced Distributed Communications Systems (EDCS)
- Enhanced Private Switched Communications Service (EPSCS) (G3i-Global, G3V2, and later releases)
- Tandem Tie Trunk Network (TTTN)
- TGU/TGE/TGI

A private network provides call routing over facilities dedicated to the customer.

Considerations

With Private Network Access, calls can be made to other switching systems without having to use the public network.

Unless prohibited by the COR, all incoming Private Network trunks except CCSA can access outgoing trunks without attendant or terminal user assistance. All incoming CCSA calls must route to an attendant or a terminal user.

When off-network calling is specified as part of the CCSA and EPSCS service, long-distance calls route as far as possible over these networks before terminating on the public network. Thus, charges for toll calls are reduced. The COR administered to individual system users determines whether access to this capability is allowed or denied.

In Italy, the Traslatore Giunzione Uscente/Entrante/Interno trunks are supported to provide Private Network Access between two systems and also provide some feature transparency for COR (Inward Restriction), DID (when reaching busy stations), and Intrusion features.

Interactions

- Attendant Call Waiting

  G3i-Global, G3V2, and later releases provide Call Waiting via Italian TGU/TGE (main/satellite) trunks. Call Waiting is also provided in Italy and all other countries through DCS.

- Attendant Intrusion

  G3i-Global, G3V2, and later releases provide Attendant Intrusion on Satellite PBX stations via TGU/TGE trunks. Attendant Intrusion is also provided through DCS.

Administration

Private Network Access is administered by the System Manager. The following items require administration:

- Tie trunk groups used with private networks.
- Whether or not access to CCSA and/or EPSCS off-network calling is provided. (This assignment is made on a per-COR basis.)

Hardware and Software Requirements

Requires one port on an analog or DS1 Tie Trunk circuit pack for each trunk assigned. No additional software is required.

Network Access — Public

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides voice terminal users and attendants with access to and from the public network.

Outgoing access is provided to the following:

- COs
- FX offices (distant COs)
- WATS offices (COs receiving toll-free calls)

Incoming access is provided from the following:

- Local COs
- FX offices
- 800 Service office (COs sending toll-free calls)

Considerations

The ARS feature can be used to select the most-preferred route, where possible, for outgoing calls to the public network. Alternatively, trunk access codes can be dialed for manual route selection. Long-distance carrier access codes can be dialed to select particular carriers.

Some central offices do not provide disconnect supervision and this information must be provided on the trunk group administration screen.

Interactions

None.

Administration

Public Network Access is administered by the System Manager. All trunk groups used for Public Network Access must be administered.

Hardware and Software Requirements

Requires one port on a TN747B CO Trunk circuit pack or TN767 DS1 circuit pack (TN464B/C/D support A-law) for each trunk assigned. No additional software is required.

Night Service — Hunt Group

Feature Availability

This feature is available with all Generic 3 releases.

Description

Hunt Group Night Service allows an attendant or a split supervisor to individually assign a hunt group or split to the night service mode. All calls terminating on the hunt group or split in the night service mode are redirected to the hunt group or split's designated Night Service Extension (NSE).

Considerations

The Hunt Group Night Service feature gives added flexibility to attendants and designated voice terminal users who are responsible for activating or deactivating individual hunt groups/splits at various times.

The system can have both Hunt Group Night Service and Trunk Group Night Service features at the same time. An incoming trunk call is redirected to the trunk group's designated NSE. If this NSE happens to be a hunt group or split that happens to be in the Hunt Group Night Service mode, the call is redirected to the hunt group or split's designated NSE.

Calls in progress, such as talking, on hold, or waiting in queue, on the hunt group or split are not affected when the hunt group or split is put in the Hunt Group Night Service mode.

Once the hunt group is in the Hunt Group Night Service mode, all calls are prevented from entering into the hunt group or split queue.

All new calls terminating on the hunt group or split in the Hunt Group Night Service mode are redirected to its designated NSE.

When the hunt group queue becomes empty, all idle members are be put in a busy condition.

If Night Service is activated for a hunt group or split, and a power failure occurs, the hunt group or split automatically returns to the Night Service mode.

Interactions

The following features interact with the Night Service — Hunt Group feature.

- ACD

  When Hunt Group Night Service is activated for a split and the night-service destination is a hunt group, the caller hears the first forced announcement for the original split, if administered, before redirecting. The call is then redirected to the night service destination hunt group. When an agent in the night service hunt group becomes available, the call goes to that agent. If all agents in the destination hunt group are busy, the caller hears the following, if assigned: forced or delayed first announcement, ringback, music-on-hold or silence, and a second announcement.

- Call Coverage

  While Night Service is activated, the NSE's normal coverage criteria and path applies to night service attempting to terminate at that NSE. If the destination of a hunt group NSE's coverage path is AUDIX, AUDIX answers with the mail of the original hunt group. If the NSE is a hunt group or split of any type, the hunt group or split's call coverage criteria and coverage path apply. The hunt group or split's coverage criteria and path can be different from that assigned to the voice terminals that are members of that hunt group or split.

  If a coverage point is a hunt group or split in night service, it is considered unavailable and the call is not forwarded to the coverage point's NSE.

- Call Forwarding — All Calls

  If the hunt group or split is in the Hunt Group Night Service mode and the hunt group or split's NSE has Call Forwarding — All Calls activated, the night service calls terminating to that NSE are forwarded to its designated extension.

  If the forwarded-to destination is a hunt group or split in the Night Service mode, the call is not forwarded and is terminated at the forwarding extension.

- Night Service

  A hunt group or split in the night service mode is unavailable as a night service destination for another group. Night service destinations cannot be chained.

Administration

Hunt Group Night Service is administered on a voice terminal basis or attendant console. The following items require administration:

- Assign ‘‘hunt-ns’’ button(s) to designated voice terminal(s). Up to three hunt group buttons can be assigned to a combination of attendant consoles and voice terminals in each hunt group. The hunt group number must be assigned for each button. These buttons should be assigned to feature buttons that have an associated status lamp. The lamp lights when Hunt Group Night Service is activated. If the assigned button has no status lamp, no visual indication of the Hunt Group Night Service status is given.
- Assign ‘‘hunt-ns’’ button(s) to attendant console(s). Up to three buttons can be assigned to a combination of voice terminals and attendant consoles assigned for each hunt group. The hunt group number must be assigned for each button.
- Assign night service extension to each hunt group that uses the hunt night service.

Hardware and Software Requirements

No additional hardware or software is required.

Night Service — Night Console Service

Feature Availability

This feature is available with all Generic 3 releases.

Description

Directs all calls for the primary and daytime attendant consoles to a night console.

Night Service — Night Console Service is typically activated when an attendant presses the Night button on the principal attendant console, and deactivated by pressing the Night button again. Night service may also be activated and deactivated from one station in the system by use of a night service button assigned to that station.

When Night Service is activated, the night service button for each attendant is lit and all attendant-seeking calls (and calls waiting) in queue are directed to the night console. (If a station has been administered to activate and/or deactivate night service for the entire system, that station's night service button is also lit.)

Considerations

Night Service — Night Console Service calls to the attendant group are still handled by an attendant, even though the primary and daytime attendant consoles are out of service.

Only one night console is allowed in the system. The night console can be activated only when the primary and daytime consoles have been deactivated. The attendant activates the night console and deactivates all other consoles by pressing the Night button on the primary console.

The night console must be identical to, and have the same features as, the primary console. A daytime console can double as the night console.

If Night Service is activated and a power failure occurs, the system, when brought back up, automatically returns to the Night Service mode.

Interactions

Activation of Night Service for the attendant consoles also puts trunk groups into night service, except those trunk groups for which a night service button is administered. See the "Night Service — Trunk Group" feature for details.

Administration

Night Service — Night Console Service is administered by the System Manager. The only administration required is the assignment of a night console and whether or not only DID LDN calls goes to the DID-LDN night service extension.

Each attendant console is required to have a night service button. One additional night service button may be assigned to another station. This allows the additional station to activate night service on the system.

Also, the Night Service Disconnect Timer should be administered. This timer determines when to drop an unanswered incoming trunk call that does not have disconnect supervision.

Hardware and Software Requirements

Requires an attendant console. No additional software is required.

Night Service — Night Station Service

Feature Availability

This feature is available with all Generic 3 releases.

Description

Redirects incoming attendant-seeking trunk calls to designated extension numbers whenever the system is placed in Night Service.

This feature is activated under the following two conditions:

- The attendant (or voice terminal user, if the switch has no attendant) has pressed the Night button on the principal console.
- A night console is not assigned or not activated.

When the above conditions have been met, incoming calls to the attendant route as follows:

- DID-LDN calls route to a designated DID-LDN night extension.
- Internal calls to the attendant route to the DID-LDN night extension (unless the system is administered so only DID-LDN calls can route to the LDN night extension).
- Incoming calls on trunk groups (other than DID trunk groups) which have the attendant as their destination route to the night destination specified for the trunk group or individual trunk. If no night destination is specified, the calls route to the DID LDN night extension.

When Night Station Service is activated, all trunk and internal calls to the attendant (other than calls redirected via Call Coverage or Call Forwarding All Calls) route to either the DID-LDN night extension, the trunk group's specified night destination, or the individual trunk's specified night destination as discussed above. A different extension number can be assigned as the night destination for each incoming central office, foreign exchange, or 800 Service trunk group. Both the DID-LDN night extension and the extension number assigned as a trunk group's night destination can be a voice terminal or an answering group, that is, DDC group, UCD group, or TEG.

Calls redirected to the attendant via Call Coverage or Call Forwarding All Calls do not route to the DID-LDN night extension. These calls enter the attendant queue, and can be answered via the Trunk Answer From Any Station feature, if administered.

Considerations

Night Station Service provides for the answering of attendant-seeking calls when all attendant consoles are out of service due to Night Service activation.

When the Night Station Service feature is active but night station extension numbers have not been established, the Trunk Answer From Any Station feature can be activated.

A ‘‘Night-Serv’’ button can be assigned to either an attendant or a voice terminal extension. This button, when pressed, puts the entire system in night service and incoming calls on all trunk groups (except DID-LDN) route to the night destination specified for the trunk group. Pressing the ‘‘night-serv’’ button on an attendant other than the principal attendant and has no result.

An individual trunk group or hunt group can be put into night service by either an attendant or a voice terminal extension with the required button (Trunk Night Service or Hunt Night Service). When the button is pressed, all calls to that particular trunk group or hunt group are routed to the night service extension assigned to that group. A second depression of the same button deactivates night service for that trunk group or hunt group.

If Night Service is activated and the DID LDN night extension is busy, an incoming DID LDN call receives busy tone or may be forwarded to another number.

If a trunk without disconnect supervision goes to night service, it is dropped after a certain period of time to avoid locking up the trunk (G3i-Global, G3V2, and later releases only). With G3V1, if Night Service is activated, and a call then returns to the console (for example, an incoming call transferred by the attendant that has not been answered and has timed out), the call is dropped. The call is not routed to the DID-LDN night extension.

Interactions

The following features interact with the Night Service — Night Station Service feature.

- Call Coverage

  A call routed to the DID-LDN night extension via Night Station Service does not go to coverage, even if the coverage criteria of the DID-LDN night extension is met.

  Calls redirected to the attendant via Call Coverage do not route to the DID-LDN extension.

If a night extension has a coverage path in which Cover All Calls has been administered, all attendant-seeking calls redirect to coverage and changes to the protocol for handling DID-LDN calls (that is, forwarding attendant-seeking calls on or off premise from the night extension) does not work.

- Call Forwarding All Calls

  Calls redirected to the attendant via Call Forwarding All Calls do not route to the DID LDN extension.

- Inward Restriction

  Inward-restricted voice terminals can be administered for Night Station Service. Night Service features override Inward Restriction.

- Night Service — Trunk Answer From Any Station

  Night Service — Trunk Answer From Any Station and Night Service — Night Station Service can both be assigned within the same system, but cannot be assigned to the same trunk group.

- Remote Access

  The Remote Access extension number can be specified as the Night Station extension number on an incoming, non-DID, trunk group.

- Timed Reminder

  Timed Reminder Calls returning to a console which has been placed in Night Service and has an assigned DID-LDN night extension are not redirected to the DID-LDN night extension, but are dropped.

Administration

Night Station Service is assigned by the System Manager. The following items require administration:

- DID-LDN night extension and permission to let calls other than DID-LDN calls redirect to the DID-LDN night extension.
- Trunk group night destination (per trunk group)
- Hunt group night destination (per hunt group)
- Night-Serv button
- Night Service Disconnect Timer

  (This timer determines when to drop an unanswered incoming trunk call that does not have disconnect supervision.)

- Hunt Night Service button
- Trunk Night Service button

Hardware and Software Requirements

No additional hardware or software is required.

Night Service — Trunk Answer From Any Station

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows voice terminal users to answer all incoming attendant-seeking calls when the attendant(s) is not on duty and when other voice terminals have not been designated to answer the calls.

The incoming call activates a gong, bell, or chime. A voice terminal user dials an access code and answers the call.

Trunk Answer From Any Station (TAAS) is activated only under the following three conditions:

- The attendant or voice terminal user, if switch has not attendant has pressed the Night button on the primary console.
- A night console is not assigned or not operational.
- The Night Station Service feature is not active.

Considerations

When Trunk Answer From Any Station is activated, any user can answer the attendant-seeking trunk call. Even though an attendant is not available, the call is still answered. This reduces the number of lost calls.

If Night Service is activated and a power failure occurs, the system, when brought back up, automatically returns to the Night Service mode.

Interactions

Inward-restricted voice terminals can activate TAAS for incoming trunk calls. Night Service features override Inward Restriction.

Calls which are redirected to the attendant via the Call Coverage and Call Forwarding All Calls features while the Night Station Service feature is activated can be answered via TAAS.

Night Service — Trunk Answer From Any Station and Night Service — Night Station Service can both be assigned within the same system, but cannot be assigned to the same trunk group. Activation at System Night Service also activates Night Service - Trunk Group, for any trunk group without an individual trunk group night service button.

Administration

TAAS is administered on a per-system basis by the System Manager. The following items require administration:

- Dial access code for TAAS (to answer a call)
- Port for the ringing device
- Enable or disable the Night Service Station from taking internal attendant greeting calls if Night Station Service is also in effect.

Hardware and Software Requirements

Requires a ringing device and one port on a TN742, TN746B (A-law), or TN769 Analog Line circuit pack. No additional software is required.

Night Service — Trunk Group

Feature Availability

This feature is available with all Generic 3 releases.

Description

The Trunk Group Night Service feature allows an attendant or a designated voice terminal user to individually assign a trunk group or all trunk groups to the night service mode. Specific trunk groups (individually) assigned to Trunk Group Night Service are in the Individual Trunk Night Service Mode. In this mode, incoming calls made on a specific trunk group are redirected to its designated Night Service Extension (NSE). Incoming calls on the trunk groups not assigned to Trunk Group Night Service are be processed normally. The specific trunk groups can be assigned to Trunk Group Night Service by pressing the individual Trunk Night Service button(s) on the attendant console or a voice terminal.

All trunk groups can be assigned to the night service mode at the same time. In this arrangement, the trunk groups are in the System Night Service mode. Any incoming calls made on the trunk groups are redirected to their designated NSE. All trunk groups can be assigned to System Night Service by pressing the System Night Service button on the principal attendant console or a designated voice terminal.

Considerations

The Trunk Group Night Service feature gives added flexibility to attendants and designated voice terminal users who are responsible for activating or deactivating all, or individual, trunk groups at various times.

All incoming calls on individual or system Night Service trunk groups goes to the trunk group’s NSE unless the trunk group member has its own Trunk Group Member Night Destination, in which case the calls are redirected to that night destination instead of the trunk group’s NSE.

Calls already in progress on a trunk group, such as talking, on hold, or waiting in queue on a trunk group, are not affected when the individual Trunk Group Night Service or System Night Service feature is activated by the attendant or a voice terminal user.

Trunk Group Night Service and System Night Service both work independently of each other. Activation or deactivation of one of these night service features does not affect the other. Specific situations are described below:

- When System Night Service is deactivated, trunks with individual Trunk Group Night Service still activated remain in night service.

- When System Night Service is activated, trunks controlled by individual Trunk Group Night Service buttons remain in day service.
- Trunks with individual Trunk Group Night Service can be taken out of Night Service even though the rest of the system remains in Night Service.
- Trunks with individual Trunk Group Night Service can be put into Night Service even though the rest of the system remains in day service.
- Trunk groups assigned to individual Trunk Group Night Service are not reassigned to System Night Service when the System Night Service feature is activated. Those trunk groups that are not currently assigned to Trunk Group Night Service are assigned to System Night Service.

If a trunk is added to a trunk group while that trunk group is in Trunk Group Night Service, the trunk is brought up in night service.

Individual Trunk Group Night Service does not apply to DID trunk groups.

If Night Service is activated for a trunk group, and a power failure occurs, the trunk group automatically returns to the Night Service mode.

If for some reason, a voice terminal with a trunk-ns button remains out-of-service after a system reboot and later comes back in service, the trunk-ns lamp shows the trunk status within 10 seconds of coming back in service. For example, a voice terminal with a trunk-ns button may be unplugged when the system is rebooted. If the voice terminal is later plugged back in, the trunk status is shown on the trunk-ns button within 10 seconds.

Interactions

The following features interact with the Night Service — Trunk Group feature.

- LDN

  In the System Night Service mode, all incoming LDN calls, except those using DID trunks, which have activated night service are redirected to their corresponding trunk group's NSE. Incoming LDN calls using DID trunks are directed to the Night Console Service, Night Station Service, or Trunk Answer From Any Station feature, respectively, whichever applies first. Non-LDN DID trunk calls terminate at the dialed extension.

- Call Forwarding — All Calls

  If the Trunk Group Night Service mode and the trunk group's NSE has Call Forwarding — All Calls activated, the night service calls terminating to that NSE are forwarded to its designated extension.

- Forced First Announcements

  An interaction occurs with system night service and Forced First Annoucement. For example, if hunt group A has a forced first announcement (delay =0), assign the incoming CO trunk to terminate at hunt group A. Assign the incoming trunk's night service destination to be another hunt group (hunt group B). Assign a night service button to the attendant.

With night service activated on the attendant, the incoming CO call routes to the night service destination hunt group B and does not play the Forced First Announcement of the incoming destination's hunt group A.

Without night service active, the incoming CO call routes to the incoming destination hunt group A and plays the Forced First Announcement.

Administration

Individual Trunk Group Night Service is administered on a voice terminal basis or attendant console. The following items require administration:

- Assign ''trunk-ns'' button(s) to designated voice terminal(s). Up to three buttons can be assigned to voice terminals in each trunk group. The trunk group number must be assigned for each button.

  If a ''trunk-ns'' button is assigned for an existing trunk, it is updated immediately to show the status of the trunk.

- Assign ''trunk-ns'' button(s) to attendant console(s). Three buttons per attendant console are allowed. The trunk group number must be assigned for each button.
- Permission to allow attendant seeking calls other than DID LDP calls redirect to the DID Listed Directory Number (LDN) night extension, if desired.

The system can have extension numbers for Night Service per trunk group member that take precedence over the one assigned to the group. The system can have Trunk Group Night Service and split Night Service at the same time, but the calls are redirected to the trunk group's NSE before it goes to the hunt group/split's NSE.

Hardware and Software Requirements

No additional hardware or software is required.

Off-Premises Station

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows a voice terminal located outside the building where the switch is located to be connected to the system. If CO trunks are used, the voice terminal must be analog and must be FCC-registered (or, outside the US, registered by the appropriate governmental agency).

Considerations

Off-Premises Stations are useful whenever it is necessary to have a voice terminal located away from the main location.

The maximum loop distance for Off-Premises Stations is 20,000 feet (6093.34 meters) without repeaters.

Interactions

The Distinctive Ringing feature might function improperly at an Off-Premises Station due to the distance. However, the Distinctive Ringing feature can be disabled when the Off-Premises Station is administered. If the Distinctive Ringing feature is not used with an Off-Premises Station, the terminal will receive one-burst ringing for all calls.

Administration

Off-Premises Stations are administered by the System Manager.

Off-Premises Stations are administered the same as on-premises voice terminals with the following exceptions:

- For voice terminals used as Off-Premises Stations the “Off-Premises Station” field must be administered as `yes`.
- For voice terminals used as Off-Premises Stations the “R Balance Network” field must be completed.

Hardware and Software Requirements

Requires cross-connecting capabilities or one port on a TN747 or TN464B/C/D DS1 Tie Trunk circuit pack for each interface to be provided.

NOTE:
TN767 supports Mu-Law, TN464B supports A-Law, and TN464C/D supports both Mu-Law and A-Law.

No additional software is required

PC Interface

Feature Availability

This feature is available with all Generic 3 releases.

Description

The PC (Personal Computer) Interface feature consists of the PC/PBX Connection, the PC/PBX Platforms, and PC/ISDN Platform product family. These products are used with the and DEFINITY System Generic 3 switches to provide users of AT&T PCs and other IBM compatible PCs fully integrated voice and data work station capabilities. PC/PBX Connection is a voice and data call management application.

NOTE:
See the AT&T Catalog or your account team for other applications available on the platforms.

The platforms consist of PC interface cards and open voice and data Applications Programming Interfaces (API). These APIs allow a variety of applications to operate on the PC and communicate with the switch.

Configurations

Several different configurations are available for the PC Interface feature. For convenience in referencing, these break down into three groups. Groups 1 and 2 use the DCP (Digital Communications Protocol) while Group 3 uses the ISDN-BRI (Basic Rate Interface) protocol.

Configuration Group 1

Group 1, consists of those DCP PC/PBX configurations that use a PC Cartridge in a 7404D voice terminal (manufacture discontinued) to communicate with the switch. This group was formerly called package 1 (now Release 3.0X) and package 2. Group 1 is shown in Figure 3-26. (Please note that this configuration is not available for new installations.)

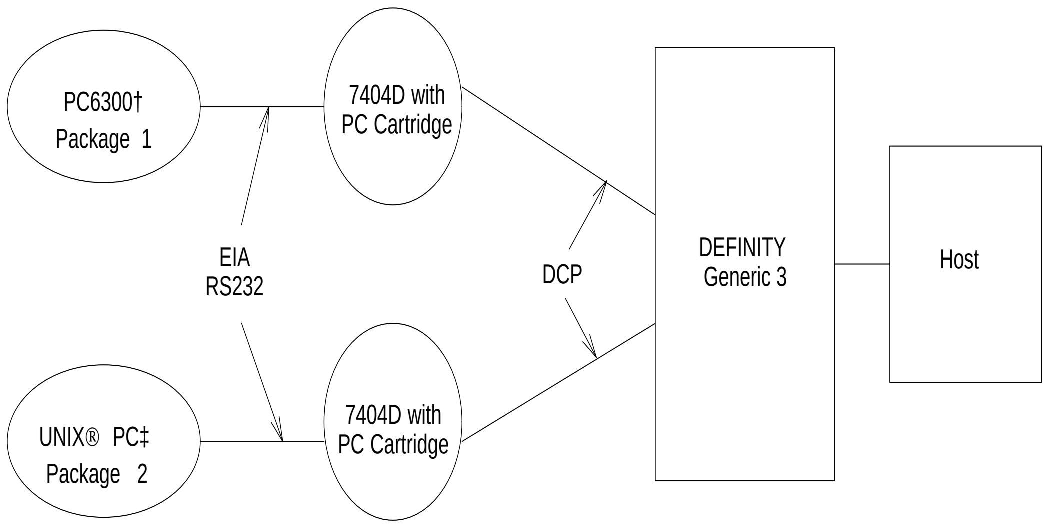


Figure 3-26. DCP PC Interface Configuration (Group 1)

[1] Other IBM-compatible PCs can be used, as well as the PC 6300.

[2] UNIX PCs were originally introduced but are no longer supported by the PC Interface feature.

Configuration Group 2

Group 2, consists of those DCP PC/PBX Connection configurations that use the PC/PBX Interface Card (formerly DCP expansion card) in the PC itself to provide the communications interface with the switch. Group 2 uses the DCP protocol and is described in detail in *PC/PBX Connection Release 3*, 555-016-715, and *PC/PBX Platform Installation and Reference*, 555-016-101. Group 2 was formerly called packages 3, 4, 5, and 6 and now are combined as Release 3.0X. Group 2 is shown in Figure 3-27.

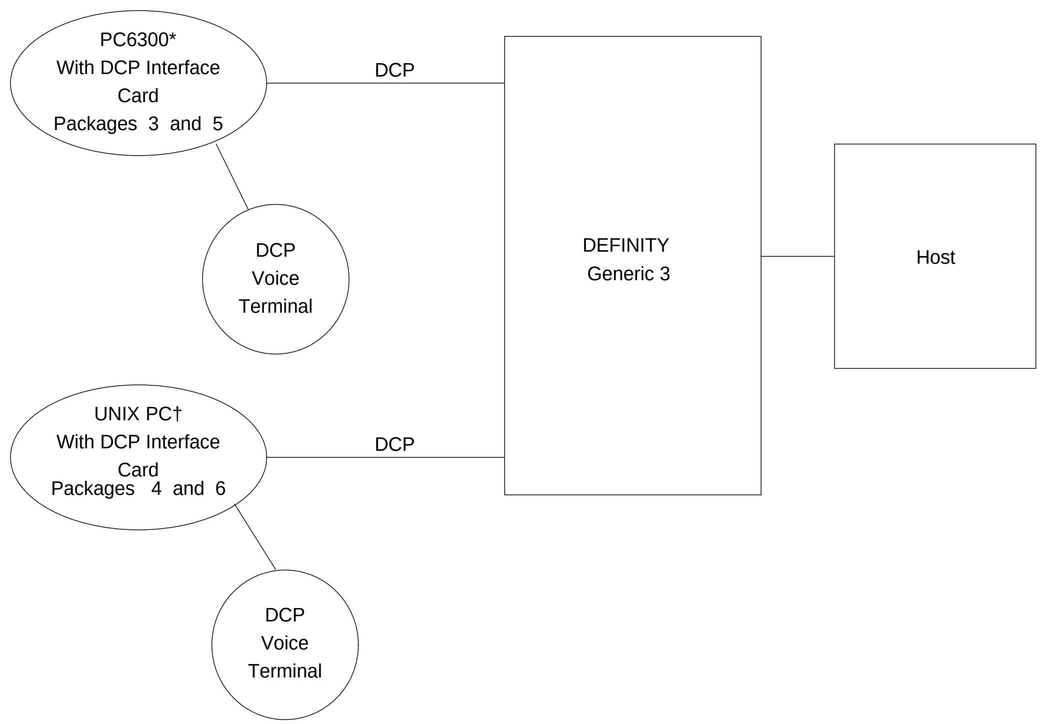


Figure 3-27. DCP PC Interface Configuration (Group 2)

[1] Other IBM-compatible PCs can be used, as well as the PC 6300. PCs that use the MicroChannel. bus (IBM Personal System/2 TM Models 50, 55, 60, 70, 80, or 100 percent compatible) can use the PC/PBX MicroChannel Platform Interface card.

[2] UNIX PCs were originally introduced but are no longer supported by the PC Interface feature.

Configuration Group 3

Group 3 consists of those configurations that use the ISDN-BRI (Basic Rate Interface) for communications. Connectivity to the switch is provided by the PC/ISDN Interface Card installed in the PC itself. Possible arrangements in this group include the PC as a stand alone terminal (PC only), or with from one to four voice terminals, handsets, or headsets. Group 3 is available on the DEFINITY switch Generic 3, but not on previous System 85 switches. Group 3, described in detail in the *PC/ISDN Platform Installation and Reference* manual, 555-016-102, is shown in Figure 3-28.

Applications and Benefits

PC Interface users receive benefits provided by both the PC and the features and services of the DEFINITY Generic 3 switch.

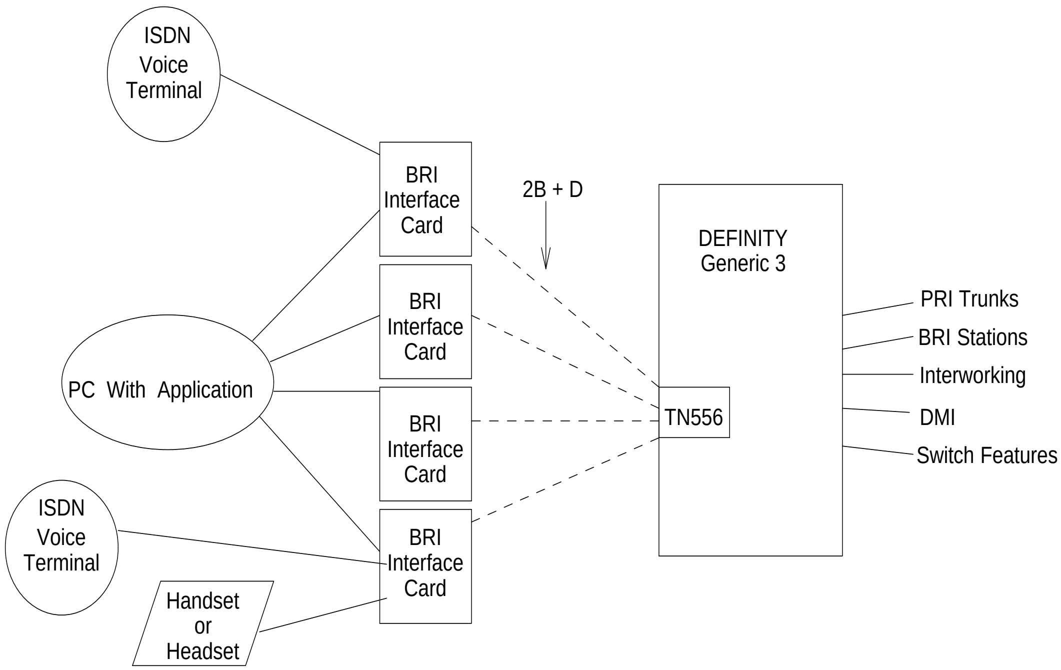


Figure 3-28. ISDN—BRI PC Interface Configuration (Group 3)

Switch Features and Services

PC Interface users have multiple appearances (depending on the software application used) for their assigned extension. One or more of these appearances can be designated for use with data calls. With the ISDN-BRI version, up to four separate PC/ISDN Interface Cards can be used on the same PC. Each interface card used can be assigned a separate extension number, and each of these extension numbers can have multiple appearances. The availability of specific features depends on the class of service of the assigned extension and the system class of service for the switch. Modem Pooling should be provided to ensure general availability of off-net data calling services.

PC Features and Services

- PC/PBX Connection

  Provides an integrated voice and data workstation interface from the System to an AT&T PC 6300 or other compatible computer. The connection provides access to a variety of host computers, allowing the PC to act as a terminal for the host.

  To administer a PC/PBX connection, the PC/PBX connection must be completed.

- Directory Service

- Last Number Dialed

  Features and services provided by the PC are a function of the PC software application in place. With the PC/PBX Connection software, they include PC based Directory Services (on line), a Last Number Dialed feature (separate from these features as provided by the switch), and basic asynchronous terminal emulation.

- 3270 Emulation

- Hayes Smart Modem Emulation

- E78 Plus/ISDN

  Other options include such functions as synchronous 3270 Emulation and Hayes Smart Modem Emulation. The E78 Plus./ISDN (no longer available) software package provides enhanced software based 3270 emulation with high speed (64 Kbps) connections (through a 3270 data module) to a local or remote 3270 cluster controller. This includes bulk file transfer capability over switched links including either ETN (Electronic Tandem Network) or ISDN circuits. Expensive fixed coaxial connections can be avoided. The user interface with the E78 Plus/ ISDN software is the same as with the widely used E78/IRMA™ (no longer available) hardware configurations.

  Refer to the AT&T document *PC/PBX Platform Installation and Reference* 555-016-101 (for Group 2) and to *PC/ISDN Platform Installation and Reference* 555-016-102 (for Group 3) for details on PC software options and features, and applications. Also see the AT&T Catalog or your account representative for more information on available applications.

  For additional PC/PBX user information, refer to *AT&T PC/PBX Connection Package 3 User's Guide,* 555-016-703. A complete set of documents is available as *AT&T PC/PBX Connection Release 3 Documentation Set*, 555-016-715.

  The PC/PBX Platforms and PC/ISDN Platform offers local custom applications programming capability. A separately orderable publication, the *PC/ISDN Interface Software Developer's Guide*, 555-016-103, provides the information that you need to customize existing applications software or to develop new applications software to meet local needs.

Feature History and Development

The DCP versions of this feature (Groups 1 and 2) were first supported in System 85 R2 V4, as the PC/PBX Connection feature. For System 85 R2 V4 and later switches, PCs have their own terminal type (PC). On earlier versions of the Release 2 System 85 switch, PC/PBX configuration Groups

1 and 2 can be used by administering them as an alias. Use terminal type BCT 510D for Release 2, Versions 2 and 3. Use 7405D with display module for Release 2, Version 1 switches.

With DEFINITY system Generic 3, the feature name is changed to PC Interface to avoid confusion with the PC/PBX Connection software package which is part of the overall offering. PC Interface Group 3, the ISDN-BRI version of this feature, is first available on DEFINITY system Generic 2.1. The PC/ISDN Platform (Group 3) cannot be used with System 85 switches.

User Operations

User operations depend on the software used with the PC. For specific user operations, refer to the appropriate PC Interface documentation set as follows:

- 555-016-715, PC/PBX Connection

  Documentation Set for the DCP versions (Groups 1 and 2) using the PC/PBX Connection software package.

- 555-016-101, PC/PBX Platforms

  Installation and Reference manual for PC/PBX Platforms.

- 555-016-102, PC/ISDN Platform

  Installation and Reference Manual for the PC/ISDN Platform (Group 3).

Considerations

- Use of Speakerphones

  Most DEFINITY system Generic 3 speakerphones can be used with the PC Interface.

- Function Key Module

  The Function Key Module of the 7405D can be used with the PC Interface feature.

- SPID (Service Profile Identifier)

  On the DEFINITY switch Generic 3, BRI terminals are normally ‘‘initializing terminals’’ and require that a SPID be assigned (*see the* "Integrated Services Digital Network (ISDN) — Basic Rate Interface (BRI)" feature).

  The PC/ISDN Platform (Group 3), in a stand alone configuration (no associated voice terminal), is a ‘‘non-initializing BRI terminal.’’ A non-initializing terminal does not require a SPID. When the PC/ISDN Platform is assigned as a stand alone non-initializing terminal, it is administered using a locally defined terminal type with General Terminal Administration, Procedure 50, Word 1 (see the *Administration Procedures Manual*, 555-105-506). When this is done, the terminal type is defined as a non-initializing terminal that does not support MIMs (Management

Information Messages). This requires a value of ‘‘0’’ in field 6 of Procedure 50, Word 1. Other specific characteristics of the locally defined terminal type will depend on the telephone manager applications software used with the PC.

The PC/ISDN Platform can also be assigned with an associated (initializing) ISDN-BRI voice terminal (such as an ISDN 7505) that uses a SPID. In this case, the station should also be assigned using a locally defined terminal type to take full advantage of the capabilities of the PC Interface. This terminal type should also be non-initializing with no support of MIMs.

- Internal Features

  Specific internal features available depend on the application package used. The PC/PBX Connection product provides extensive local directory capability (up to 32,000 entries). They also provide a Last Number Dialed capability that is separate from the Last Number Dialed feature on the DEFINITY system Generic 3.

- Data Modules

  Voice terminals with data modules are not recommended for use with the PC Interface feature (except for 3270 Data Modules when 3270 emulation is used). If a DCP data module (such as a DTDM) or ISDN data module (such as an ADM-T) is attached to the voice terminal used in conjunction with the PC Interface card (either DCP or ISDN-BRI), the data module will be bypassed (not used). All interface functions are performed by the interface card even if a data module is present.

- Display Modules and Terminals

  Display modules can be used. See ‘‘Feature History’’ for administration limitation.

- Call Appearances

  The PC/PBX Connection package supports five appearances, one of which must be dedicated to data use. This configuration limit may not be optimal for some users (such as attendants) who need many call appearances. The PC/PBX Platforms can support the maximum allowed by switch administration (52). The PC/ISDN Platform supports many more call appearances. On the DEFINITY system Generic 3, the PC/ISDN Platform is limited by switch administration to a maximum of 208 call appearances (52 call appearances per interface card X a maximum of four interface cards = 208 appearances). Note that each extension number is limited to 12 appearances. This means that each interface card can accommodate four and one third extension numbers (without any bridged appearances). Special applications software must be installed on the PC to provide access to all these appearances.

- 7404D Voice Terminal

  The 7404D voice terminal with *messaging* cartridge cannot be used with the PC Interface. The 7404D is manufacture discontinued and may no longer be ordered. The 7404D with PC cartridge is used only with Group 1.

Interactions With Other Features

The PC Interface feature interacts with most other features in the same way as other DCP and ISDN-BRI voice and data terminals.

- Data Communications Access

  The PC Interface feature uses a digital interface (either DCP or ISDN-BRI) and is not directly compatible with the Data Communications Access feature which uses an analog interface. Modem Pooling like conversion must be applied if these two features are to be used together.

- Data Protection

  The PC Interface feature is used for data communications, therefore Data Protection—Permanent should be assigned.

- Host Computer Access

  Both the PC Interface feature and the Host Computer Access feature use digital interfaces. These features are directly compatible (no Modem Pooling conversion needed).

- ISDN-BRI (Basic Rate Interface)

  The ISDN-BRI feature must be active on the switch to use the PC/ISDN Platform (Group 3). With ISDN—BRI configurations, up to four PC/ISDN interface cards can be installed in one PC. When multiple cards are used, each card is assigned to a separate and distinct interface on the switch. Separate interface cards cannot share the same ELL (Equipment Line Location). Each separate interface card can have its own separate voice terminal or voice-calling device. When a voice terminal is used, special applications software is not required on the PC. However, to use a handset or headset alone, special applications software is needed.

- Modem Pooling

  Modem Pooling is needed if the PC Interface feature is used to place calls to, or receive calls from, off-premises stations over analog trunks.

Restricting Feature Use

The voice and data appearances used by the PC Interface are subject to the same general restrictions as other like appearances. These can be applied either as fixed restrictions through the extension class of service, or as temporary restrictions through the Attendant Control of Voice Terminals feature. Other restrictive measures, such as the FRL (Facilities Restriction Level) and Attendant Control of Trunk Group Access features also apply to stations using the PC Interface feature.

Hardware Requirements

The specific hardware required by the PC Interface varies depending on the configuration group used.

Group 1

Configuration Group 1 consists of those configurations that use the DCP PC interface cartridge in the 7404D voice terminal.

- 7404D VDS (Voice Data Station)-*Manufacture Discontinued*
- 31815 PC cartridge
- Standard, EIA RS232C connecting cables
- One of the following Personal Computers:
  - AT&T PC 6300 and compatibles, with MS-DOS™* Version 2.11 or later
  - AT&T PC 6300 Plus, with MS-DOS Version 3.1 or later
  - IBM PC and PC/XT™, with PC-DOS™ Version 2.0 or later existing versions
  - IBM PC/AT™, with PC-DOS Version 3.1 or later
- PC Accessories:
  - Serial asynchronous communications port (standard on AT&T PC 6300 and 6300 Plus)
  - 256K RAM minimum (PC/PBX Connection application package)

    or

    384K RAM recommended minimum for integrated software applications such as concurrent Lotus 1-2-3.™.

Group 2

Configuration Group 2 consists of those DCP configurations that use the PC Interface card in the PC itself.

- Most DEFINITY system Generic 3 7400-series (DCP) digital voice terminals can be used
- PC/PBX Interface Card
- One of the following Personal Computers:

- — AT&T PC 6300 series and compatibles, including PC 6300, 6300 WGS, 6300 Plus, 6310, 6312 WGS, 6286 WGS, and 6386 WGS, with MS-DOS Versions up to 4.X (except 3.0) DOS releases beginning with 1 (for example 1.XX) cannot be used.
- — IBM PC, PC/AT, PC/XT, and Personal System/2 Model 30 and Model 30/286 with PC-DOS Versions up to 4.X (except 3.0)
- — IBM Personal System/2 models 50, 60, 70, 80, or 100% compatible
- — Memory Configurations (depending on application being run): 320K RAM minimum or 448K RAM recommended for large integrated software applications such as concurrent Lotus 1-2-3.

Group 3

Configuration Group 3 consists of BRI configurations that use a PC expansion card in the PC itself (there is no BRI cartridge option).

- Voice Terminal or Voice Calling Devices
  - — Any DEFINITY system Generic 3 ISDN-BRI Voice Terminal (7500 Series) can be used with the Group 3 configurations.

    or
  - — With appropriate applications software, an AT&T R-Type replacement Handset (PEC X10150) can be plugged into the ISDN-BRI PC expansion card instead of a BRI voice terminal.

    or
  - — With appropriate software, a headset can be used instead of either a BRI voice terminal or the R-Type Handset. The following Plantronics headsets can be used:

    StarSet. Series Communications Headset model StarMate. MH0228-3

    Supra. Series Communications Headset model MH0528-3

    Supra Series Communications Headset model MH0529-3
- PC/ISDN Expansion Card (up to four per PC)
- One of the following personal computers:
  - — AT&T Personal Computers

    AT&T PC 6300 (and compatibles)

    AT&T PC 6300 Plus

    AT&T 6300 WGS

    AT&T PC 6310

    AT&T PC 6312 WGS

AT&T PC 6286 WGS

AT&T PC 6386 WGS

— IBM Personal Computers

IBM PC

IBM PC/XT

IBM PC/AT

PS/2™ Model 30

PS/2 Model 30-286

— Compaq™ Personal Computers

Compaq DeskPro286

Compaq DeskPro386

- PC Accessories:
  - For AT&T, Compaq, and compatible PCs, 2.0 and all later releases (except 3.0) of MS-DOS
  - For IBM PCs, 2.0 and all later releases (except 3.0) of PC-DOS

System Connections

Group 1

- The RS232C port of the 7404D voice/data set is connected via an EIA RS232C cable to the RS232C port of the PC.
- The 7404D voice/data set has a modular digital port connection to the DEFINITY system Generic 3.
- Standard operating distance of the 7404D from the PBX is 5,000 feet maximum for 24-gauge wire or 4,000 feet for 26-gauge wire.

Group 2

- The PC Interface card plugs into an expansion slot on the PC. The card has two standard, 8-pin modular jacks (line and phone).
- The digital phone plugs into the phone jack on the PC Interface card.
- The line jack on the card provides a digital port connection to DEFINITY system Generic 3.
- Standard operating distance of the PC Interface card from the PBX is 5,000 feet maximum for 24-gauge wire or 4,000 feet for 26-gauge wire.

Group 3

- The PC/ISDN Interface card (from one to four cards) plugs into an expansion slot on the PC. The card provides two standard 8-pin modular jack connections for both a line connection (to the switch) and a phone connection. A standard 4-pin modular jack is also available for use with a hand-set or head-set rather than a voice terminal.
- Each expansion card provides separate and distinct line and phone connections.
- Standard and maximum operating distances for ISDN-BRI terminals are a function of a number of factors including loop signal loss and power. The factors affecting operating distance limits are discussed in detail under Distance Specifications in the *DEFINITY Communications System Generic 3 System Description and Specifications*, 555-230-206.

Feature Administration

Administration guidelines in this section pertain to the DEFINITY switches Generic 3. For additional information and for information on administering the PC itself, refer to the *PC/PBX Connection Installation and Reference* (part of documentation set 555-016-715), *PC/PBX Platform Installation and Reference*, 555-016-101, or *PC/ISDN Platform Installation and Reference*, 555-016-102.

The PC Interface feature is administered on a per-station basis.

On DEFINITY switches Generic 3, this feature is assigned using the DEFINITY system Manager II.

This feature can also be administered using the Manager IV.

Table 3-72 shows the applicable administration procedures.

Table 3-72. PC Interface Administration Procedures

Procedure	Words	Purpose	SMT
050	1 & 2	Used for General Terminal Administration to locally define new terminal types (PC/ISDN Platform).	N/A
051	1	Assigns and removes terminal types.	Yes
051	2	Assigns the SPID (Service Profile Identifier) used with the voice terminal portion of the PC/ISDN interface.	Yes

Continued on next page

Table 3-72. PC Interface Administration Procedures — *Continued*

Procedure	Words	Purpose	SMT
052	1	Assigns extension numbers and images to an ELL and to specific buttons.	Yes
052	2	Assigns additional extension and appearance (image) characteristics to ELL and buttons.	Yes
054	1	Assigns miscellaneous feature buttons, including the *Wait For Principal* button to a multiappearance voice terminal. The Wait For Principal button is used only with the PC Interface feature in a Message Center role.	Yes
054	2	Assigns and removes Custom Calling feature buttons.	Yes
054	4	Assigns and removes Display feature buttons.	Yes
055	1	Assigns and removes Terminal Busy feature buttons.	Yes
055	2	Assigns and removes One Button Transfer feature buttons.	Yes
056	1	Assigns and removes Intercom buttons.	Yes
057	1-3	Assigns and removes CO line buttons.	Yes
058	1	Performs a station swap between two equipment locations.	Yes
059	1-5	Assigns and removes Abbreviated Dialing buttons.	Yes
063	1	Assigns and removes Automatic Message Waiting buttons.	Yes
070	4	Displays information about the terminal equipment assigned to an equipment location.	Yes

PC/PBX Connection

Feature Availability

This feature is available with all current Generic 3 releases except new installations.

Description

PC/PBX Connection is a PC application that runs on the PC/PBX platform. Refer to the section "PC Interface" in this manual for more details. This feature brings the voice terminal and PC together into an integrated voice and data workstation. The PC can be any IBM-compatible PC.

AT&T provides the hardware for this connection. The hardware consists of a PC expansion card and a 7-foot modular cord. The software includes a variety of packages available from different vendors. Contact your AT&T representative for more information.

For information on the UNIX PC, see the section entitled "PC Interface" in this manual.

Security Measures

There are two areas where unauthorized use may develop with this feature. The first involves unauthorized users who come to a PC and attempt to make calls from it. The software in the PC installed by the feature has a security setting. A user can place the PC in Security Mode when it is to be left unattended. Automatic Security can also be optioned by the administration program on the PC. With Automatic Security enabled, when the software is executed, it brings up the program in security mode. This mode is password-protected on the PC. The password should be changed often, and records of it should be secured.

The second area involves remote access to the PC over its data extension. Files stored in the PC can be deleted or copied with this feature. This can also be password-protected, and this password should be changed frequently and protected. Consult the *GBCS Product Security Handbook*, 555-025-600, for additional steps to secure your system and to find out about obtaining information regularly about security developments.

Considerations

By providing PC users with the voice and data capabilities of a fully integrated voice/data workstation, the PC/PBX Connection feature makes communications more efficient. Also, PC users with the PC/PBX Connection feature are linked for easy access to other PCs, modem pooling, and on- and off-site computers.

Interactions

When a station is optioned for PC/PBX on the station form, and if the set is a display set, the call display is slightly altered. The appearance identifier (a=, b=, c=, d=) is omitted. If the Call Log feature is enabled in the PC/PBX Connection software, the display information is captured to disk. The log can be searched. The search criteria begins at the left of the displayed information. By omitting the appearance designator, a user can enter a search string without contending with the appearance identifiers in the displayed information.

Administration

The PC/PBX Connection feature is administered on a per extension basis by the System Manager. A PC is assigned to the system just as any other station would be with the station type administered as ‘‘pc.’’ An additional field is then specified for the type of digital voice terminal to be connected to the PC.

Hardware and Software Requirements

A port on a TN754 Digital Line Circuit Pack (TN413, TN754B) is required for each PC to be connected.

Personal Central Office Line (PCOL)

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides a dedicated trunk for direct access to or from the public network for multiappearance voice terminal users.

Each PCOL can have appearance at multiple multiappearance voice terminals. Users assigned this feature press the PCOL feature button to answer and place calls — dial access is not provided. The status lamp associated with the PCOL button indicates the busy or idle status of the trunk.

An incoming PCOL call rings all voice terminals assigned the feature (ringing can be either audible or silent, depending on administration). The PCOL button status lamp flashes even if all call appearances at the voice terminal are active. If a call appearance is idle, the status lamp associated with that appearance also flashes.

CO, FX, and WATS trunks can be assigned to this feature.

PCOLs are not assigned a COR.

Considerations

PCOLs are useful to users such as executives, dispatchers, or buyers with a high volume of calls going outside the system, and businesses with specialized incoming calls (such as a service department).

Interactions

The following features interact with the PCOL feature.

- Abbreviated Dialing

  Abbreviated Dialing can be used with the PCOL feature. However, the accessed lists are associated with the individual voice terminals.

- Bridged Call Appearance

  If a user is active on his or her primary extension number on a PCOL call, bridged call appearances of that extension number cannot be used to bridge onto the call. The call can only be bridged onto if another voice terminal is a member of the same PCOL group and has a PCOL button.

- Call Coverage

  AUDIX cannot be in the coverage path of a PCOL group.

- Hold

  When a user, active on a PCOL call, puts the call on Hold, the status lamp associated with the PCOL button does not track the busy/idle status of the PCOL.

- LWC

  LWC messages can be stored for a PCOL group. The messages are retrieved by an authorized systemwide message retriever. When a message is stored, the remote Automatic Message Waiting lamp assigned for the PCOL group lights. One remote Automatic Message Waiting lamp is allowed per group.

- CDR

  The CDR feature can be activated for PCOL calls, but the CDR record does not specifically identify the call as PCOL. A PCOL call can, however, be identified by the trunk access code used on the call. The call is recorded to the extension number assigned to the voice terminal where the call was originated or answered.

- Send All Calls

  Send All Calls cannot be activated for a PCOL group.

- Temporary Bridged Appearance

  When a PCOL is shared (assigned to a group), any group member can bridge onto a PCOL call through the Temporary Bridged Appearance feature. The Privacy — Manual Exclusion feature can be activated on such a call if the voice terminal is assigned an Exclusion button.

- Transfer

  A PCOL can be transferred to an extension that does not have a button for that PCOL.

The following features cannot be used with the PCOL feature:

- ARS
- Call Forwarding All Calls
- Ringback Queuing

Administration

PCOLs are administered by the System Manager. The following items require administration:

- Group number
- Group type (CO, FX, or WATS)

- Group name (optional, used for display purposes)
- Data Restriction activation
- CDR activation
- Call Coverage path (redirection criteria can be Don't Answer and Cover All Calls)
- Extension numbers of voice terminals assigned to PCOL group (up to four terminals can share a PCOL)
- PCOL button (per terminal assigned to the PCOL group)
- Exclusion button (optional on a per-terminal basis)
- Remote Automatic Message Waiting lamp (one allowed per PCOL group)
- Audible or silent ringing

The following items can be administered for the CO, FX, or WATS trunk used for the PCOL:

- Circuit pack port number
- Trunk type
- Trunk name (for display purposes)
- Trunk access code (nondialable, used to identify the trunk for CDR)
- Outgoing dialing type
- CO disconnect timing
- Terminating area code
- Prefix for code conversion
- Toll table index for code conversion
- Prefix 1 (needed for CO and FX trunks if the prefix 1 is needed for toll calls)
- Timers based on board capabilities

NOTE:
Timer administration is available for G3i-Global, G3V2, and later releases.

Hardware and Software Requirements

Requires one port on a TN747 CO Trunk circuit pack for each CO, FX, or WATS trunk assigned as a PCOL. No additional software is required.

Personalized Ringing

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows users of certain voice terminals to uniquely identify their own calls. Each user can choose one of a number of possible ringing patterns.

The eight ringing patterns are tone sequences consisting of different combinations of three tones. The eight different combinations are listed below. The tones are heard in the sequence given for each combination.

- 750 Hz, 750 Hz, 750 Hz (normal ringing)
- 1060 Hz, 1060 Hz, 1060 Hz
- 530 Hz, 530 Hz, 530 Hz
- 530 Hz, 1060 Hz, 1060 Hz
- 1060 Hz, 1060 Hz, 530 Hz
- 1060 Hz, 530 Hz, 530 Hz
- 1060 Hz, 530 Hz, 1060 Hz
- 530 Hz, 1060 Hz, 530 Hz

Each ringing pattern requires 0.6 second (0.2 second for each tone) in the 5.2 seconds ringing cycle. This 0.6 second of personalized ringing occurs at the given time during the ringing cycles of the following types of calls (times indicated are in seconds):

- Internal voice terminal, internal tie trunk, and remote access calls

  0.6 on, 0.6 personalized ringing, 4.0 off
- Attendant extended, attendant originated, and incoming trunk calls, including external tie trunk calls

  0.2 on, 0.4 off, 0.6 personalized ringing, 4.0 off
- Automatic Callback, Priority Calling, and Ringback Queuing Callback calls

  0.1 on, 0.1 off, 0.1 on, 0.3 off, 0.6 personalized ringing, 4.0 off
- Intercom Calls (7404D and 7407D voice terminals only)

  0.6 personalized ringing, 4.6 off

One of the eight ringing patterns can be specified for each eligible voice terminal (7303S and 7305S) by the System Manager. In addition, the 7404D, 7406D, 7407D, 7410D, 7505D, 7506D, 7507D, 8503T, and 7103A programmable voice terminal users have the capability of setting their own ringing pattern. The 7404D user can select the desired ringing pattern via the given menu options. The 7406D or 7407D user can select the desired ringing pattern by using the Select Ring and PR (Personalized Ringing #) buttons. The 7103A programmable voice terminal user can select one of four ringing patterns via a slide switch on the voice terminal.

Considerations

With Personalized Ringing, users working closely in the same area can each specify a different ringing pattern. This enables the users to distinguish their own ringing voice terminal from other voice terminals in the same area.

Up to eight different ringing patterns are available.

Interactions

The normal ringing cycles are altered as described in the Description section of this feature. With Administrable Distinctive Ringing (G3i-global, G3r, G3V2, and later releases) the association between the number of ring bursts and the call types is administrable.

Administration

Personalized Ringing is administered for the 7303S and 7305S voice terminals on a per-voice terminal basis by the System Manager. Administration consists of assigning one of the eight ringing patterns to each eligible voice terminal. Also, a 7404D, 7406D, 7407D, 7410D, 7505D, 7506D, 7507D, 8503T, or 7103A programmable voice terminal user can specify his or her own ringing pattern. The user specified ringing pattern for a 7404D, 7406D, or 7407D, however, is lost in the event of a power failure. The user specified ringing pattern for a 7410D is lost if the set loses auxiliary power.

Hardware and Software Requirement

No additional hardware or software is required.

Power Failure Transfer

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides service to and from the local telephone company CO, including WATs, during a power failure.

Considerations

Power Failure Transfer provides certain voice terminals with the capability to access the local CO and to answer certain incoming calls during a power failure. These voice terminals can be used to make or answer important or emergency calls.

Each voice terminal can be connected to a separate CO trunk for the Power Failure Transfer feature. The Power Failure Transfer feature is available in multiples of five.

Local CO trunks (including incoming WATS lines) can be used for Power Failure Transfer.

The 500-type (rotary dial) or 2500-type (touch-tone or DTMF) voice terminals must be used for Power Failure Transfer. Rotary dialing must be used if the local CO accepts dial pulses only. When a G3 system is not in the power failure mode, power failure transfer terminals (500-type rotary dial) can be used as regular extensions.

Interactions

During the Power Failure Transfer mode, no other system features can be activated.

If Night Service is activated and a power failure occurs, the system, when brought back up, automatically returns to the Night Service mode.

Administration

None required.

Hardware and Software Requirements

One emergency transfer panel is required for every five or six trunks assigned to Power Failure Transfer, depending on the transfer panel used. Two emergency transfer panels are available:

- Z1A Panel — Each unit serves up to six power failure transfer terminals. A ground-start key is required at each preselected voice terminal when ground-start trunks are used.
- Porta-Systems Model 574-5 Panel — Each unit serves up to five failure transfer terminals. The unit provides automatic ground start or loop start.

No additional software is required.

Priority Calling

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides a special form of call alerting between internal voice terminal users. The called voice terminal user receives a distinctive, administrable alerting signal (default is a three-burst alerting signal).

An active single-line voice terminal user who receives a Priority Calling call hears a distinctive priority Call Waiting tone (the number of bursts is administrable; see ‘‘Distinctive Ringing’’ for details).

A multi-appearance voice terminal user receives the Priority Calling call on an idle call appearance. If all call appearances, including the call appearance normally reserved for call origination, are active, the caller receives a busy tone. If the call appearance normally reserved for call origination is the only idle call appearance, an incoming priority call rings at that call appearance.

A user activates priority calling by dialing a Priority Calling access code or pressing a Priority button, followed by the desired extension number. DCS priority calling from the attendant station is *not* available.

Whether or not a user can activate Priority Calling is determined by the user’s COS.

Considerations

With Priority Calling, a voice terminal user can ring another voice terminal with a distinctive signal that tells the called party the incoming call requires immediate attention. The called party can then handle the call accordingly. DCS priority calling from the attendant station is *not* available.

Call Coverage Consult calls and callback calls from Automatic Callback and Ringback Queuing Attendant Intrusion are Priority Calling calls.

Interactions

The following features interact with the Priority Calling feature.

- Abbreviated Dialing

  If a priority call is to be made to a number in an abbreviated dial list, an Abbreviated Dial button must be used. The Abbreviated Dial Feature Access Code is not valid after priority calling has been activated.

- Automatic Callback and Ringback Queuing

  Callback calls do not redirect, do not forward, and cannot be picked up by a Call Pickup group member.

- Bridged Call Appearance

  A Bridged Call Appearance receives ringing on a priority call the same as the called primary extension.

- Call Coverage

  Priority Calling calls do not redirect to coverage unless the caller activates Go To Cover. If the call redirects, it remains a Priority Call, and the covering user receives a distinctive (default is three-burst) ringing signal.

- Call Forwarding All Calls

  Priority Calling calls (except callback calls) are forwarded, and the forwarded call remains a Priority Calling call.

- Call Waiting Termination

  A Priority Calling call waits on an active single-line voice terminal even if the Call Waiting Termination feature is not assigned to the voice terminal. The active single-line voice terminal user receiving the call hears a distinctive (default is three-burst) priority Call Waiting tone.

- Consult

  A Consult call acts as a priority call and waits at a single-line voice terminal, even if the single-line voice terminal does not have Call Waiting Indication assigned.

- Dial Access to Attendant

  A Priority Calling call cannot be originated to the attendant. However, the attendant can originate Priority Calling calls.

- DCS

  On a DCS tandem call to a single-line voice terminal, the called party does not receive priority ringing if the calling party activates Priority Calling after he or she has already made the call. The called party in this situation only receives priority ringing if the calling party activates Priority Calling prior to dialing the extension.

- Ringing

  Single-line voice terminals (2500 series) can be administered so that distinctive signals are not provided. In this case, one-burst ringing is provided for Priority calls.

- Last Number Dialed

  If a priority call is to be made to the last number dialed, the Last Number Dialed button must be used. The Last Number Dialed Feature Access Code is not valid after priority calling has been activated.

Administration

Priority Calling is administered by the System Manager. The following items require administration:

- Priority Calling access code or priority button
- Permission to activate Priority Calling (per COS)
- Type of alerting signal heard when this feature is used. (The default is a three-burst alerting signal.)

Hardware and Software Requirements

No additional hardware or software is required.

Privacy — Attendant Lockout

Feature Availability

This feature is available with all Generic 3 releases.

Description

Prevents an attendant from reentering a multiple-party connection held on the console unless recalled by a voice terminal user.

Considerations

Privacy — Attendant Lockout provides privacy for parties on a multiparty call held on the console. The held parties can hold a private conversation without being interrupted by the attendant.

Interactions

The following features interact with the Privacy — Attendant Lockout feature.

- Trunk-to-Trunk Transfer

  Privacy — Attendant Lockout does not function when a call using the Trunk-to-Trunk Transfer feature is held on the console.

- Individual Attendant Access

  Privacy — Attendant Lockout applies only to attendant group calls. Individual attendant calls are not affected.

- Attendant Recall

  Use Attendant Recall to recall the attendant to a call that is originally extended by the attendant.

Administration

Privacy — Attendant Lockout is administered on a per-system basis by the System Manager. The only administration required is to administer whether or not attendant lockout is active.

Hardware and Software Requirements

No additional hardware or software is required.

Privacy — Manual Exclusion

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows multiappearance voice terminal users to keep other users with appearances of the same extension number from bridging onto an existing call.

Exclusion is activated by pressing the Exclusion button on a per-call basis. If the Exclusion button is pressed while other users are bridged onto the call, the other users are dropped from the call. The Privacy — Manual Exclusion feature is automatically deactivated when the Exclusion button is pressed a second time or when the party who activated Privacy — Manual Exclusion is dropped from the call.

Privacy — Manual Exclusion is used with the PCOL, TEG, and Bridged Call Appearance features.

Considerations

Privacy — Manual Exclusion prevents users who have an appearance of another terminal's extension from bridging onto that extension.

Interactions

The following features interact with the Privacy — Manual Exclusion feature.

- Bridged Call Appearance

  When Privacy — Manual Exclusion is activated, all other users are prevented from bridging onto the active call.

Administration

Privacy — Manual Exclusion is administered on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Exclusion button to the desired voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Property Management System (PMS) Interface

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides a communications link between the System and a customer-owned PMS. The PMS allows a customer to control certain features used in both a hospital-type and a hotel/motel-type environment.

The communications link allows the PMS to interrogate the system and allows information to be passed between the system and the PMS. Routine operations related to the following features are simplified through this message exchange capability:

- Message Waiting Notification
- Controlled Restriction
- Housekeeping Status
- Check-In/Check-Out
- Room Change/Room Swap.
- Names Registration
- Guest Information Input/Change
- Support of five-Digit Extension Numbers

Message Waiting Notification, Controlled Restriction, and Housekeeping Status are optional features. A customer may elect to operate each of these features from the system only or to operate each of these features from either the system or a PMS.

Check-In/Check-Out, Room Change/Room Swap, Names Registration, and Guest Information Input/Change are controlled from the PMS as long as the communications link between the system and the PMS is operational. If the link is not operational (the link is down), these features are affected as follows:

- Control of Check-In/Check-Out transfers to the system. With the system in control, Check-In/Check-Out operations are performed via feature buttons.
- Control of Guest Information Input/Change transfers to the system. With the system in control, Guest Information Input/Change operations are performed by the System Manager via system administration commands.

- The system does not support Room Change/Room Swap as such. However, the equivalent of Room Change/Room Swap is executed through the system by activating Check-Out followed by Check-In.
- Names Registration information which is normally sent automatically from the PMS to the switch, can be entered manually at the switch by the System Manager.

The PMS Interface provides the following:

- A communications protocol for controlling message exchange between the system and a PMS
- An application module for controlling the operation of the PMS features
- Status data on all guest/patient rooms for selected features

The protocol is full-duplex, asynchronous and provides the mechanisms for setting up a data session with the PMS, message exchange control, error identification, and recovery. The interface supports standard data rates (1200, 2400, 4800, or 9600 bps in G3i).

Two protocol modes are provided; the normal protocol mode as described above, and the transparent protocol mode. The transparent protocol mode supports ASCII character transmission and is required for PMS features such as Names Registration and Guest Information Input/Change. Systems may be administered to use either the normal or transparent protocol mode.

The application module of the PMS Interface implements requested features and provides backup procedures if the communications link between the PMS and the system is down. Whether or not the communications link with the PMS is operational, the system always maintains the following data for each room:

- Whether the room is vacant or occupied
- Whether the voice terminal's Message lamp is on or off
- Whether a Controlled Restriction is active at the voice terminal and, if so, which one
- The guest's name and coverage path

When the link to the PMS is down, the system automatically activates Check-In/Check-Out for the attendant console and front desk terminal with display capability, and continues to support PMS features that are activated from guest/patient room voice terminals.

When the link is again operational, the system sends one of the following messages to the PMS:

- No room status changes occurred during loss of communications.
- Room status changes did occur during loss of communications; therefore, a status data exchange is needed to synchronize the system and the PMS data bases.

- The system failed momentarily, destroying its record of room status; therefore, a room status data exchange (full transfer of data from the PMS to the system) is needed to synchronize the system and the PMS databases.

Also, when the PMS link is down or if a PMS is not used, the system maintains a log, called an audit trail report, of all events that would normally be sent to the PMS.

The audit trail data (accessed via the Management Terminal) is a sequential listing of all PMS transactions executed by the system when the PMS data link was down. Also included in the audit trail are some error events that may have occurred when the link was either up or down. If a printer is configured in the system, copies of the audit trail data will help the administrative staff to restore the room status of system and the PMS.

In addition to the PMS audit trail report, if the system has an operational PMS log printer and the PMS link is down, Housekeeping Status changes will be printed as they occur. The Housekeeping Status report will contain the following information:

- Room number
- FAC dialed
- Any additional information digits that were dialed
- Reason for the entry (error message)
- Time the error occurred

In addition to the PMS log printer, a PMS Journal/Schedule printer is also available. The PMS Journal/Schedule printer prints reports on Automatic Wakeup activity, Emergency Access to the Attendant activity, and scheduled reports.

A supporting function called Room Data Image synchronizes the switch and PMS databases after a PMS link goes down and comes back up. The information included in the Room Data Image is as follows:

- Room extension
- Whether the room is occupied or vacant
- Message Waiting lamp status
- Controlled Restriction status
- Guest’s name
- Call Coverage path

Message Waiting Notification

Message Waiting Notification requests are originated from attendant consoles, front desk terminals, or PMS terminals. When a request is entered, the PMS sends a Message to the system to change the state of the Message lamp associated with a certain extension number. If the Message lamp has been turned on by the AUDIX or Leave Word Calling (LWC), the PMS cannot be used to turn the lamp off. However, in the transparent mode, certain events may cause the switch to inform the PMS that the Message lamp has been turned on by AUDIX or LWC.

Any console or terminal used to activate and deactivate Message Waiting Notification must be assigned a Console Permissions Class of Service (COS). The affected extension must have a Client Room COS.

Controlled Restriction

When the Controlled Restriction feature is activated through the PMS, the PMS sends a message to the system to assign one of the following restrictions to the voice terminal in a guest/patient room:

- No restriction
- Outward restriction
- Total restriction
- Station-to-station restriction
- Termination restriction
- Combined outward and termination restriction
- Combined Outward and Station-to-Station restriction
- Combined Termination and Station-to-Station restriction

If a PMS is not used or if the communications link is down, the attendant or front desk user can still set the Controlled Restriction for a voice terminal, because activation of this feature is independent of the PMS. When the communications link is again operational (if a PMS is used), the system asks for a data base exchange and all status changes are sent to the PMS. At this time, controlled restrictions can still be assigned.

If user controlled restrictions are activated or deactivated from the switch, the PMS receives a message with this information.

Housekeeping Status

The housekeeping staff can enter status information using voice terminals located in guest/patient rooms or using designated terminals. Up to 10 Housekeeping Status Access Codes can be assigned in the system.

- Room Voice Terminal Access Code

  After the Access Code is dialed, the system accepts up to six additional information digits. These information digits can be used for items such as maid identification.

- Designated Voice Terminal Access Code

  After the Access Code is dialed, the system waits for the room extension to be entered and then will accept up to six additional information digits.

If a PMS is used, the system notifies the PMS when Housekeeping Status information is entered. If a PMS is not used, if the communications link is down, or if a PMS is connected, but housekeeper information is not sent (the "Housekeeper Information Configuration" field on the Hospitality-Related System Parameters screen is administered as act-nopms), then Housekeeping Status information is written to a log, can be accessed through the Management Terminal, and can be sent to a printer to obtain a hard copy. If the system has a PMS log printer and the PMS link is down, each entered event will be printed as it occurs.

If a PMS is used, but goes down, and a PMS log printer is not operational, the Housekeeping Status access codes cannot be used.

Check-In/Check-Out

A Check-In request deactivates the Outward Controlled Restriction level on the terminal in a guest/patient room. A Check-Out request deactivates any Controlled Restrictions and changes the Controlled Restriction level to Outward Restriction, checks for any messages, clears the wakeup request (if there is one), and deactivates Do Not Disturb (if activated).

If a PMS is not used or if the communications link is down, Check-In and Check-Out can be activated from an attendant console or a front desk terminal with display capability and console permission. Two buttons are required, Check-In and Check-Out. Pressing either button places the display in the respective mode and allows the touch-tone or DTMF buttons to be used for entering data (rather than for placing calls).

The user exits the Check-In or Check-Out mode by pressing another button associated with the display (such as the Normal Mode button). This restores the display and the touch-tone or DTMF buttons to normal operation.

A Check-In/Check-Out request also sends information for the Names Registration feature to the switch. This information includes the guest's name (up to 15 characters), room extension, and Call Coverage path. The switch must be assigned the transparent communications protocol mode for this information to be transferred between the switch and the PMS. If the PMS link is down, and Check-In is done from an attendant console or display-equipped front desk terminal, the guest's name and coverage path information is not automatically updated at Check-In.

If a guest/patient room has both a voice and a data extension, the Check-Out request applies only to the voice extension.

Room Change/Room Swap

These features are provided only through a PMS and must be activated from a PMS terminal. When either Room Change or Room Swap is activated, the PMS sends a message to the system. When Room Change is activated, data pertaining to the old room, including a pending wakeup request, the guest's name (transparent mode), and the guest's Call Coverage path (transparent mode), is moved to the new room. When Room Swap is activated, the data pertaining to the two rooms are swapped. When either feature is activated, if the occupancy status is inconsistent, the system sends an error message to the PMS.

Names Registration

Automatically sends a guest's name and room extension from the PMS to the switch at Check-In, and automatically removes this information at Check-Out. In addition, the guest's call coverage path (for example, voice mail or hotel operator) will also be sent from the PMS to the Switch during Check-In. The guest's call coverage arrangement is set to the administered ‘‘Default Call Coverage Path for Client Rooms’’ at checkout. The Names Registration feature is described in detail elsewhere in this document.

Guest Information Input/Change

Guest Information Input/Change, allows guest information (name or coverage path) to be entered or altered subsequent to the check-in message. Hotel personnel can change this information at the PMS and it is automatically sent to the switch.

The Guest Information Input/Change function is used in those situations when the guest's name associated with an extension must be changed, input of a guest's name must be made after the checking sequence has taken place, or a change in call coverage arrangement must be made.

Considerations

The PMS Interface feature provides a collection of features needed in a hospital or hotel/motel environment: Message Waiting Notification, Controlled Restriction, Housekeeping Status, Check-In/Check-Out, Room Change/Room Swap, Names Registration, and Guest Information Input/Change. All these features, except Room Change/Swap and Check-In/Check-Out, can operate through the system with or without a PMS. When these features are entered through a PMS, the system provides the necessary communications interface.

A customer may elect to use LWC or Integrated Message Center Service for the hospital or hotel/motel staff and Message Waiting Notification for guests/patients. However, if Message Waiting Notification is not used, Integrated Message Center Service can be used for both.

A PMS extension cannot be removed while the PMS link is active. PMS link parameter changes do not take effect until the PMS link is reset.

The normal protocol mode allows extensions of up to four digits in length. The transparent protocol mode allows extensions of up to five digits in length.

When a ‘‘save translations’’ is done on a system with the transparent protocol mode active, station names with Client Room COS are saved as ‘‘blank’’ and coverage paths are saved as the ‘‘Default Coverage Path for Client Rooms.’’

The PMS interface may not work correctly when there are p-extensions with the same leading digit and adjacent lengths. For example 3 and 4 p-extensions with the same leading digit may cause problems. The same applies to 4 and 5, and 5 and 6. If the lengths differ by 2, the PMS interface works correctly. For example, lengths of 3 and 5, or lengths 4 and 6.

A room extension may begin with “0” only if the PMS sends a prefix digit or if the PMS sends a fixed number of digits. Refer to the “Administration” section for system parameters.

Interactions

The following features interact with the PMS Interface feature.

- **Attendant Console or Front Desk Terminal**

  The Controlled Restriction, Check-In/Check-Out, and Message Waiting Notification features can be activated at an attendant console or a front desk terminal with console permission. Also, the attendant console can receive visual notification of the status of the communications link between the system and the PMS.

- **AUDIX Interface**

Message lamps activated by this feature cannot be deactivated by feature buttons or by feature messages from the PMS.

- **Automatic Wakeup**

  An Automatic Wakeup request for a guest/patient room is set or canceled as a result of Room Change/Room Swap or Check-Out.

- **Do Not Disturb**

  A Do Not Disturb request for a guest/patient room is set or canceled as a result of a different Controlled Restriction, Room Change/Room Swap, or Check-Out.

- **Leave Word Calling (LWC)**

  Message lamps activated by this feature cannot be deactivated by Manual Message Waiting feature buttons.

  With a system using the transparent protocol mode, any LWC messages that are present when a check-In request is processed are deleted.

  If Room Change is activated, LWC messages for the old room will *not* be moved to the new room. If Room Swap is activated, LWC messages for the two rooms will *not* be swapped. Therefore, use of the LWC feature, should not be encouraged in guest/patient rooms.

- **Restriction — Controlled**

  Controlled Restriction for a group of users extensions, when activated from the switch, is not conveyed to the PMS. Also, the PMS is not able to remove such restrictions by sending feature messages.

Administration

The following Hospitality System Parameters may be administered:

- **Number of Digits from PMS**—specifies the number of digits the switch expects from the PMS. A blank (the default) indicates that a variable number of digits are expected. If the protocol mode is normal, then the range is 1-4; if the protocol mode is transparent, then the range is 1-5. If the field is blank and the “PMS Sends Prefix” field is no, then the switch will not support extensions with a leading digit of 0 as PMS rooms.
- **“PMS Sends Prefix?” field**—specifies whether the switch should consider the first digit (if it is designated as a prefix digit in the Dial Plan) as a prefix and strip it off before trying to find a PMS record for the extension. Values are y or n. If the field is n and the “Number of Digits from PMS” field is blank, then the switch will not support extensions with a leading digit set to 0 as PMS rooms.
- **Message Waiting Notification**—one of two choices must be administered:
  - Active with no PMS message exchange (act-nopms)

  - — Active with PMS message exchange (act-pms)

- **Controlled Restriction**—one of two choices must be administered:
  - — Active with no PMS message exchange (act-nopms)
  - — Active with PMS message exchange (act-pms)
- **Housekeeping Status Information**—one of two choices must be administered:
  - — Active with no PMS message exchange (act-nopms)
  - — Active with PMS message exchange (act-pms)

  If active is selected, the following additional administration is required:
  - — The number of additional information (Housekeeper Identification) digits from zero to six that can be dialed.
- **Extension Numbers**—assigned to the PMS Journal/Schedule printer and PMS log printer, if used, and to the PMS. Before an extension is assigned, the System Manager should check to make sure that the extension is not already assigned as a Call Detail Recording (CDR) or PSC extension.
- **Seconds Before PMS Link Idle Time-Out**—specifies the number of seconds that the system will wait before it concludes that the PMS is not sending data across the transmission link. Choice is a number of seconds from 5 to 20.
- **Milliseconds Before PMS Link Acknowledgment Time-Out**—specifies the maximum time the system expects acknowledgment from the PMS that a message was received correctly. Choice is a number of milliseconds from 100 to 500.
- **PMS Link Maximum Retransmissions**—specifies the maximum number of times that the system will retransmit a message in response to a negative acknowledgment or send an inquiry for an acknowledgment from the PMS for a message before giving up on the message transmission. Choice is a number from 1 to 5.
- **PMS Link Maximum Retransmission Requests**—specifies the maximum number of times that the system will accept requests from the PMS to resend a reply (acknowledgment or negative acknowledgment) that the system did not receive before giving up on the incoming message. Choice is a number from 1 to 5.
- **PMS Protocol**—specifies the communication protocol mode used between the switch and the PMS. The choices are either normal or transparent.

- **Default Coverage Path for Client Rooms**—specifies the coverage path value that is set for an extension when the switch receives a ‘‘check-out’’ message while in the ‘‘transparent’’ communication protocol mode, or when a save translation is stored for extensions with a Client Room COS. The choice is a number from 1 to 600.
- **Number of Digits in PMS Coverage Path**—located on the second page of the “system-parameter hospitality” form. The coverage path can be set to “3” or “4”. This is used to determine if the PMS adjunct software is using a 3- or 4-digit format for coverage path messaging.

Hardware and Software Requirements

A PMS, if used, can be connected through an MPDM and port on a Digital Line circuit pack or through an ADU and a port on a Data Line circuit pack. A DTDM (with a null modem), 7400A data module, and 7400B data module can also be used for the PMS link. Journal/Schedule and PMS log printers can be used and also require at least an MPDM and a port on a Digital Line circuit pack or an ADU and a port on a Data Line circuit pack. The Journal/Schedule and PMS log printer functionality can be on the same or two distinct printers.

For G3r, there is no means for connecting directly to the switch processor circuit pack, like with G3i. G3r requires, in addition to the data module to connect directly to the adjunct, a TN726B and a TN553 circuit pack combination.

Pull Transfer

Feature Availability

This feature is available with G3i-Global and all Generic V2 and later releases.

Description

NOTE:
G3i-Global, G3V2, and later systems come with this feature turned off by default.

Pull transfer is an enhancement of the standard transfer operation. Standard transfer allows voice terminal users to transfer trunk or internal calls to other voice terminals within the system without attendant assistance. The Pull Transfer feature allows either the calling or the called party (the party to whom the held party will be transferred) to complete the transfer operation.

Analog telephone called parties who wish to pull transfer the party that the controlling party has on hold should momentarily flash the switchhook (or press the Flash key or the Recall button). This completes the transfer of the held party to the called party.

Digital telephone called parties who wish to pull transfer the party that the controlling party has on hold should press the Transfer key. This completes the transfer of the held party to the called party.

Please see the Transfer feature for a description of the regular (push) transfer feature.

Considerations

The pull transfer feature provides a convenient way to connect a party with someone better qualified to handle the call. Attendant assistance is not required and the call does not have to be redialed.

If the attendant is the controlling party, any attempt to complete a Pull Transfer operation by the called party is ignored. Pull Transfer cannot be completed if the Attendant is the called party. A held party can only be transferred by the attendant with Push Transfer.

Pull Transfer can only be completed if the calling party is a station on the same switch as the called party, or if the called party on another switch has been reached via an Italian TGU/TGE tie trunks.

Interactions

- Analog Station Recall Operation and Feature Activation: If the controlling party (with a party on hold) is talking with the called party, and either analog station recall or feature activation is initiated by the called party, the controlling party will not be put into the Hold for Transfer mode but will be Pull Transferred.
- Digital Station Transfer Operation: if the controlling party (with a party on hold) is talking with the called party and the transfer operation is initiated by the called party, the controlling party will not be put into the Hold for Transfer mode but will be Pull Transferred.
- CDR: Checks will be made to ensure that calls are correctly recorded with CDR when a Pull Transfer operation is completed.
- Non-BRI Stations:

  If the “Pull Transfer” field on page 6 of the System Parameters Features form is set to “y,” the following could occur: Station A calls Station B, who answers, then presses the Transfer button and dials Station C. Station C answers and presses the Transfer button. The call from Station A is transferred to Station C.
- BRI Stations:

  If the “Pull Transfer” field on page 6 of the System Parameters Features form is set to “y,” the following could occur: Station A calls Station B, who answers, then presses the Transfer button and dials Station C. Station C answers and presses the Transfer button to “pull” the transfer. Instead, Station C goes off hook on another call appearance as if a new transfer is being originated.

Administration

The “System Feature’’ form is used to enable Pull Transfer.

Hardware and Software Requirements

No special hardware is required to implement the Pull Transfer feature in a standalone configuration. However, in a network environment, the TN497 TGU/TGE tie trunks are the only trunks that support the flash signalling necessary to complete the Pull Transfer operation between switches.

QSIG Global Networking

Feature Availability

This feature is available with Generic 3 Version 2 and later releases.

Description

QSIG Global Networking provides compliance to the International Organization for Standardization (ISO) ISDN-PRI private networking specifications.

QSIG Global Networking Basic Call

QSIG Global Networking Basic Call is available with G3V2 and later releases. The Basic Call complies with ISO standard 11572 for layer 3 to support private network connectivity. This means it provides the ISDN-PRI connection from PBX-to-PBX.

QSIG Global Networking Platform

The QSIG Global Networking Platform, available in G3V3 and later releases, provides a private network protocol for the support of QSIG supplementary services. This platform meets requirements for the ISO Private Network Generic Functional Procedures (Standard 11582) for Call-Related supplementary services. (Supplementary services are services provided by the PBX beyond voice or data connectivity and number transport and display. Examples of supplementary services include Name Identification, Call Forwarding, and Transfer.)

The QSIG Global Networking platform is based on Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI) basic call setup protocol. It supports Call Related supplementary service transport, Abstract Syntax Notation 1 (ASN.1)/Basic Encoding Rules (BER), and Remote Operation Service Element (ROSE) services/protocols.

"QSIG" is the generic name for a family of signaling protocols based on ITU-T (formerly CCITT) Q.931 access protocols across the Q reference point. (The Q reference point or interface is the logical point where signaling is passed between two peer entities in a private network.) The QSIG signaling is used to provide PBX-to-PBX private networking in a multivendor environment. QSIG is being defined by the International Standards Organization (ISO) to be the worldwide standard for PBX-to-PBX private networks. These standards are also supported by the ISDN Private Networking Specification (IPNS) Forum, which is a consortium of PBX vendors working to establish worldwide standards for vendors to follow on a voluntary basis.

The G3V3 release of QSIG Global Networking is a foundation for the QSIG supplementary services. It provides the Name Identification supplementary service, as defined by ISO Standards 13864, 13868, and 11572. G3V4 adds to the QSIG supplementary services by providing for Call Forwarding as defined by ISO Standards 13872 and 13873 and Call Transfer as defined by ISO Standards 13865 and 13869.

DEFINITY system G3V3 and later releases provide QSIG Global Networking in addition to some national protocols and the European Telecommunications Standards Institute (ETSI) protocols for supplementary services. The protocols are mutually exclusive. Users must specify on the “ISDN-PRI Trunk Group” form which protocol they wish to use for ISDN-PRI supplementary services. Therefore, users who require the QSIG Global Networking Platform must administer it on the “ISDN-PRI Trunk Group” form. On this form, the national supplementary service protocols are defined as protocol ‘‘a’’, the QSIG Global Networking protocol is identified as protocol ‘‘b’’, and the ETSI Supplementary Service protocol is identified as protocol “c”.

Supplementary Services

Identification Services

The Identification Services feature allows the PBX to send or receive the calling number, calling name, connected number, and connected name. Furthermore, it allows you to administer ‘‘yes,’’ ‘‘no,’’ or ‘‘restricted’’ for sending identification information. ‘‘Restricted’’ means that DEFINITY system sends the information but sends it ‘‘presentation restricted’’, which indicates to the receiving switch that the information should not be displayed. In the current release, a received restricted name or number is included on the Call Detail Record (CDR). Due to current networking limitations, a received Restricted Calling Party Number is displayed. However, a received restricted Connected Party Number or Calling/Connected Name is *not* displayed.

The Identification Services feature displays up to 15 characters for the calling/connected name and up to 15 digits for the calling/connected number across ISDN-PRI interfaces. When the DEFINITY system acts as a transit PBX (a PBX, other than the originating or terminating PBX, through which a call passes), it usually passes the name information as it receives the information. (A "Transit" PBX is defined as a PBX that routes an incoming call administered for Supplementary Services protocol “b” to a trunk also administered for Supplementary Services protocol ‘‘b.’’) However, in the case of tandemed calls, trunk group administration may override incoming tags, as long as doing so does not lower the restriction on the information. For example, a tandemed call that comes in as restricted remains restricted even if the outgoing trunk is administered for ‘‘presentation restricted.’’ However, nonrestricted data is sent restricted if the trunk group administration is set for ‘‘presentation restricted.’’

The ability to send Calling Name, Connected Name, Connected Number, and Calling Number, or to send them restricted, is administered on the ‘ISDN-PRI

Trunk Group' form. Also, the 'ISDN Public-Unknown-Numbering' and/or 'ISDN Private-Numbering' forms must be administered. The numbering form used to create the number is based on the 'Numbering Format:' field on the 'isdn-pri Trunk Group' form. If ***public*** or ***unknown*** is specified, the 'ISDN Public-Unknown-Numbering' form is used. If ***private*** is specified, the 'ISDN Private-Numbering' form is used.

Call Forwarding (Diversion)

QSIG Call Forwarding is based on the DEFINITY system Call Forwarding All Calls and Call Forward Off Premise features. It extends the feature transparency aspects of Call Forwarding if there is a QSIG link between any of the PBXs involved.

The feature is activated either by dialing a feature access code or by pressing a Call Forwarding button. See the "Call Forwarding All Calls" feature for a detailed description of how to use the Call Forwarding feature.

QSIG Call Forwarding differs from other Call Forwarding features in that additional call information is available to both the caller and the diverted-to terminal above what is provided if the call is forwarded over a link that is not administered for QSIG Supplementary Service protocol "b." The originator (caller) of the call will see "forward" on their display. The diverted-to user will receive information that the call has been forwarded with the redirection symbol "f." Depending upon QSIG Identification Services administration, the originator will see the connected party's name or number followed by "forward." The diverted-to user will see the originator's identification (name or number) and the diverting user's (called party) identification (name or number) followed by "f."

Up to 16 digits can be stored as the forwarded-to number. A total of 20 digits can be sent at the time of forwarding (including the TAC or AAR/ARS access codes).

When a call has already been forwarded three times, it will not be forwarded again but instead will terminate at the final forwarded-to terminal. Remote activation and deactivation are not supported.

Because QSIG Call Forwarding is compatible with the ISO QSIG standards, the DEFINITY system can provide feature transparency for Call Forwarding with any PBX designed to these standards.

Transfer

QSIG Transfer is based on the current DEFINITY system Transfer and Trunk-to-Trunk Transfer features. QSIG Transfer signalling will occur as long as one of the calls involves a QSIG link between the two PBXs. QSIG Transfer conforms to ISO 13869 for QSIG Call Transfer By Join. This means that the transfer occurs through the switch where the transferring user resides and if the transfer involves two trunks, neither will be released after the transferring party is dropped from the call.

The user activating the Transfer feature will see no difference between QSIG Transfer and the standard DEFINITY system Transfer or Trunk-to-Trunk Transfer features. See the "Transfer" feature and "Trunk-to-Trunk Transfer" feature descriptions for basic feature operation.

QSIG Transfer differs from the standard DEFINITY system Transfer feature in that additional call information is available for the connected parties after the transfer completes. However, the information is only sent on QSIG links, that is, if one call is local to the transferring PBX, that user will not receive any new call information.

Depending upon QSIG Identification Services administration, the connected parties' displays will show each other's name and/or number. If the name and number are not available, the display of a connected party is updated with the name of the trunk group involved.

Interactions

Identification Services Interactions

- Distributed Communications Systems (DCS)

  There is no interworking for Calling/Connected Numbers information or for Connected Name information in a DCS network. The Calling Name from a DCS network is received by the QSIG Global Network platform. However, the Calling Name is not sent by the QSIG Global Network platform to a DCS network.

- Non-DCS Non-QSIG Global Networking ISDN-PRI Networks

  Calling/Connected Name and Number information may be sent to and from a non- DCS non-QSIG Global Network platform, depending on the protocol version (country protocol) for the interface.

Call Forwarding (Diversion) Interactions

All interactions that apply to the standard DEFINITY Call Forwarding -- All Calls feature also apply to Call Forwarding with QSIG. See "Call Forwarding All Calls" on page 3-396 for a description of these interactions. The following are additional interactions.

- Distributed Communications Systems (DCS)
- Call Forwarding feature transparency will not exist on calls tandemed between a QSIG (Supplementary Service protocol B) network and a traditional DCS network. However, the basic call will continue.
- Forwarding and Coverage

  If the last coverage point in the coverage path is a number that routes over an ISDN PRI SSB trunk, the call will be treated as a QSIG Diverting call.

- QSIG Identification Services

  Availability of name and/or number display at the originating and diverted-to users depends upon how QSIG Identification Services have been administered for the PBXs involved.

- Terminating Call has Coverage Active

  If a call is forwarded off switch, the terminating switch has call coverage activated, and the criteria are met, the call does not route to the forwarding party's coverage path, it routes to the terminating station's coverage path.

Transfer Interactions

- Distributed Communications Systems (DCS)

  The only DCS transparency that will exist when a call is transferred in a DCS network and passed over a QSIG administered trunk is calling name.

- QSIG Identification Services

  Availability of name and/or number display at the connected parties depends upon how QSIG Identification Services have been administered for the PBXs involved.

Administration

QSIG Global Networking and the Supplementary Services it provides require ISDN-PRI. Therefore, ISDN-PRI must be enabled on the "System-Parameter Customer-Options" form.

On the 'ISDN-PRI Trunk Group' form, the "Supplementary Services Protocol" field must be administered to ''b'' to enable the QSIG supplementary services protocols.

See *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for information about administering QSIG Name and Number Identification, Call Forwarding and Transfer.

Hardware and Software Requirements

QSIG Global Networking does not require any hardware or software beyond what is required for normal ISDN-PRI connectivity. The TN767 circuit board is used for 24 channel applications and the TN464 with suffixes C and greater are used for both 24 channel and 32 channel applications.

Users must have display-equipped voices terminals for displaying the call identification information.

Queue Status Indications

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides indications of queue status for ACD calls based on the number of calls in queue and time in queue. These indications are provided via lamps assigned to the terminals or consoles of split agents or supervisors. In addition, an auxiliary warning lamp can be provided to track queue status based on the number of calls or time in queue. Also, display-equipped voice terminals and consoles can display the time in queue of a split's oldest call and the number of calls in that split's queue.

Two types of Queue Status Indications are provided:

- Number of Queued Calls

  The Number of Queued Calls status indication is based on the total number of calls, excluding Direct Agent Calls (DACs), in queue at a split. DAC calls are not counted here, nor are they included in the count when checking whether a threshold warning should be given.

  The status indication can be provided by an Number of Queued Calls button with associated lamp on a voice terminal or console. Each split is assigned a Number of Queued Calls warning threshold. When this threshold is reached, the lamp associated with the Number of Queued Calls button flashes. If there are calls in the queue, but the threshold is not reached, the lamp lights steadily. If there are no calls in queue, the lamp goes dark.

  In addition to the Number of Queued Calls button(s), the Number of Queued Calls status indication can be provided by an auxiliary queue warning lamp. This lamp can be installed at any location convenient to the split agents. When the Number of Queued Calls warning threshold is reached, the auxiliary queue warning lamp lights steadily.

- Oldest Queued Time

  The Oldest Queued Time status indication is based on the time in queue of the oldest call in a split queue. The status indication can be provided by an Oldest Queued Time (Oldest Queued Time) button with associated lamp on a voice terminal or console. Each split is assigned an Oldest Queued Time warning threshold of 0 to 999 seconds. When the oldest call in queue has been in queue for this length of time, the lamp associated with the Oldest Queued Time button flashes. If there are calls in the queue, but the threshold is not reached, the lamp lights steadily. If there are no calls in queue, the lamp goes dark.

In addition to the Oldest Queued Time button(s), the Oldest Queued Time status indication can be provided by an auxiliary queue warning lamp. This lamp can be installed at any location convenient to the split agents. When the Oldest Queued Time warning threshold is reached, the auxiliary queue warning lamp lights steadily.

Each Number of Queued Calls and Oldest Queued Time button is associated with a specific split. Display-equipped voice terminals and consoles can display queue status information for a split by pressing the Oldest Queued Time or Number of Queued Calls button. The same information is displayed no matter which of the two buttons is pressed. The split name (or extension if name is not assigned), OQT, and Number of Queued Calls are displayed for five seconds unless the displaying terminal or console receives an incoming call or the display is put into another mode. Otherwise, at the end of five seconds, the display returns to its previous condition. If the display has two lines, the queue status information is displayed on the second line.

In addition to providing queue status information for splits, the Queue Status Indications feature can be used to provide status information for attendant groups or other hunt group types (DDC and UCD). The feature works the same with attendant groups as it does with splits, except the button names are different and the display shows ‘‘OPERATOR’’ instead of the split name or extension, and all status information applies to the attendant group queue. The attendant buttons are the (Attendant group’s Queued Time) (AQT) and the Attendant group’s Queued Calls (AQC) buttons.

Considerations

The Queue Status Indications feature allows split agents, split supervisors, and attendants to monitor queue activity. This information is extremely useful in that it allows the agents, supervisors, and attendants to better manage their time.

An NQC, OQT, AQC, and/or AQT button can be assigned to any multifunction voice terminal or console.

Interactions

The following features interact with the Queue Status Indications feature.

- Attendant Display and Voice Terminal Display

  The timer and the queue status information may be displayed at the same time. When this happens, the timer occupies the last eight display positions and the number of queued calls is not displayed. This applies only to one-line displays. With a two-line display, the timer is displayed on the first line and the queue status information is displayed on the second line.

- Move Agent From CMS

  When the CMS is used to move an agent from one split to another, all split associated buttons (including NQC and OQT buttons) become associated with the new split.

Administration

The Queue Status Indications feature is administered by the System Manager. The following items require administration:

- Buttons:
  - — NQC (Number of Queued Calls)
  - — OQT (Oldest Queued Time)
  - — AQT (Attendant Queued Time)
  - — AQC (Attendant Queued Calls)
- NQC warning threshold (per split or attendant group)
- OQT warning threshold (0 to 999 seconds) (per split or attendant group)
- Port number assigned to auxiliary queue warning lamp (per split)

Hardware and Software Requirements

Each auxiliary queue warning lamp requires one port on a TN742, TN746B (A-law), or TN769 Analog Line circuit pack. A beehive-type lamp may be used as an auxiliary queue warning lamp. This lamp is available from the Custom Work Group. The lamp operates on ringing voltage and can be mounted at a location convenient to the group. A ‘‘lamped’’ button on a multiline set is required for the NQC/OQT/AQT or AQC buttons.

No additional software is required.

R2-MFC Signaling

Feature Availability

This feature is available with G3i-Global and G3V2 and later releases.

Description

Multifrequency Compelled (MFC) signaling is a form of number signaling similar to Dual-Tone MultiFrequency (DTMF) in that tones convey the dialed number.

With MFC, the origination and destination switch exchange call progress tones that have specific meanings according to the MFC protocol. R2-MFC is a version of MFC recommended by the CCITT for signaling between a central office (CO) and a PBX over analog or digital CO, DID, or Tie trunks.

With R2-MFC administration, care must be taken to identify the origination switch and the destination switch. The switch where the call is made is the origination switch; the switch where the call is answered is the destination switch. The origination switch creates *forward* signals; the destination switch creates *backward* signals. Forward signals are classified as group I and group II signals. Backward signals are classified as group A and group B signals. Group I and group A signals comprise the basic signaling set, while more elaborate signaling schemes require group II and group B capabilities.

Considerations

The R2-MFC signaling can be used in CO, DID, DIOD, and TIE trunks. Both non-group II signaling and group II signaling are supported on incoming R2-MFC calls. The group II signaling protocol has an extra signal which provides the caller category information. Only group II signaling is supported on outgoing R2-MFC calls. Tandem R2-MFC trunk calls are also supported. That is, an incoming group II R2-MFC call can route to either a group II R2-MFC trunk or a non-group II R2-MFC trunk after all address signals are collected from the incoming R2-MFC trunk.

ANI (Automatic number identification) is supported on both incoming (G3V4 and later releases only) and outgoing R2-MFC calls. When ANI is collected on an incoming R2-MFC call, it is displayed on the terminal and recorded in the Call Detail Recording (CDR) record. On a tandem R2-MFC call, if ANI is requested on an outgoing R2-MFC call, then collected ANI from the incoming trunk is sent out.

Interactions

- ASAI

  ANI collected from incoming R2-MFC calls is not passed to the ASAI interface.

- ARS

  For outgoing R2-MFC calls, the number of digits to be collected from a station and to be outpulsed is based on the ARS translation.

- Call Coverage

  The incoming R2-MFC call may be redirected to coverage according to the coverage criteria assigned for external calls.

- Call Detail Recording (CDR)

  If ANI is collected on incoming R2-MFC calls, it is recorded on the CDR record.

- Call Forwarding

  If the called station has the Call Forwarding feature activated, the incoming R2-MFC call is forwarded to the designated destination.

- Call Pickup

  Incoming R2-MFC calls can be answered by Call Pickup (if Call Pickup is assigned to the station being rung by an incoming R2-MFC call).

- Call Redirection

  Calls will be redirected if any of the following are active: Call Forwarding, Call Coverage, Send All Calls, or Night Service and the call is redirected to a station other than the one indicated by the R2-MFC signals received from the CO. For incoming group II R2-MFC calls, the backward signal B.X sent by the PBX will correspond to the status of the terminating endpoint after redirection.

- Call Vectoring

  ANI collected from incoming R2-MFC calls cannot be used for routing on vector calls.

- Call Waiting

  If this feature is activated on an analog station, and only one call is active at the analog station and an incoming group II R2-MFC call terminates at the analog station, then status of the analog station is treated as idle and the corresponding B.X signal for the ringing is sent to the CO.

- DID No Answer Timer

  The DID NO Answer timer is applied to R2-MFC DID calls.

- Display Feature

  When the display feature is provided, incoming R2-MFC calls will be identified. If ANI is received on incoming R2-MFC calls, then the caller number is displayed. If the display feature is activated on the trunk group, the display is updated after the outgoing R2-MFC call is answered.

- ISDN

  If ANI is collected on incoming R2-MFC calls, it is sent out on ISDN outgoing trunk calls. If an incoming ISDN call is tandemed out to an outgoing R2-MFC trunk and ANI is requested on the outgoing R2-MFC trunk, ANI from the ISDN call is sent out.

- LDN and Multiple LDNs

  R2-MFC DID calls to LDN numbers should be routed to a LDN extension with unique identification of each LDN.

- Malicious Call Trace

  ANI collected from incoming R2-MFC calls is not recorded on malicious call trace records.

- Night Service

  If night service is activated, R2-MFC DID calls route to a designated night extension.

- Night Service No Answer Timer

  If a R2-MFC DID call is routed to the night service destination and the call is not answered, the R2-MFC DID call will be dropped when the Night Service no answer timer is expired.

- Send All Calls

  If this feature is activated, the incoming R2-MFC call is immediately redirected to coverage.

Administration

Group I, group II, group A, and group B signal meanings, and the values for the timers used for MFC signalling, can be administered.

MFC Signaling is initially assigned on a per-system basis by an AT&T service technician. When the R2-MFC Signaling feature is activated, the following items are administered by either the System Manager or the Service Technician:

- System Parameters Multifrequency — The following protocols are specified:

  — Incoming call type (options are group II call type and non-group II call type; non-group II call type is the default)

  — Outgoing call type call (options are group II call type and blank; blank is the default)

The MF DID Intercept Treatment and the request incoming ANI are also selected.

- Trunk groups — The trunk groups used with MF Signaling are specified.

The system manager administers the following MFC Signaling options:

- Trunk Group — Select the trunk groups to use with MF signaling
- System Parameter MultiFreq — Test call extension
- System Parameter MultiFreq — Select MFC Interdigit Timer
- Signal meanings for each of these call types are administrable.

Hardware and Software Requirements

A TN744 v7 or later Call Classifier circuit pack or a TN2182 Tone Clock circuit pack is required as are analog/digital CO/DIOD//DID/TIE trunks.

Recall Signaling

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the user of an analog station to place a call on hold and consult with another party or activate a feature. After consulting with that third party, the user can conference the third party with the original party by another recall signal, or return to the original party by pressing Recall twice or by flashing the switchhook twice. (Digital stations have similar functionality using their ‘‘Hold’’ and ‘‘Conference’’ buttons.)

The recall signaling can be accomplished by pressing the flashhook, using a Ground Key on a Rotary or DTMF station, or by using the Recall Button on a DTMF station.

Considerations

Use of the flashhook for recall signaling may at times place calls on hold when the user of the analog station intended the previous call to be dropped before dialing the third party.

Administration

The length of the time during which the system recognizes the press of the flashhook as recall signaling is administrable. (Administration of Recall Signaling is available on G3i-Global, G3V2, and later releases only.) Administrators may also choose not to administer recall signaling at all. In addition, the recall signaling can be disabled for particular analog stations.

Hardware and Software Requirements

Some earlier hardware versions of the Analog Line Board do not support the administration of the length of the flashhook and disabling the feature.

Recent Change History

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the user to view or print a history report of the most recent administration and maintenance changes. In G3V4 and later releases, the Recent Change History Report also lists each time a user logs in or off the system. The report may be used for diagnostic, information, or security purposes.

The system maintains a log in a software buffer of the most recent administration and maintenance commands. The log is called the transaction log. Commands must be ‘‘data affecting’’ and successfully entered to be saved in the transaction log. The ‘‘data affecting’’ commands are called data commands.

The transaction log can be displayed or printed as a history report by entering the **list history** or **list history print** command at the Management Terminal, or a remote terminal. The report can be generated by any login with Display Admin and Maint Data permissions.

Commands

A command is made up of multiple words, typed on a Management Terminal keyboard, that instruct the system to do a task. The system command structure is made up of an Action, Object, and Qualifier format.

The first command word entered is the **action.** It specifies the operation to be performed (add, display, change, remove, and so on).

The second command word entered is the **object.** It specifies the specific object to be operated on (station, trunk group, hunt group, and so on).

The third command word entered is the **qualifier.** The Qualifier is one or more words or digits used to further identify or complete the Object. Depending on the Object used, a Qualifier may or may not be used. Some commands do not have a qualifier, such as the Dial Plan and Feature Access Codes.

Data Commands

With the exception of login and logout, only those administration and maintenance commands that change the data state associated with any object and qualifier are maintained in the transaction log. The commands that change data are called data commands.

For example, the **change station 3600** command would change the state of the translation data and so would be classified as a data command and entered in the log. However, the command **display station 3600** would not change the state of the translation data and would not be entered in the log.

The following are the commands that are classified as data commands and are saved in the transaction log:

- add, change, remove, duplicate
- backup
- busyout, release
- cancel
- clear
- configure
- enable, disable
- format
- mark
- recycle
- refresh
- restart
- save
- set, reset
- start
- test
- upgrade
- wp (write physical)

The following are the commands that are not classified as data commands and are not saved in the transaction log:

- copy
- download
- get
- list, display, status
- load, restore
- monitor
- rp (read physical)
- upload

Transaction Log and History Report

Associated data is saved in the transaction log for every data command. This data is:

- Date, time
- Port, login
- Action, object, qualifier

A history report of the transaction log data can be displayed or printed by the system administrator by entering the **list history** or **list history print** command. The data commands are displayed or printed in last in, first out order.

An example of a recent change history report is shown in Screen 3-34.

```
                            HISTORY
                     Date of  Loaded Translation: 9:53pm Wed Jul 13, 1994
Date   Time    Port  Login   Actn   Object        Qualifier

07/18  12:34   EPN   scust   logf
07/18  12:23   EPN   cust    cha    dialplan
07/16  09:44   EPN   ncust   rel    station       504
07/16  09:22   EPN   ncust   busy   station       504
07/15  15:25   EPN   cust    cha    station       507
07/15  15:19   NET   cust    cha    system-param  features
07/15  15:18   NET   inads   dup    station       20001 start 30001 call count 8
07/15  15:16   EPN   cust    add    station       507
07/15  15:15   EPN   ncust   logn
07/15  15:09   NET   cust    add    station       505
07/15  15:06   NET   cust    cha    station       504
07/15  15:04   EPN   cust    add    station       504
07/15  15:02   NET   cust    add    station       503
07/15  15:01   NET   cust    add    station       502
07/15  14:56   NET   cust    add    station       501
07/15  14:23   EPN   cust    cha    dialplan
```

Screen 3-34. Transaction Log

The following is a brief description of the report entries:

- Date of Translation Loaded — The time and date that the translation is saved on tape. When a translation is saved on tape, by entering the **save translation** command, the time and date of the save is logged on the tape. Whenever the system is cold started or rebooted, the transaction log is loaded from the tape and the time and date are included on the Recent History Report; for example, ‘‘9:53 pm Wed Jul 13, 1994.’’
- Date — The date the data command was entered; for example, ***07/18***.
- Time — The time the data command was entered; for example, ***12:34.***

- Port — The port, or group of ports, the user was connected to. The users are defined as:

  G3i, G3s, and G3vs Port Types

  — G3-MT

  — INADS

  — EPN

  — NET

  G3r Port Types

  — SYSAM-LCL

  — SYSAM-RMT

  — MAINT

  — SYS-PORT

Table 3-73 shows the way the software correlates the port number to the user that is displayed under Port on the report.

Table 3-73. Software Port Correlations for G3i, s, vs, and r

Port No.	Access Method	Intended Use	Displayed	G3r (Only)
0	MB (EPN)	G3-MT	EPN	SYSAM-LCL
1	MB (EPN)	(not used)	EPN	SYSAM-RMT
2	MB (EPN)	(not used)	EPN	MAINT
3	Netcon		NET	SYS-PORT
4	Netcon		NET	
5	Netcon		NET	
6	Netcon		NET	
7	MTP	SAT	G3-MT	
8	MTP	INADS	INAD	
9	MTP	CDR	CDR	

Table Legend:

EPN — Expansion Port Network
INADS — Initialization and Administration System
MB — Maintenance Board
MTP — Maintenance Tape Processor
Netcon — Network Controller
G3-MT — Management Terminal
CDR — Call Detail Recording
SAT — System Administration Terminal
SYSAM-LCL— System Administration — Local
SYSAM-RMT — System Administration — Remote
MAINT — Maintenance Port
SYS-PORT — System Port

- Login — The system login of the user entering the **data** command; for example, ***cust***.
- Action — The first command word entered; specifies the operation to be performed; for example, **add**, **change**, **remove**. Also indicates when a user has logged into or off of the system.

- Object — The second command word or words entered; specifies the specific object to be acted on; for example **station**, **trunk group**. (If the object is multiple words, only the first word will be displayed. All succeeding words will be treated as qualifiers.)
- Qualifier — The third command word or words entered; one or more words or digits used to further identify or complete the object; for example, ***1120*** (the station number). Some commands do not have a qualifier, such as **dialplan**.

Considerations

Any login IDs that have the “Display Admin” and “Maint Data” fields set to **y** on the “Command Permissions Categories” form have permission to access the list history log.

Interactions

The following features interact with the Recent Change History feature.

- Call Processing

  There are no interactions with any call processing features.

- Other Users

  When a user requests a Recent Change History report, it takes a little time to read all the pages of the report. If during this time other users are entering data commands and altering the transaction log, the oldest entries in the transaction log may have been overwritten by the data commands entered by these other users.

- Set Time Command

  The use of the maintenance **Set Time** command to change the system clock could make the ‘Recent Change History’ report look as if it were not in true last-in, first-out order.

Administration

None.

Hardware and Software Requirements

No additional hardware or software is required.

Recorded Announcement

Feature Availability

This feature is available with all Generic 3 releases. Barge-in announcements are available with G3V3 and later releases. The capability to install multiple Integrated Announcement boards is available with G3V4 and later releases. Also, the capability to install external lineside T1 (DS1) connected announcements is available with G3V4 and later releases.

Description

Provides a recorded announcement to callers under a variety of circumstances. For example, announcements can be used to let callers know that a call cannot be completed as dialed, that their call is in queue, or that all lines are busy. By letting announcements perform these tasks, attendants and other users are free to perform other operations.

Announcements may be required under any of the following circumstances:

- DID calls cannot be completed as dialed
- Incoming Private Network Access calls cannot be completed as dialed
- Calls enter a split or skill (first announcement)
- DDC, UCD or Direct Agent calls have been in queue for an assigned interval
- ACD calls have been in queue for an assigned interval
- A call's destination is a Recorded Announcement extension
- A call routes to a vector that contains an ‘‘announcement,’’ ‘‘disconnect with announcement,’’ or ‘‘collect after announcement,’’ step. When a vector “wait-time” step is programmed with an alternative audio/music source. Or when a “route-to” step routes to an announcement extension.
- An announcement extension is specified as a coverage point
- An announcement is specified as the incoming destination of a trunk group
- VDN of Origin Announcement
- Security Violation Notification
- The Hospitality Automatic Wakeup feature is in use

The following are types of recorded announcements:

- Analog line

  An analog announcement requires an external announcement machine connected by way of an analog line port.

- DS1 (DS1FD, DS1SA or DS1OPS)

  An analog-type announcement connected by way of a DS1 line port. This announcement type is used with Conversant VIS equipped with lineside T1 circuit packs providing standard announcements for the DEFINITY switch.

- Auxiliary Trunk

  An auxiliary trunk announcement requires an external announcement machine connected by way of an auxiliary trunk.

- Integrated

  An Integrated Announcement is stored internally on the switch on an Integrated Announcement board (TN750). Each circuit pack has 16 ports available for playing announcements.

With all announcement types, multiple callers can be connected to the beginning of an announcement. In addition, auxiliary trunk and Integrated Announcements can be administered to allow callers to “barge-in” to announcements. The following paragraphs describe announcement operation when barge-in is not administered. Barge-in operation is described later in this section.

If an announcement port is available when a call arrives, the call is connected to the announcement. If an announcement port is not available, and the announcement is administered with “n” for the queue option, then the caller hears busy or other feedback depending upon how the announcement was accessed.

If an announcement port is not available, and the announcement is administered with a queue, a call enters the announcement queue. When a port becomes available, the switch takes the oldest call waiting in queue for the announcement as well as any other calls in queue for that announcement up to the maximum number that can be connected. All these calls are connected to the beginning of the announcement. The announcement queue length is predetermined for Integrated Announcements and is administered for analog or aux-trunk announcements.

With barge-in announcements, only one port plays the announcement at any time. Calls routed to that announcement connect immediately to the port and hear the announcement from whatever point it was playing when the call connected. Barge-in announcements are generally administered to repeat continually while any callers are connected to the port. In this way, a caller that connects to the middle of an announcement can continue listening until the whole announcement is heard.

Integrated Announcements

An Integrated Announcement is stored on the DEFINITY switch on a TN750 Integrated Announcement circuit pack. Multiple announcements can be stored on each board up to the system capacity. (See Appendix A, ''System Parameters'' for Integrated Announcement board system capacities.)

Each Integrated Announcement board has 16 ports, and can play up to 16 simultaneous announcements. Multiple users can be connected to each of these announcements.

Any announcement stored on a board can play through any port on the board. And, any announcement (not administered for barge-in) can play simultaneously through multiple ports. All 16 ports could play the same announcement at the same time.

Integrated Announcements stored on a TN750(A) are stored at a compression rate of 32 Kbps. With the TN750B and TN750C, Integrated Announcements are stored at one of three compressions rates. The compression rate is administered individually for each announcement extension. In this way, announcements with different compression rates can be stored on the same board. During playback, the switch sets the port to the correct compression rate for the announcement that is being played.

- 64 Kbps compression rate allows for 128 seconds of recorded announcement per board.
- 32 Kbps compression rate allows for 256 seconds of recorded announcement per board. This is the default compression rate.
- 16 Kbps compression rate allows for 512 seconds of recorded announcement per board. The 16 Kbps rate does not provide a high quality recording. It is not recommended for customer announcements but is adequate for VDN of Origin Announcements.

Single Integrated Announcement Boards

With releases prior to G3V4, only one Integrated Announcement board can be installed in any DEFINITY switch. This board can be a TN750, TN750B, or TN750C.

With a TN750 or TN750B when a board is removed from the switch, or when power is lost to the switch, any announcements stored on the board are lost. Therefore, announcements stored on the TN750 and TN750B boards must be backed up to the Mass Storage System (MSS). When a board is inserted, reset, or during system powerup, the switch determines if announcements are present on the board. If announcements are not present, they are automatically restored from the MSS. See the *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for detailed Save and Restore procedures.

The TN750C board has on-board FLASH memory backup. When the board is removed from the switch or power is lost, the announcements are retained on the board. Therefore, the TN750C does not require the Save and Restore procedure. However, the Save and Restore procedure can still be used to copy the contents of a TN750C to another board.

The TN750C substantially reduces the time required for power-up restore and eliminates the need for a manual save of the board contents. However, a manual save can also be performed if additional backup is desired.

Multiple Integrated Announcement Boards

With G3V4 and later releases, multiple Integrated Announcement boards can be installed in G3i and G3r switches. Only one of these boards can be a TN750 or TN750B. Because the switch will Save and Restore only one board, all additional boards must be TN750C. The use of multiple Integrated Announcement boards greatly increases the announcement capacity of the switch.

CAUTION:

Do not copy announcements from a TN750C to a TN750 or TN750B, as it may corrupt the announcement data.

End User Operation

Integrated Announcements can be recorded, played back, or deleted by initiating an announcement session. Announcement sessions can be accessed only by users with console permissions assigned to the Class of Service (COS) for the internal station or Remote Access barrier code. Announcement sessions always use port 0 on the Integrated Announcement board.

To begin an announcement session, the user must dial the administered Feature Access Code (FAC) followed by the announcement extension. If an announcement session is already in progress, or if a **save** or **restore** command is in progress, the caller hears reorder tone (fast busy) and is dropped from the call.

If port 0 is in use, the user hears reorder tone followed by silence. This indicates that the port has been reserved for an announcement session. The user should repeatedly redial the FAC and extension to gain access.

Once an announcement session is accessed, the user can dial “1” to record an announcement, “2” to play an announcement, or “3” to delete the announcement. If the user gains access to an announcement session and hears stutter dial tone, it indicates the board memory is more than 90% full. The user should still begin speaking to record the announcement.

If “1” is dialed, the switch attempts to start a recording session for the extension. If the announcement is protected (designated as protect=y), the user hears intercept tone. If the announcement is currently being played to callers, the user hears reorder tone. If the recording session is started, the user hears record tone and can begin recording.

When the recording is complete, dial # if using a hybrid telephone or hang up if using an analog or digital telephone. When using an analog or digital telephone, ending with a “#” puts a tone in the message. After hanging up, the user may record another announcement session operation for this extension. If the board memory becomes full during recording, the user hears reorder tone and is dropped.

Upon completion of the recording session (drop), a 15 second timer is set. During this time, announcements can be recorded but playback is restricted for the just recorded message. To listen to the recorded announcement, dial the FAC, the extension, and press 2 (playback). This allows the user to listen to the announcement before it is available for general playback.

The announcement is not available by directly dialing the announcement extension or by other call processing features using the announcement extension until the 15 second time period expires. During the 15 seconds, the announcement can be heard by dialing the announcement FAC, the announcement extension, and pressing 2 (playback).

If “2” is dialed, the user hears the announcement recorded for that extension followed by dial tone. Once dial tone is heard, the user can perform another operation, such as recording a new announcement.

If “3” is dialed, the announcement is deleted and the user hears confirmation tone. If the announcement is protected, or is currently being played, it is not deleted and the user hears reorder tone.

Considerations

In G3V3 and later releases, the Recorded Announcements feature enhances the Automatic Wakeup feature by allowing Automatic Wakeup to use the built-in TN750B or later suffix announcement board in place of the more expensive Audichron adjunct.

Interactions

Recorded Announcement is used in conjunction with ACD, Automatic Wakeup, Call Vectoring, VDN of Origin Announcement, Call Prompting, Intercept Treatment, DDC, and UCD features.

Administration

Recorded Announcement is administered by the System Manager. Announcements can be recorded by any user with console permissions. See the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for instructions for administering Recorded Announcements.

Hardware and Software Requirements

Hardware requirements may include any of the following:

- For Integrated Announcements, one or more Integrated Announcement boards (TN750, TN750B, or TN750C)
- For analog announcements, either AT&T (KS-23395 L4) or other external announcement hardware for connection to an analog port on either a TN742 or TN746.
- For auxiliary Trunk announcements, auxiliary trunk hardware (AT&T 13A or 15A, or Cook Electric Announcer), requires 4-wire E&M connection to a port on an auxiliary trunk circuit pack TN763B/C/D

Recorded Telephone Dictation Access

Feature Availability

This feature is available with all Generic 3 releases.

Description

Permits voice terminal users, including Remote Access and incoming tie trunk users, to access dictation equipment.

The dictation equipment is accessed by dialing an access code or extension number (depending on how the feature is administered). After the dictation equipment is accessed, the start/stop function can be voice- or dial-controlled. Other functions such as initial activation and playback are controlled by additional dial codes. The specific dial codes depend on the dictation equipment selected.

Considerations

This feature provides dictation equipment which users can access at their own convenience. Dictation can be recorded, corrected, and played back by the user.

Interactions

The Recorded Telephone Dictation Access feature cannot be used with the following features:

- Automatic Route Selection
- Conference — Attendant
- Conference — Terminal

Administration

Recorded Telephone Dictation Access is administered on a per-system basis by the System Manager. The following items require administration:

- One port on an Analog Line circuit pack (per dictation machine) and an extension number

  or

- One port on an Auxiliary Trunk circuit pack and a trunk access code

Hardware and Software Requirements

Requires telephone dictation machines and, depending on the type of machine, one port on a TN742 or TN746B (A-law) Analog Line circuit pack or one port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) for each machine assigned. No additional software is required.

Redirection On No Answer (RONA)

Feature Availability

Redirection On No Answer (RONA) is available with G3V2 and later releases.

Description

RONA is an ACD hunt group optional feature that redirects an unanswered ringing ACD call after an administered number of rings. The call is redirected back to the split after making the agent unavailable and notifying the Call Center manager. RONA is an enhancement to the Automatic Call Distribution feature and to the related Call Vectoring/Call Prompting features.

When Expert Agent Selection (EAS) is active, RONA is relevant to calls distributed from skills (splits). RONA is also relevant for direct agent calls, which are redirected to the direct agent's coverage path instead of back to a split. RONA can be used for live agent applications that use a manual answering operation as well as for VRU applications and other adjunct applications, such as Home Agent. R3V2 CMS is required to support the specific RONA exception reporting.

RONA applies to split and direct agent ACD calls that are delivered to a manual answering agent extension from an ACD split which are not answered within a reasonable time interval. The call might not be answered in time because the agent has left the agent position while remaining in the available state (for example, the agent fails to enter AUX-Work state or log out). Currently, ''split don't answer'' coverage is relevant for a call that is distributed from a non-vector-controlled split and then times out while ringing an agent before routing to the split coverage path. Vector-controlled splits differ in that they do not provide a ''don't answer'' timeout. Regardless of what type of split is involved, RONA can make the non-answering agent unavailable and then re-route the call. An ACD split call is routed back to the split so that another agent can answer. A direct agent ACD call is routed to the agent's coverage path.

RONA is especially effective in cases where the ''agent'' line is terminated to a Voice Response Unit (VRU) or to another adjunct and the adjunct port experiences a failure. In such a case, if RONA is not in effect, an adjunct port failure is not detected by ACD call processing. As a result, calls are distributed to the failed port, and ringing at that port continues until the caller gives up. On the other hand, if RONA is in effect, it takes that port out of service and provides notification of the failure.

Typically, RONA is used with VRU applications in auto-available split configurations. RONA is used to detect VRU failures and to provide an alternate operation when failures are detected. In addition, Call Vectoring is used to provide maximum flexibility in selecting alternate treatments under VRU failure. RONA can be assigned to a **converse-on** vector command split, which is connected to the Conversant or to equivalent VRU ports. The ‘‘converse’’ hunt group is an ACD type. It is also vector-controlled, and administered as an Auto-Available Split (AAS). Whenever RONA times out on a ringing call that was delivered via the **converse-on** command to a VRU port, the agent port is logged out and the call is redirected back to the **converse-on** split to be distributed to the next available agent or VRU port. In the event of complete VRU failure, all VRU ports are eventually logged out, and vector processing for the **converse-on** command bypasses that step for new calls.

The following example vector shows how backup for a complete VRU failure can be automatically provided in vector processing.

1. wait-time 0 secs hearing ringback
2. converse-on split... (VRU returns the digit ‘‘1’’ as a return code followed by additional digits for the application)
3. collect 1 digits after announcement none
4. goto step 6 if digits = ‘‘1’’
5. goto vector xxx (for backup when the VRU fails)
6. collect 2 digits after announcement none
7. ...

In this example, the application works as expected as long as the VRU returns the digit string, which includes a *return code* of ***1***. In such a case, the condition in Step 4 is satisfied, and a branch is made to Step 6, which provides the normal application processing.

On the other hand, if all VRU ports in an AAS split are removed from service (logged out) by a RONA timeout on each port, the **converse-on** command step (Step 2) is skipped, and no digits are collected by Step 3 (after the 10 second timeout). Accordingly, the condition in Step 4 is not satisfied and vector processing proceeds to Step 5, which branches to vector xxx to provide a live agent or other backup treatment.

Considerations

The following sections discuss the considerations that apply to RONA.

Determining Which Agents Have Timed Out

Notification of RONA timeout occurring for an agent/port on the split is indicated by the RONA (Noans-alrt) lamp lighting steady and also by an exception report to R3V2 and later releases CMS.

The RONA lamp remains lit until it is cleared by operation of the associated button.

The Call Center manager/supervisors can find out which agents had RONA timeout by looking at BCMS or CMS reports. With R3V2 and later releases CMS, the exception report has a listing of all agents that were timed out and made unavailable. Without CMS, BCMS/CMS reports can be used to determine which agents are in AUX-Work or logged-out (with AAS).

With BCMS, the only way to see which agents are logged out (for AAS applications) is to use the SAT to do a list of ‘‘unstaffed agents’’ for the split. With EAS, list ‘‘agent-loginid’’ specifying ‘‘unstaffed’’ and ‘‘AAS" = yes.

With R3 CMS, the real time ‘‘Split Status’’ report can be used to see which agents are in AUX-Work. A custom report is required to see the logged out agents.

See the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for information about logging back in AAS agents that have experienced a RONA timeout.

Using BCMS/CMS Reports With RONA

Whenever RONA is activated on one or more splits in the system, various RONA functions, such as the RONA time out initiated notification, making the agent unavailable, and redirection of the call show up on BCMS/ CMS reports in a number of ways depending on what is being impacted by the RONA operation.

BCMS

In BCMS, the agent state change to AUX for non-Auto-Available Splits (AAS) appears in the ‘‘BCMS Split (AGENT) Status’’ report. In an AAS split, the agent logs out and therefore does not appear in the Split Status report. When the call is requeued, the System Status report shows only the AVG ANSW SPEED time and AVG ABAND TIME time for the requeued call. The historical Split and System reports show both a FLOWOUT (primary split) and FLOWIN (redirected split) for the call that is requeued by RONA, while the VDN report shows only a FLOWOUT.

Direct agent calls are recorded as ACD split calls. However, except the flowout is recorded only if the agent’s coverage path requeues the call to a split.

R3 CMS

On R3 CMS, the agent state change to AUX is shown in the real time ‘‘Split Status Report.’’ If AAS splits are involved, the agent state change logs out the agent. The redirected split call is shown as a FLOW OUT (primary split) and a FLOW IN (redirected split) in the Split historical reports (Daily Split and Interval Split). The requeued call continues to be shown as one call in the real time status reports. Direct agent call redirection is shown in the database items

‘‘DA-OTHERCALLS’’ and ‘‘DA-OTHERTIME.’’ The RONA redirection for the direct agent call adds a count to the OTHER-CALLS item, and the time already spent in queue plus the ringing time is added to the OTHERTIME item. This is the same action taken for any direct agent coverage redirection.

Interactions With Other Ringing Call Timers

There are a number of other features that can time the ringing when an ACD call is delivered to an agent. By design, RONA permits the other timers to continue in conjunction with RONA don't answer timing. Whichever timer expires first applies to the call. RONA is canceled if any of the other timers expire first, except in the case of the coverage timers. When a coverage timer expires, RONA timing continues and is canceled only when the call successfully goes to the coverage point. If RONA times out first, the other timer(s) either may continue timing or be stopped and later be reset or not, depending on where the call is redirected and also on the type and objective of the timer.

The timers that may interact with RONA are listed in the following tables. The first table provides the name for each timer and a short description. The second table lists each timer and indicates what happens to the timer if RONA times out first.

TIMER	DESCRIPTION
Split DA	Split Call Coverage Don't Answer (non-vector controlled)
Covering DA	Covering Point DA - Subsequent Redirection No Answer
Agent DA	Agent DA Coverage (direct agent calls)
NATO	DID/CO Trk No Answer Timeout
WAST	Wait Answer Supervision Timer

TIMER	RONA TIMEOUT	RESTARTED AFTER REDIRECTION?
Split DA	Stopped	If requeued or delivered to another agent
Covering DA	Stopped	Only if redirects to covering point
Agent DA	Stopped	Only if covers to direct agent with coverage
NATO	Continues	N/A
WAST	Stopped	When ringing redirected-to destination or if RONA redirection fails

If RONA notification and redirection is desired, and inasmuch as these timers continue while RONA is timing, the number of rings for a RONA timeout must be set to a shorter period (rings or equivalent time) than the anticipated timeout period of the other timers. A coverage DA timer (including Split DA, Covering DA, and Agent DA timers) starts when the call is first placed in queue, and it continues timing when the call rings the station. Since RONA applies only when the call is ringing the agent, the interval is usually set to two or three rings, while the coverage timer should probably be set to 10 or more rings. Since the queue time is variable and the call could remain in queue until the coverage timer is just about to expire (if it has not already expired while in queue), it is difficult to ensure that RONA will always apply. Accordingly, a coverage timeout period that is greater than the longest expected queue time for the majority of the calls plus three or four rings (the time the call could ring the agent) should be established. RONA then applies to that period when the call rings the agent, and it should be set to fewer than three rings.

The situation with the NATO timer is similar to the coverage timer since it starts before the call is distributed to and rings the agent. The difference is that the NATO timer starts when the call first seizes the incoming trunk and that the timer could be timing even before the call is put in queue during vector processing of the call. Therefore, the NATO timer should be set to a period greater than the longest expected period of time before the call rings the agent (before and after being queued) for the majority of the calls plus three or four rings.

Since the WAST timer starts when the call starts ringing the agent (the same as RONA), just setting the RONA timer to a shorter equivalent time interval (fewer than 10 rings) than the WAST 50 second time period ensures RONA action.

Miscellaneous Considerations

RONA can time out while the agent is actually at the terminal if the agent does not answer soon enough and has perhaps selected another work mode while the call is ringing the agent (pending mode situation). RONA notification, making the agent unavailable, and redirection of the call still takes place. With non-Auto-Available Splits, the agent with a multifunction set knows that the position has been made unavailable because the AUX-Work lamp will light. The agent can then operate the Auto-In or Manual-In button to become available again.

Auto-Available Split (AAS) agent ports are logged out. This requires the agent position to be logged back into the split. AAS should not be used with actual agent positions. Therefore, AAS RONA timeout should be due only to VRU port malfunctions.

Interactions

The following features interact with RONA:

- Abandon Call Search

  If applicable to the trunk type used for the incoming call, Abandon Call Search is reapplied to a call that is requeued by RONA whenever the call is distributed to another agent.

- Agents in Multiple Splits

  For non-auto-available splits, whenever a RONA timeout occurs, the agent state is changed to AUX work (and the proper state change message is sent to CMS) for all splits to which the agent is logged. When the agent returns, the agent is responsible for becoming available in each split, as desired. For auto-available splits, whenever a RONA timeout occurs, the agent is logged out from each of the splits to which the agent is logged. The agent must be logged back into the AAS splits to which the agent extension is assigned.

- Agent Login/Out

  No interaction exists with the agent login because an agent cannot receive an ACD call when the agent is logged out. An agent can log out from a multifunction station set while an ACD call subject to RONA is ringing the set. However, if the agent logs out before a RONA timeout occurs on an ACD call that is ringing the agent, RONA timing is canceled, and RONA redirection and notification treatment takes effect immediately. The notification message (naevent) is sent to R3V2 and later releases CMS before the log out state change is sent. The agent state change to AUX-Work is not performed.

- Agent Position

  RONA applies to ACD calls that ring an agent position.

- Agent Work Modes

  RONA applies to ACD calls that ring an agent available in the Auto-In or Manual-In modes.

- Agent Work States

  For the After Call Work (ACW) state, whenever an agent is using the ACW button while an ACD call is ringing the agent set, the change of state becomes pending. If the agent has a pending change of state to ACW before a RONA timeout occurs on an ACD call that is ringing the agent, RONA timing continues. Upon timeout, the call is redirected and notification is performed, and the agent state change to AUX-Work overrides the pending ACW state. The final state after the call is redirected or otherwise cleared is ‘‘AUX-Work.’’ The RONA notification message must be sent to CMS before the state change message is sent.

  For the AUX-Work state, If an agent is using the AUX-Work button while an ACD call is ringing the agent set, the change to state to AUX-Work becomes pending. With non-AAS splits, since the RONA time-out changes the state to AUX-Work, there is no conflict with the pending AUX-Work state change. With AAS splits, an agent-initiated state change to AUX-Work is denied per existing operation.

- Analog Station Sets

  RONA applies to ACD split calls or direct agent ACD calls ringing at an analog station in an ACD hunt group with manual answering assigned.

- ASAI

  In cases where RONA is assigned to the associated split, RONA applies to vector processed calls that successfully adjunct route to an ACD split or to an agent as a direct agent call.

  RONA can also be assigned to ASAI adjunct-monitored splits, and it can be applied to adjunct-monitored calls. An event report is not sent to the ASAI adjunct whenever a RONA timeout puts an agent into the AUX-Work state because event reports are not sent to the adjunct for agent work state/mode changes for agents in non-adjunct controlled splits. The adjunct determines the state of an agent via use of an agent query (which is part of the value query capability group) whenever the adjunct needs to know the agent state (for example, before sending a direct agent call). Once the call is requeued back to the split, the ASAI adjunct receives an event report for the call queued event if event reporting is active for the domain (VDN or non-vector controlled split).

  A adjunct-monitored split that is not adjunct-controlled can be assigned as an auto-available split. The logout event for an AAS split is sent to the adjunct when RONA timeout logs out the agent.

  RONA cannot be assigned to an adjunct-controlled split. (This combination is blocked via administration). An adjunct-controlled split cannot be auto-available. A split for an ASAI-connected VRU does not need to be auto-available because the ASAI link messaging allows the VRU/adjunct to login/logout and change the work modes/states for the agent lines over the link. The adjunct-controlled split option requires that all state changes and telephone functions for the agents in this split, including the connection of calls to the agents, are controlled by the adjunct over the ASAI link. This option is used in applications where the agent performs all functions through the host-connected data terminal and where the DEFINITY system voice terminal is used only for voice connectivity. Typically, the terminal in these applications is an analog set with auto-answer. With auto-answer analog lines in an adjunct controlled split, the set must remain off-hook to stay logged in. Although they are not blocked, manual answering analog lines are not intended to be used in adjunct-controlled splits. Therefore, calls do not ring analog lines in adjunct-controlled splits, precluding operation with VRUs.

  ASAI CONVERSANT voice services (VRU) applications are configured with non-vector controlled splits by using manual answer operation on analog lines to the CONVERSANT ports. The ASAI link provides event notification for the ACD split, etc., for enhanced services. In addition, the link uses request feature capabilities to log in and log out the ports, as required. (AAS splits are not used for this application because the ASAI link controls the login or logout). RONA can be applied to these splits to detect failure conditions in the same manner as non-ASAI VRU applications. No notification of the RONA time-out state change to AUX-Work is made to the CONVERSANT. In addition, ASAI CONVERSANT does not support a value query, and it cannot query to determine the states of its ports. Restoration of the port(s) after the failure is corrected is needed to be performed manually via appropriate CONVERSANT management screens (for example, log out the port and then log it back in). Complete failure is automatically restored when the CONVERSANT re-initializes.

The next table indicates the ASAI events that the switch sends to the ASAI adjunct for various stages of the RONA call. Also included in the table are the ASAI associations (assuming they are active) for which the events are provided. For the split to have the Notification association active, the split must not be vector-controlled or adjunct-controlled.

STAGE OF CALL	ASAI EVENT	ASAI ASSOCIATIONS
A) RONA timeout	Logout (for AAS)	Domain (Agent) Control
B) Call redirected to split	Call Redirected	Domain (Station) Control (for agent ext call is leaving)
	Queued (only if the call queues)	Domain (Station) Control, (for new agent & for internal originator) Call Control, Notification
C) Call Delivered to Agent	Alerting	Domain (Station) Control, (for new agent & for internal originator) Call Control, Notification
D) Call routed to split's coverage path	Call Redirected followed by existing operation of ASAI Events	Domain (Station) Control (for agent ext call is leaving)
E) Infinite feedback to caller	Call Redirected	Domain (Station) Control (for agent ext call is leaving)
F) Continue Vector Processing	Call Redirected followed by existing ASAI Events	Domain (Station) Control (for agent ext call is leaving redirecting to vdn)
G) Call routed to Direct Agent's coverage path	Call Redirected followed by existing operation of ASAI Events	Domain (Station) Control (for agent ext call is leaving)

- Attendant Return Call

  This is a call that is extended by the attendant but for which the return call timer is not activated if the call is extended to an ACD split or to a VDN and, therefore, does not interact with RONA. A call extended by the attendant to the logical agent ID (direct agent call) and, therefore, does not interact with RONA does not qualify as an Attendant Return Call. The Attendant Return Call Timer is *not* set if an attendant is extending the call to another attendant.

- Attendant—Timed Reminders

  The Attendant Held Call feature applies only to calls that are placed on hold by the attendant and, therefore, it does not interact with RONA. The Attendant No Answer Timeout feature applies only to calls that ring the attendant (via the attendant group). The feature does not apply to hunt groups (or direct agent calls) and therefore, it does not interact with RONA.

- Audio Information Exchange (AUDIX)

  AUDIX controls the log-in/log-out state of the ports, and it has its own means to detect port failures. Since AUDIX is unable to detect the change of state via the switch, the RONA assignment to AUDIX splits is blocked.

- AUDIX Transfer

  RONA applies to a call transferred by AUDIX to an ACD split. A redirected call to AUDIX will not go to split or agent coverage after it is transferred out of AUDIX. If RONA times out on this type of transferred call, the call is to be given ‘‘call-can-not-be-redirected treatment’’ instead of being sent to a split or to the agent’s coverage path.

- Auto-Available Splits

  AAS with RONA is recommended for VRU ACD (non-ASAI adjunct controlled) split applications. AAS can be assigned only to ACD type hunt groups. RONA logs the agent out from all splits to which the agent is logged, and the appropriate messages are sent to CMS. The AAS agent must be manually logged back into each of the splits. When all lines in a vector-controlled AAS split are logged out, the split is considered unavailable, and vector processing subsequently skips the step in the vector for new calls.

  Any remaining calls queued to a split that has been taken out of service may be left at this split. On system reinitialization, all busied-out ports are automatically logged back in to the AAS splits. This is acceptable for RONA since the Call Center administrator was notified due to previous RONA timeouts. New calls cause a RONA timeout if the adjunct or agent still does not answer after the system initializes.

- Automatic Answering with Zip Tone

  If the agent extension is assigned automatic answering, RONA timing is canceled when the call is connected to the agent following the application of a zip tone without ringing the station set.

- BCMS

  After a RONA timeout on a non-AAS split, BCMS shows the agent state as AUX-Work. A RONA timeout on an AAS split shows the agent as logged out.

  The requeuing/forwarding of the redirected call is indicated in BCMS. If the call is not processed by Call Vectoring, the requeuing is recorded as one flow out and one flow in to the split. With Call Vectoring, only a flow out is recorded. Since BCMS does not support exception reporting, a RONA event is not reported by BCMS. Call Centers with only BCMS can use the RONA split lamp indicator for RONA event indication.

- Bridged Call Appearance

  A bridged terminal cannot pick up a ACD split call or a direct agent ACD call. Therefore, bridged call appearances do not interact with RONA.

- Call Coverage Features

  Direct agent non-covered RONA timed-out calls are redirected to the agent's coverage path. A temporary bridged appearance is not maintained for a call directed to an ACD hunt group or VDN, or for a direct agent call. Therefore, a temporary bridged appearance does not interact with RONA.

  Whenever a covered call is redirected to an ACD split or to a direct agent logical ID (as a point in the coverage path) via Expert Agent Selection (EAS), a system parameter coverage—subsequent redirection no answer timer (also called covering point don't answer) is started on the call. Covered calls do not cover to the covering point's coverage path. Instead, they go to the next point in the principal's coverage path. If no other point is available to accept the call, the call remains queued, or it continues to ring the current coverage point. When RONA times out at the coverage point, a variation of the procedures in this paragraph along with the changed condition of the covering point (made unavailable by RONA) must be considered. The interaction requirements of covered calls with RONA are included in the following list.

  — If the coverage point is either a split with RONA or a logical agent ID whose associated split has RONA (as is possible in EAS Direct Agent Calling), RONA timing is started on the call when the call rings an agent without resetting the subsequent redirection no answer timer. If the subsequent redirection no answer timer times out first, the RONA timer is canceled when the call is successfully redirected to the next coverage point of the principal's coverage path according to the existing subsequent redirection operation. If RONA times out first, the subsequent redirection no answer timer is canceled and RONA redirection treatment is applied. This treatment includes notification, making the agent unavailable, and redirecting the call.

  — If the coverage point for a covered call is a direct agent logical agent ID whose associate split has RONA, and if RONA times out first, the call is sent to the next point in the principal's coverage path (if available) and not to the agent's coverage path. The subsequent redirection no answer timer is reset upon redirection to the next coverage point.

  — If the ACD split call to which RONA applied was a previously redirected coverage call (that is, the RONA split was a point in the coverage path), RONA is used to attempt to requeue the call as specified for a non-covered call. However, the call is not designed to go to split coverage or forwarding. The system parameter coverage—subsequent redirection no answer timer is reset if RONA requeues the call to the RONA split. Both the RONA timer and subsequent redirection no answer timer are reapplied.

— If the ACD call to which RONA applied was a previously redirected coverage call (that is, the RONA split was a point in the coverage path), the call is redirected to the next coverage point (if any) of the principal's coverage path if the call cannot be requeued to the RONA split. The subsequent redirection no answer timer is reset upon redirection to the next coverage point.

— If there is no other point in the coverage path, or if the remaining points are unavailable, either the RONA timed-out split covered call that cannot be requeued or the RONA timed out direct agent covered call is given the RONA call-cannot-be-redirected treatment, that is, the caller continue to hear the previously provided feedback.

- Call Detail Recording (CDR)

  Whenever the answering agent is administered to be recorded on the CDR record as the ‘‘called number,’’ the RONA ‘‘redirected-to’’ answering destination is recorded as the final ‘‘called’’ number. CDR can be administered to record either the VDN, the hunt group or the answering agent as the ‘‘called’’ number.

- Call Forwarding All

  If Call Forwarding is activated for the agent before a non-EAS direct agent call (that is, an adjunct direct agent call is made to the agent's physical extension) is redirected due to a RONA timeout, forwarding does not take precedence over agent coverage. Therefore, the RONA timed-out non-EAS direct agent call follows the coverage path. A call that is forwarded via Call Forwarding to a RONA split or logical agent ID whose associated split has RONA is sent to the forwarding extension's coverage path instead of going to the split's coverage path (if the call cannot be requeued) or to the agent's coverage path (for a direct agent call) on RONA time out redirection.

- Call Management System Interface

  CMS interacts with RONA and R3V2 and later releases CMS requires changes to support the RONA exception reporting.

- Call Pickup

  An ACD call being timed for RONA can be picked-up by a member of the agent's pickup group via the Call Pickup feature. RONA is to be canceled upon call pickup.

- Call Vectoring

  — RONA applies to vector-controlled ACD splits when calls are queued via the **queue-to main split**, or **converse-on split**, or **check-backup split** Basic Call Vectoring command. Also, RONA applies to non-vector-controlled or vector-controlled ACD splits when calls are routed to the split via a **route-to** or a **messaging split** Basic Call Vectoring command. Basic Call Vectoring treats an AAS split with all agents logged out as an unavailable split (failed attempt), and it skips the relevant step (in the same manner as it does for a split with all queue slots busy). This requirement applies to vector-controlled splits referenced in a **queue-to main split**, **check-backup split**, or **converse-on split** command. In addition, this requirement is relevant in the case where a (vector-controlled or non-vector-controlled) split is the destination of a **messaging split, adjunct routing, route-to without coverage** command. With an **adjunct routing** or **route-to with coverage** step that routes to a vector-controlled split with all agents logged out, the call is given a busy tone in the same manner as when the call cannot queue to a non-vector controlled split according to the existing operation.

For a call that RONA times out on the last VRU port in an AAS split, the call is not requeued to the ‘‘converse’’ split. Instead, it is processed by the next vector step.

If an AAS split is involved, it is possible that all agent ports experience RONA timeout. Under this condition, it does not make sense to queue the call to the split because the agents (or the VRU) are not available to answer the call. In addition, no one will know about the call because the logged out ports cannot receive ACD calls. Moreover, the logged-out state can be changed only via the administration terminal.

The additional interaction with Call Prompting capabilities is the **route-to digits** interaction described previously in this section.

Three events have been added to the error trace log for calls processed by Call Vectoring. A vector event is generated for a RONA timeout in a split whenever the call is processed by a Call Vectoring **converse-on** vector command. Another vector event is generated when a Call Vectoring step is skipped because all agents of an AAS split are logged out. A third vector event is included for a RONA redirection failure and is also only applicable for the **converse-on** vector command.

When EAS is active, ACD skill groups are used instead of split hunt groups. RONA can be activated for an ACD skill group. After a RONA timeout on the call, the call is put back into the queue at the highest priority for the skill from which the call was distributed (after an attempt is made to deliver the call to another agent) without regard to any previous multiple skills queuing.

RONA timed-out ACD split calls in a vector-controlled split are requeued at the highest priority. As a result, RONA calls are distributed before any other split calls.

Since a call subject to RONA is already removed from vector processing by the time the call is delivered to the agent, the RONA timed-out call is requeued only to the split from which the call was distributed without regard to previous multiple split queuing.

RONA can be applied to both non-vector-controlled splits and vector-controlled splits. Vector-controlled splits cannot be assigned split coverage, split forwarding, or split night service since such functions are provided via Call Vectoring. Therefore, split coverage/forwarding/night service does not apply to RONA-redirected calls that involve vector-controlled splits.

- Calling/Called Number Display

  A RONA-redirected call acts like a direct call to the split. If the call goes to coverage, the destination station display appears as it does for a normal covered call.

  An internal or DCS caller (originator) to an ACD hunt group or VDN sees a display of the hunt group or Vector Directory Number (VDN) name and extension whenever the ACD call is placed to the hunt group or VDN. The display does not change when the call is ringing an agent. A voice terminal-initiated direct agent call (with EAS) displays the agent name and logical ID when the call is ringing the agent's terminal. If the ACD split or direct agent call goes to coverage, the name remains, but the extension or logical ID portion changes to ''cover.'' The operation just described continues when RONA redirects the call.

- Direct Agent Calling

  Direct Agent Calls can be placed by an adjunct using ASAI and/or voice terminals using logical agent IDs with the EAS feature. When the agent is not immediately available, the direct agent call is queued to the agent through an ACD split called the associated split. Without EAS active, the direct agent call queues to the split specified by the adjunct via ASAI. The agent must be logged into that split. With EAS active, the direct agent call (initiated by either an adjunct or by a voice terminal) queues to the first primary skill hunt group to which the agent is logged.

  A RONA timeout applies to direct agent calls that are initiated by an adjunct or voice terminal if the associated split has RONA assigned. RONA timing applies only after the direct agent call is delivered to an available agent with manual answering assigned and while the call is ringing the agent terminal. The sequence of operation (provide notification, make agent unavailable, and redirect call) is the same as specified for ACD split calls. Also, the agent is to be put into AUX-Work or logged out (AAS splits) even if this is the last agent in the split and if there are ACD split calls in queue. Direct agent calls already in queue for the agent are to remain in queue. Such calls are not delivered to an agent that has RONA timed out since the agent has been made unavailable. Don't answer coverage continues to apply to the these calls.

If don't answer (DA) coverage is assigned to the agent extension, the DA timing also to applies to the call. If DA coverage times out first, the RONA timing is canceled when the call successfully routes to coverage, and agent DA coverage is applied per existing operation. If RONA times out first, the coverage DA timing is canceled, and RONA notification is provided, the agent is made unavailable, and call redirection is provided. Typically, the agent coverage don't answer interval is set for a much longer interval than the RONA time-out interval because the coverage DA timing applies initially while the call is in queue and continues when the call rings the agent. The DA interval is normally set for a long enough time to allow the call to remain in queue for a reasonable period of time and to ring the agent two or three times.

After the agent is taken out of service for a RONA time-out on a non-covered direct agent call, and if a coverage path is assigned to the agent, the call is sent down the agent extension don't answer coverage path in the same manner as the coverage don't answer time-out operation.

When EAS is not assigned, the coverage path used is the agent's physical extension coverage path. When EAS is in effect, the coverage path used is the agent's logical extension coverage path for direct agent calls that are initiated by an adjunct or a voice terminal.

If a don't answer coverage path is not assigned to the agent, or if the path is unavailable, the call is given the RONA call-can-not-be-redirected treatment, that is, the caller continues to hear the previously provided feedback.

If the split associated with the direct agent call has split forwarding or night service active, the call is immediately forwarded, precluding RONA timing. If the agent has Call Forwarding or send-all-calls activated, the direct agent call is immediately forwarded (for non-EAS applications only), or it goes to coverage, precluding RONA timing.

The Direct Agent Announcement (DAA) capability is available for Direct Agent calls that are in queue for an agent. RONA does not interact with DAA since the latter capability applies while the Direct Agent call is in-queue for the agent. RONA, on the other hand, applies while the call is ringing the agent. Also, there is no interaction with the forced-first Direct Agent Announcement option since the DAA announcement is played to completion before the call is placed in-queue for the agent.

- DID/CO Trunk No Answer Time Out

  The DID NATO feature begins timing at the original routing of DID/CO trunk calls and continues timing through call forwarding or coverage without being reset. The timer ceases if the call is answered. If the timer expires, the call is routed to the attendant or recorded announcement. NATO, when activated, applies to calls to ACD splits or VDNs.

  If a call with NATO timing is ringing an agent in a split with RONA active, both timers apply to the call, and whichever timer expires first takes precedence and is applied. If the NATO timer expires first, the RONA timer is canceled. If the RONA timer expires first, the NATO timer continues and is canceled when the call is answered. NATO applies to a call that is given the call-can-not-be-requeued treatment. The assigned NATO treatment is given and RONA processing is terminated if the NATO timer expires at any time during or after RONA redirection of the call.

- Direct Department Calling

  RONA applies to DDC ACD type hunt group ACD calls.

- Display - Agent Terminal

  Connected party display does not come into play since the call has not as yet been terminated. (Also, see the explanation for ‘‘Calling/Called Number Display’’ in this section.)

- Delay Announcements

  Split assigned delay announcements for non-vector-controlled splits are applied for requeued RONA calls per normal split redirect call operation.

- Home Agent

  RONA applies to home agent lines that terminate on the Conversant home agent system as a means to detect port failures. The Home Agent split agent lines are not present in AAS and use manual answer operation. Once the RONA notification is made, service can be restored manually on the Conversant system after the failure is corrected.

- Hybrid Stations

  RONA applies to an ACD split or to direct agent ACD calls ringing at a hybrid station in an ACD hunt group with manual answering assigned to the agent extension.

- Inbound Call Management (ICM) with Host

  RONA applies to ICM-managed calls ringing an agent in an ACD split with RONA assigned. (See the "CallVisor Adjunct/Switch Applications Interface (ASAI)" feature interactions for details.)

- Manual Answering with Ringing

  RONA will only apply to ACD split or direct agent ACD calls ringing an agent line/extension with manual answering operation.

- Message Center/Server Service

  RONA can be assigned to Message Center/Server ACD splits and, as a result, it can apply to calls ringing a Message Center/Server split port in an ACD split.

- Most Idle Agent Distribution

  See the "Automatic Call Distribution (ACD)" feature.

- Multi-Function Station Sets

  RONA applies to ACD split or direct agent ACD calls ringing at a multifunction station in an ACD hunt group with manual answering assigned to the agent extension.

- Music-on-Hold Access— Music on Transferred Trunk Call

  When assigned to the system, all trunk callers that are transferred to another destination continue to hear music (or silence, if so administered) instead of ringback while the call rings at the transferred destination. This applies while the transferred call is queued to a split or is ringing an agent. If the trunk call (either as an ACD split call or a direct agent call) is transferred to a split with RONA active, timeout applies to the call in the same manner as to any other call to the split. However, the caller continues to hear the previous feedback instead of ringback when the call either is redirected or ‘‘rings in limbo’’ because the call could not be redirected.

- Music-on-Delay

  Non-vector-controlled split assigned music-on-delay will apply after a delay announcement to requeued RONA timed-out ACD calls per normal operation.

- Off-Premises Station (Terminals)

  RONA applies to Off-Premises Station (OPS) lines in an ACD split.

- Priority Calling

  Priority calling can be activated on ACD split calls or on direct agent calls. With priority calling, ACD split calls or direct agent calls do not go to coverage. Instead, these calls continue to ring the agent.

  If RONA times out a Priority Calling split call or direct agent call, the call is given the call-can-not-be-redirected treatment instead of being sent to the split’s or agent’s coverage path.

- Priority Queuing

  RONA timed-out ACD split calls in a non-vector-controlled split are requeued at the highest priority. As a result, such RONA calls are distributed before any other split calls but after all direct agent calls.

- Queue Size Limiter

  When all assigned queue slots for the ACD split with RONA are taken, RONA timed-out calls cannot be requeued. Instead, the RONA call is sent to split coverage if assigned/active or the call is to be given the RONA call-can-not-be-redirected treatment.

- Queue Status Display/Indications/Warning Indication

  A call that is requeued via RONA is counted in the ‘‘calls queued to the split total’’ as a new call to the queue. The call wait time of the call is reset when the call is requeued. Therefore, the oldest call waiting (OCW) time is not affected by RONA.

- Split/Group Overflow-Forwarding

  If RONA is active for a non-vector-controlled split with split don’t answer (DA) coverage assigned, RONA timing is applied at the same time as the DA timing on ACD calls that ring the agent. When coverage times out, the

RONA timer is canceled only after the call successfully covers to the coverage point. When RONA times out, the coverage timer is canceled. This requirement applies even if RONA cannot redirect the call. Split DA coverage still applies per existing operation when the call is redirected by RONA back to the split while the call is in queue, and it continues when the RONA-redirected call is distributed to an agent. Split busy coverage applies to the RONA-redirected call when there are no queue slots remaining and when no agents are available.

Under these circumstances, the requeue attempt redirects to the split coverage path (unless the call was previously covered). If no coverage point is available, the caller is given the RONA call-can-not-be-redirected treatment, that is, the caller continues to hear the previously provided feedback.

Split don't answer coverage times the call while it is in queue for the assigned number of rings, and it redirects the call to the split coverage path. If no coverage point is available, the call remains in queue.

Split forwarding (interflow) immediately forwards an ACD split call or a direct agent call, precluding RONA operation. If split forwarding is activated for the split before the ACD call is requeued due to a RONA timeout, the call is forwarded.

- Split Night Service

  Hunt group night service is activated by using the hunt group night service feature button. When the feature is activated, all calls for the hunt group are redirected to the assigned night station extension. If night service is activated before the RONA timed-out call is requeued to the split, night service redirection applies. If the night service split has RONA assigned, RONA timing is reapplied to the redirected call.

- Uniform Call Distribution (UCD)

  UCD assigned to a hunt group provides most idle agent call distribution to calls to that hunt group. RONA applies to ACD calls to UCD ACD type hunt groups.

- Voice Response Integration

  RONA can be assigned to **converse-on** command splits. Upon such an assignment, RONA timing applies to calls that are queued and delivered by a **converse-on** command. RONA timing is canceled if the call is delivered to an agent in another split to which a previous queuing attempt was made.

  The **converse-on** split is a vector-controlled split, and it could also be an auto-available split (AAS). In cases where a **converse-on** split is involved, a call that is processed by the **converse-on** command remains in vector processing. Such a call could also be queued to other splits as a result of previous vector commands (for example, **queue-to main split** or **check-backup split**). In addition, the caller continues to hear the previous feedback from vector processing while the call queues to the converse split or while the call rings the converse split agent port (that is, the caller does not hear ringback during ringing if the caller is already hearing music or silence from a previous vector command).

If the **converse-on** split is an AAS, RONA timed out agent lines/ports are logged out. If all agent ports of an AAS split are logged out, new calls are not put into queue but are processed by the next vector step (see the "Auto-Available Split (AAS)" and Vector Controlled Splits feature interactions).

IF RONA times out on a call while ringing a converse split agent port, and if the call cannot be requeued due to a queue full or all AAS agents logged out condition, the call is processed by the next vector step while the caller continues to hear the previous vector feedback.

- Wait-Answer Supervision Timer—Administrable

  The wait-answer supervision timer (WAST) prevents stations on the switch from ringing more than 50 seconds. WAST can be turned on or off for the system. WAST applies to all calls ringing stations except for calls ringing at the attendant console or calls for which the DID or CO trunk no answer timer applies. On calls to hunt groups (completed directly or via Call Vectoring), WAST does not start timing until the call is distributed from the queue and rings an agent. Agent answer cancels the timer. After WAST expires, the call is given intercept treatment.

  If a split/direct agent ACD call is ringing an agent in a split with RONA active in a system that is equipped with wait-answer timing, both timers apply to the call. Whichever timer expires first takes precedence and is applied. If the WAST timer expires first, the RONA timer is canceled. If the RONA timer expires first, the WAST timer is canceled. WAST is reset and applied to a RONA-redirected call that rings the redirected call to a destination. Also, the WAST timer is reset if RONA fails to redirect the call.

Administration

RONA requires administration for the assignment of the feature to an ACD split, the per split don't answer time out interval, and the notification lamps.

A field on the "Hunt Group" form is provided to assign RONA to an ACD split. The split must be the ACD type, and it cannot be an adjunct-controlled or AUDIX split. The assignment of RONA to an adjunct-controlled or AUDIX split (or vice-versa) is blocked. Also, the ACD split with RONA may have one or more of the following split type assignments:

— ASAI-Monitored

— Auto-Available (AAS)

— Skill Hunt Group

— Message Center/Server

— Vector controlled

Once ACD is optioned for the system and the ‘‘ACD?’’ field on the "Hunt Group" form is set to ‘‘y,’’ a new ‘‘Redirect On No Answer (rings)’’ field appears on the "Hunt Group" form. This form is used to administer a don’t answer timeout period (ring cycles) from 1 to 20 rings. A blank value deactivates the feature, and a value between ‘‘1’’ and ‘‘20,’’ inclusive, activates the feature.

Upon submittal of the “Hunt Group” form, and if RONA is administered, end validation checks that the split is not an adjunct-controlled or AUDIX split. If the split is one or the other or both, the form cannot be submitted, and the cursor is placed in the ‘‘Redirect On No Answer’’ field, and the following error message is provided: *Entry must be blank if Controlling Adj not ‘none’ or Message Center ‘audix’*.

The “Basic Station Administration” form is used to assign the RONA notification lamp feature button. This button is administered by using the name **noans-alrt** (that is, no answer alert). The **noans-alrt** button can be assigned to a feature button on any station set, and it is administered on a per split/skill basis. The system wide maximum is one button per split/skill. The auxiliary data for this button consists of the desired split/skill number. A lit noans-alrt button indicates that a RONA timeout has occurred for the split. The split does not need to be administered before the noans-alrt button. If the split is administered, RONA need not be enabled on the split before the noans-alrt button is administered.

Hardware and Software Requirements

ACD hunt groups are required. If vector-controlled split applications are involved, Call Vectoring and/or Call Prompting is/are required. RONA exception reporting/notification to CMS requires that an R3V2 CMS is assigned and operational.

On hybrid, DCP, or BRI multifunction sets, a normal feature button lamp is assigned as a RONA notification lamp for a split. An activated split RONA notification lamp is cleared via operation of the associated feature button. The existing CMS interface is utilized for RONA notification to CMS. VRUs are connected to (typically Auto-Available) ACD splits via (typically analog) line ports with manual answering operation. Other adjuncts, such as Home Agent, are connected via non-Auto-Available ACD splits with analog lines assigned for manual answering operation.

Remote Access (with Security Measures)

Feature Availability

This feature is available with all G3 releases. Logoff Notification and the **status remote-access** command are available with G3V4 and later releases.

Description

Permits authorized callers from remote locations to access the system via the public network and then use its features and services.

Remote Access users can dial into the system using DID, CO, FX, or 800 Service trunks. The Remote Access feature is assigned an extension number, as any voice terminal. When a call is received on a trunk group dedicated to Remote Access, the system routes the call to the assigned extension number. If DID is provided and if the Remote Access number is within the range of numbers that can be accessed by DID, then the Remote Access feature can be accessed through the DID feature.

After access to the feature, the user hears system dial tone, and, for system security, may be required to dial a Barrier code. If a valid Barrier code is dialed, the user may again hear dial tone, and can place local or long-distance calls as allowed. An authorization code may be required to place calls.

The destination of incoming, non-DID, trunk calls can be an attendant or an extension number. The destination is specified on each individual trunk group. When the trunk group is dedicated to Remote Access, the Remote Access extension number is specified. In this case, the user does all dialing. If an attendant is needed on a call, the user dials the public network telephone number assigned, the Barrier code, and **attd** (the attendant access code). To provide attendant-assisted calling, service can be arranged so the attendant handles calls during the day, but Remote Access applies after normal business hours. This is accomplished by setting the trunk group destination as ''attd'' (the attendant), and specifying the Remote Access extension number as the Night Station number. Incoming calls route to the attendant unless the Night button on the primary console is pressed. When Night Service is in effect, incoming calls route to Remote Access.

Setting up an Abbreviated Dialing List on Remote Access Trunks

Users can access the system, group, and enhanced Abbreviated Dialing lists via the remote access trunk. To set up an Abbreviated Dialing list on a remote access trunk, perform the following steps:

1. Set up the Abbreviated Dialing list on the “Console” form.

2. Administer the Abbreviated Dialing list entries.
3. Dial into the system over the remote access trunk.
4. Dial the feature access code followed by the dial code of the list entry.

   NOTE:
   If a barrier code and authorization code are administered, dial them first.

Remote Access Status (G3V4 and later releases)

G3V4 provides the ability to check the status of the remote access feature and barrier codes. The **status remote-access** command displays information that can help in determining why and when use of the remote access feature or a particular barrier code was denied. The display indicates if the remote access feature is:

- Not administered
- Enabled
- Disabled
- Disabled following detection of a security violation

It also gives the date and time that the remote access feature was last modified.

For each barrier code the command displays:

- The date the code was administered, reactivated or modified
- Expiration date
- The number of calls that can be placed with the code
- The number of calls that have been placed using the code
- Whether the code is active or expired
- The date and reason a code expired

For a detailed description of the **status remote-access** command and display, see the *GBCS Products Security Handbook*, 555-025-600.

Security Measures

The Remote Access feature has inherent risks, it can lead to large scale unauthorized long-distance use. For security purposes, a 7-digit barrier code with Remote Access Barrier Code Aging should be used. The Remote Access Barrier Code Aging feature limits the length of time an access code remains valid, and/or the number of times an access code can be used. The ability to define the life span and number of times a barrier code can be used reduces the opportunity for unauthorized use of the Remote Access feature.

Following are security measures for remote access.

Barrier Codes

Remote Access Barrier Code Aging gives the customer the ability to specify, through the administrative interface, both the time interval a code is valid, and/or the number of times a code can be used to access the Remote Access feature. A barrier code automatically expires if an expiration date or number of accesses has exceeded the limits set by the switch administrator. If both a time interval and access limits are administered for a barrier code, the barrier code expires when one of the conditions is satisfied. Expiration dates and access limits must be administered for each barrier code. There are 10 possible barrier codes. Each one can be from 4 to 7 digits long. If there are more than 10 users of the Remote Access feature, the codes have to be shared. When a barrier code is no longer needed it should be removed from the system. Barrier codes should be safeguarded by the user and stored in a secure place by the switch administrator.

If barrier codes are administered, a special answer-back tone is provided that causes a calling modem to leave dial mode. A modem's dialer is sometimes used to gain access (this tone also cancels echo suppressors in the network preventing DTMF tones from breaking dial tone from a switch.) Barrier codes can be used alone or together with authorization codes.

Use the **status remote-access** command (G3V4 and later releases) to view the status of a remote access barrier code.

NOTE:
Barrier codes are *not* tracked by the CDR. Barrier codes are "incoming" access codes. Whereas, authorization codes are primarily "outgoing" access codes.

Authorization Codes

Authorization codes are also used to manage system access. CDR can track use of authorization codes. There are four areas involving authorization code management.

- Assigning codes - Codes should be randomly generated, following no pattern. Assign one each to an individual responsible for protecting the code.
- Periodic change - If codes are changed regularly, they are harder to break.
- Deleting codes - Delete when no longer needed.
- Monitor use - Performed via CDR output analysis.

Maximum length authorization codes should be used to prevent unauthorized access.

Alternate Facility Restriction Levels (AFRL)

For G3rV1 and G3V2 and later releases, consider changing FRLs with the AFRL feature after normal business hours to restrict where calls can be made over your facilities. Take care however, not to restrict callers from summoning emergency services after hours. See the "Alternate Facility Restriction Levels (AFRL)" feature.

Class of Restriction

The COR of an authorization code supersedes that of a barrier code.

Time of Day Routing

This is controlled by the time-of-day entries in COR or by the partition. See the "Time of Day Routing" feature for more details.

Toll Restriction and Analysis

Also controlled by COR. See the "Restriction — Toll" feature description in this document for more details.

Trunk Access Code

This interacts with toll restriction. A customer can translate the switch so that users can make toll calls via ARS but not using a TAC.

Trunk Administration

Trunks can be restricted that are assigned to Remote Access.

Consult the *GBCS Products Security Handbook*, 555-025-600, for additional steps to secure your system and to find out about obtaining information regularly about security developments.

Logoff Notification

Logoff Notification, available with G3V4 and later releases, is recommended for use in cases where Remote Access is enabled, but is not being actively used. Logoff Notification notifies the system administrator at logoff that the Remote Access feature is enabled. It guards against inadvertently leaving Remote Access active and can also alert the system administrator to unauthorized feature activation.

Logoff Notification is administered on a login ID basis.

Considerations

Remote Access provides a caller with access to the system and its features from the public network. An executive can make business calls from home or use the Recorded Telephone Dictation Access feature to dictate a letter. Remote Access may also be used from any extension on the switch. This allows authorized users to access system features from any voice terminal extension.

Ten barrier codes, each with a different COR and COS, can be administered. The barrier codes can be from 4 to 7 digits, but all codes must be the same length. Barrier codes not only provide system security but also define the calling privileges through the administered COR.

Ringback Queuing and Automatic Callback cannot be used on a Remote Access call since the system does not have access to the calling (outside) number.

Any feature requiring recall dial tone (for example, Hold and Transfer) cannot be accessed remotely.

The Remote Access caller must use a touch-tone voice terminal, or equivalent.

After a DTDM's baud rate is changed from 9600 to 1200, the DTDM cannot be accessed by Remote Access until an internal call is made to the DTDM. A Remote Access user attempting the call before an internal call is made receives intercept treatment.

NOTE:
AT&T has designed the Remote Access feature incorporated in this product that, when properly administered by the customer, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require that the customer pay all network charges for traffic. AT&T cannot be responsible for such charges, and does not make any allowance or give any credit for charges that result from unauthorized access.

Interactions

The following features interact with the Remote Access feature.

- Authorization Codes

  When a remote access caller dials the assigned remote access number and establishes a connection to the system, the system may request the caller to dial an authorization code in addition to a barrier code. Dial Tone between the barrier code and authorization code is optional. Calling privileges associated with the COR assigned to the authorization code supersede those assigned to the barrier code.

- Class of Restriction (COR)

  COR restrictions do not block access to the Remote Access feature.

- Night Service — Night Station Service

  The Remote Access extension number can be specified as the Night Station extension number on an incoming, non-DID, trunk group.

- Integrated Services Digital Network Primary Rate Interface (ISDN-PR)I

  A problem may exist when attempting to make a Remote Access call via a public and private trunk. For example, a caller dials 957-5730. The caller receives the beep-beep tones for Remote Access, and dials the barrier code. At this point, the caller may still hear the system dial tone, indicating the digits are *not* being received by PBX#1. To correct the problem, change the UDP table on PBX#2 to give the extension its own RNX of 555.

Administration

Remote Access is administered by the System Manager. The following items require administration:

- Extension number
- Barrier code length (from four to seven digits or blank [no barrier codes])
- Barrier codes
- COR/COS (per Barrier code)
- Authorization Codes
- Whether or not Dial Tone is applied between the barrier code and authorization code
- Barrier Code expiration date and number of access attempts (per barrier code) (G3V3 and later releases)
- Logoff Notification (G3V4 and later releases)

Remote Call Coverage

Feature Availability

This feature is available with Generic 3 Version 2 and later releases.

Description

Provides automatic redirection of certain calls to alternate answering positions in a Remote Call Coverage path for any valid dialed number up to 16 digits that begins with an Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) feature access code (FAC), outgoing trunk dial access code (TAC), or UDP/DCS extension. Once a remote call coverage point is administered for a user, calls to that user are directed to the remote coverage point when the user and any other coverage points are unavailable. The call may have been previously directed to up to two alternate answer points before being redirected to the remote call coverage point. Once a call is directed to the remote call coverage point, it cannot be picked up by the principal user and that user's bridged appearance lamp is extinguished. Calls redirected to a remote call coverage point do not time out and go to a subsequent coverage point.

NOTE:
A vector directory number (VDN) can be administered as the last point in a coverage path.

Hardware Requirements

None.

Report Scheduler and System Printer

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the System Manager to schedule selected administration commands to be printed by an asynchronous printer. Reports are scheduled at 15-minute intervals for any combination of days of the week. Most **list**, **display**, or **test** commands may be scheduled.

Reports may be scheduled, changed, listed, and removed via the system's Management Terminal.

Scheduling (Adding) Reports

The System Manager can schedule a report on the Report Scheduler by using the **schedule** command line option on the Management Terminal (for example, **list configuration all [schedule**]). The system then verifies the command to be scheduled, and the 'Report Scheduler' screen is displayed as shown in Screen 3-35. By setting the "Print Interval" field (described below) to scheduled or deferred, the additional fields appear as shown in the screens below. If the Report Scheduler is full (already has 50 jobs scheduled), the error message Maximum number of reports scheduled; cannot schedule new report is displayed on the G3 Management Terminal.

When using the **schedule** command line option to schedule a report, the Report Scheduler contains the following fields.

- **Job Id:** (display only) Shows the report identification number

  (1 through 50), provided by the system.
- **Command:** (display only) Shows the command to be executed.
- **Print interval:** This field has three options: immediate, scheduled and deferred. Screen 3-35 shows a 'Report Scheduler' screen with the **immediate** option. Screen 3-36 shows a 'Report Scheduler' screen with the **scheduled** option. Screen 3-37 shows a 'Report Scheduler' screen with the **deferred** option.

  The **scheduled** option is used to schedule a report to be printed at a later time.

  The **deferred** option is used to schedule a report to be printed once at a later time.

The **immediate** option is used if the System Manager would like to print the report immediately. If the printer link is not up, the scheduler will attempt to bring up the link and print the report. If the link is already up, the scheduler will mark the report for printing during the current 15-minute time interval. If the printer link cannot be established, the report will be placed at the head of the queue and will be printed the next time the link is established.

If the printer link fails before the report has completed printing, no attempt will be made to print the report when the link is finally established.

The immediate option allows one-shot printing of reports.

- **Days of Week**: If the **scheduled** or **deferred** option of the “Print Interval” field is chosen, the System Manager will be prompted for the days of the week and time of day for the report to be printed. A maximum of one day of the week may be selected for deferred reports.
- **Print time:** Reports may be scheduled at 15-minute intervals within a given hour (0, 15, 30, 45).

```
list configuration all                                  Page 1 of 1

                            REPORT SCHEDULER

    Job Id: 10                  Job Status: none

    Command: list configuration all
    Print Interval: immediate
```

Screen 3-35. Report Scheduler Example 1

See the another example below in Screen 3-36

```
list configuration all                                  Page 1 of 1

                            REPORT SCHEDULER
    Job Id: 10                      Job Status: none

    Command: list configuration all
    Print Interval: scheduled
    Print Time: __21:__15
    Sun:___n Mon:___y Tue:___n Wed:___y Thu:___n Fri:___y Sat:___n
```

Screen 3-36. Report Scheduler Example 2

Screen 3-37 provides a third example.

```
list configuration all                                          Page 1 of 1

                              REPORT SCHEDULER
      Job Id: 10                      Job Status: none
      Command: list configuration all
      Print Interval: deferred
      Print Time: __21:__15
      Sun:___n Mon:___y Tue:___n Wed:___n Thu:___n Fri:___n Sat:___n
```

Screen 3-37. Report Scheduler Example 3

Changing Scheduled Reports

The System Manager may change a scheduled report using the **change report-scheduler** command. When this command is entered, the ‘Report Scheduler’ screen is displayed, as shown in Screen 3-38. This screen is similar to the ‘Report Scheduler’ screen displayed with the **schedule** command line option, but has an additional field. This is the “Job Status” field which shows one of the following:

- **print-next** — Indicates that the report is scheduled to be printed in the current time interval.
- **printing** — Indicates that the report is currently being printed.
- **printed** — Means that the report has been successfully printed.
- **waiting** — Means that the report is not scheduled for any activity during the current 15-minute time interval.

If the Print Interval of a report is changed so that its scheduled time now falls inside the current 15-minute time interval, the report will not be printed in the interval. Instead, the report will be printed during its next scheduled time interval.

If a report is scheduled for a given time period, other than the current 15-minute interval, and has its “Print Interval” field changed from **scheduled** to **immediate**, the report will be printed immediately.

Screen 3-38 provides another example of a ‘Report Scheduler’ screen.

```
change report-scheduler 10                                       Page 1 of 1
                              REPORT SCHEDULER
      Job Id: 10                 Job Status: printed

      Command: list configuration all
      Print Interval: scheduled
      Print Time: __21:__15
      Sun:___n Mon:___y Tue:___n Wed:___y Thu:___n Fri:___y Sat:___n
```

Screen 3-38.   Report Scheduler Example 4

Removing Scheduled Reports

The System Manager may remove a scheduled report using the **remove report-scheduler** command.

If the Job Status of the report is ‘‘print-next,’’ ‘‘printed,’’ or ‘‘waiting’’ (that is, not being printed), it will be removed immediately. If the report is being printed (***printing*** state), not only will the command be removed, but the printer link will be torn down as well. The link will be brought up during the next 15-minute time interval or if an immediate report is scheduled, whichever comes first.

Listing Scheduled Reports

The System Manager may display a list of the scheduled reports on the Management Terminal, or its printer, using the **list report-scheduler** command. A sample list is shown in Screen 3-39. The reports are displayed in the order they will be printed. The id of the user who scheduled the command is also displayed. This field is used to identify who scheduled the command.

Reports that are scheduled for immediate execution will be listed at the top of the queue.

Reports with the same scheduled printing time are displayed according to their order in the report scheduler queue. The first report in the queue will be displayed first.

The “Job Status” field indicates the status of a report. There are four possible values; waiting, print-next, printing, and printed.

The System Manager may send the output of the **list report-scheduler** command to the printer attached to the Management Terminal by using the **print** option.

Establishing the Printer Link

The system will attempt to bring up the link to the Report Scheduler printer at the beginning of each 15-minute time interval, provided there are reports to be printed, or when an immediate report is to be printed. After all reports for which the link was brought up have been printed, the system will tear down the link to preserve system resources.

Screen 3-39 provides another example of a ‘Report Scheduler’ screen.

```
                                                                          Page 1
                              REPORT SCHEDULER
     Job Id       Days (smtwtfs)           Time    User   Status     Type
          Command

       4          immediate                18:53   bcms   printing   immediate
           list bcms split 7 time hh:mm
       2               nynynyn             19:00   bcms   waiting    scheduled
           list bcms split 2 time hh:mm
       7               nynnnnn             19:15   bcms   waiting    deferred
           list bcms system
       23              nyyyyyn             22:45   bcms   waiting    scheduled
           list bcms agent 4000 day        09/15
          Note:  hh:mm is used to indicate field size but is not displayed.
```

Screen 3-39. Report Scheduler Example 5

Considerations

With the Report Scheduler and System Printer, the System Manager can schedule most **list**, **test**, and **display** administration commands to be printed at various times on an asynchronous printer. By scheduling these reports to print automatically at the desired times, the System Manager saves valuable time which can be used to perform other administrative duties.

The System Manager can schedule a maximum of 50 individual reports. The system has a single asynchronous printer connection dedicated for use by the report scheduler. Other printers in the system include those connected to the Management Terminal, the CDR printer, and the Journal Printer. These are not used by the Report Scheduler feature.

Reports scheduled for the same time and day are printed according to their order in the Report Scheduler queue. The first report in the queue will be printed first.

In order to present the least possible impact on system performance, it is recommended that reports be scheduled at off-peak hours and staggered so that they are not all scheduled to be printed at the same time.

Reports that are added to the scheduler queue, and are scheduled to be printed during the current time interval, will not be printed until the next scheduled time.

If a system error is encountered while trying to print a scheduled report, the error will be printed on the report, just as it would be displayed for the same command on the ‘Management Terminal’ screen.

Interactions

There is only one processor board EIA port available for asynchronous output. The port cannot be administered for both CDR and the Report Scheduler System Printer on the “System-Parameter Feature” form. Also, the Report Scheduler System Printer and the Journal Printer used with hospitality features cannot share the same printer.

Administration

The System Manager may schedule, list, change, and remove the desired reports as previously described in this description. Before these procedures can be done, however, the System Manager must supply printer information on Page 4 of the “System-Parameters Features” form by entering the following information:

- **Printer Extension:** ‘‘EIA’’ for the EIA port or a valid data module extension if the EIA port is not to be used.

  The System Manager must specify the printer link by selecting either the EIA port, if available, or a data-module extension. If the data-module extension is chosen, the System Manager must have previously administered the extension using the **add data-module** command.

- **EIA Device Bit Rate:** The speed of the printer (1200, 2400, 4800 or

  9600 baud). Default is 1200.

- **Lines per Page:** The number of printed lines per page (24 to 132). Default is 60.

Hardware and Software Requirements

The asynchronous printer can be connected to the switch using either of the following methods:

- The printer can be connected directly to the EIA port on the switch's processor board. In this case the appropriate cable is required.
- The printer can be connected to the switch with a MPDM or a 7400A data module and a port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law).
- The printer can be connected to the switch with an ADU and a port on a TN726 Data Line circuit pack.

There is a single EIA port in DEFINITY system Generic 3i. There may be contention between the CDR and the Report Scheduler feature for use of this port. If the Report Scheduler feature is using the EIA port and you would like to enable the CDR feature, it is recommended that you disconnect the system printer from the EIA port and use a data module for its connection, freeing the port for use by CDR.

The EIA port on the processor board is not available in a duplicated system. Therefore, the data connection to the Report Scheduler System Printer must interface through a data module. When a processor switch occurs, the link to the printer is dropped and re-established.

An AT&T 475 or AT&T 572, which uses a serial interface, or compatible printer, may be use Printer. A PC may be connected to the system printer port for collection of data; however, a serial interface on the PC must be provided for the connection.

Restriction — Controlled

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows an attendant or voice terminal user with console permission to activate and deactivate the following restrictions for an individual voice terminal or a group of voice terminals:

- Outward — The voice terminal(s) cannot be used for placing calls to the public network. Such call attempts receive intercept tone.
- Total — The voice terminal(s) cannot be used for placing or receiving calls. DID calls are routed to the attendant or a recorded announcement. All other calls receive intercept tone. As an exception, the following call types are allowed: calls to a remote access extension, terminating trunk transmission tests, and Emergency Access to Attendant calls.
- Station-to-Station — The voice terminal cannot receive or place station-to-station calls. Such call attempts receive intercept treatment.
- Termination — The voice terminal cannot receive any calls. Incoming calls are routed to the attendant, are redirected via Call Coverage, or receive intercept treatment.

To activate the desired Controlled Restriction, the attendant or voice terminal user with console permission dials the feature access code for either the extension or the group, followed by either 1 for Outward, 2 for Total, 3 for Termination, or 4 for Station-to-Station, and then dials the voice terminal extension number (Attendant Control — Extension) or the COR for a group of voice terminals (Attendant Control — COR).

Considerations

Controlled Restriction gives the attendant control of outward, total, station-to-station, and termination restriction for voice terminals or groups of voice terminals.

All voice terminals with the same COR are affected by a group restriction. When a call is placed, both the individual and the group restrictions are checked.

Interactions

The following features interact with the Controlled Restriction feature:

- Call Coverage

  Controlled Restrictions are not checked for covering users.

- Call Forwarding

  Controlled Restrictions for the forwarded-to extension are only checked when Call Forwarding All Calls is activated. Once calls are redirected, the forwarded-to extension's restrictions are not checked.

- COR

  Both COR and Controlled Restrictions are checked when a call is authorized.

- UDP

  Calls dialed through the UDP are not restricted by Outward Restriction.

Administration

Controlled Restriction is administered on a per-system basis by the System Manager. The following items require administration:

- Controlled Restriction Activation and Deactivation access codes. Separate access codes are needed for individual (user) and group controlled restriction.
- Type of Intercept Treatment for each type of controlled restriction.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Fully Restricted Service

Feature Availability

This feature is available with G3i-Global, G3rV1, G3V2, and later releases.

Description

Fully Restricted Service is a Class of Restriction (COR) that prevents assigned stations from having access to public network calls. Stations have access to internal calls only. In addition, fully restricted station users cannot use authorization codes to deactivate this feature.

Any calls from the public network to a station with Fully Restricted Service are redirected to intercept treatment or to the attendant. If the call is redirected to the attendant, the attendant's display indicates the call is being redirected because of Fully Restricted Service. The reason-code displayed is FULL.

When the call is redirected to the attendant, the following may be appropriate actions:

- The attendant connected with a CO call may call or intrude on the called station user.
- The attendant cannot extend, conference or bridge the redirected call.
- The attendant can place a CO call on hold and call the station with Fully Restricted Service for consultation.

NOTE:
The name of this feature may imply that there are no circumstances where a station with Fully Restricted Service can access or be accessed by the public network. However, there are circumstances when this can occur. See the ''Interactions'' section below for specific details.

Considerations

There is no limit to the number of COR's that can have the "Fully Restricted Service" field marked ***y.*** If a user with this COR attempts a call and the call is to the public network, then the call is routed to intercept treatment.

Interactions

Because of interactions with other features, Fully Restricted Service should not be assigned to a station under certain conditions (even though the switch allows the assignment). Conditions that prohibit the proper operation of this feature are called prohibitive interactions. Conditions that permit the proper operation of this feature are called allowable interactions. These interactions, beginning with prohibitive interactions, are listed on the following page.

Prohibitive

- Station has Abbreviated Dialing - All the calls automatically dialed from a Privileged List are completed without any restriction checking.
- Station is an Attendant - Each Attendant is assigned a COS and a COR. When the attendant is called using its assigned extension, the COR assigned in the ‘Attendant’ screen is used for authorization checks. Any calls redirected to the attendant group uses the COR assigned in the ‘Console Parameters’ screen for authorization checks. The attendant group and individual attendants should not be assigned a COR with Fully Restricted Service.
- Station has mismatched Bridged Call Appearance - The appearance of a voice terminal’s primary extension number at another voice terminal is called a bridged call appearance. Calls made to the bridged call appearance uses the COR of the station that has the same primary extension as the bridged call appearance for authorization checks. Stations with Fully Restricted Service should not be assigned bridged call appearances without Fully Restricted Service as the restriction can be defeated by using the bridged appearance.
- Station is a call coverage and/or Send All Calls point - The covering station is only checked for Fully Restricted Service when a call is redirected to coverage; no other COR checks are made. Calls from the public network are not redirected to any stations that have Fully Restricted Service. When defining Call Coverage Paths, the coverage points (Point1, Point2, and Point3) can be an extension number, hunt group, coverage answer group, or attendant.
- Station is a Call Forwarding destination for outside calls - Calls from the public network art not forwarded to a station with Fully Restricted Service instead they are terminated at the dialed station.
- Station is a Call Pickup point for outside calls - A station with Fully Restricted Service cannot pickup calls from the public network to another station in the same pickup group.
- Station number is used for Night Service - The following describes the interaction with Night Service. Night Console Service directs all calls for daytime attendant consoles to a night console when Night Service is activated. The COR assigned in the ‘Console Parameters’ screen is used for authorization checks. Multiple Night Service Extension redirects incoming calls from specific trunk groups to assigned Night Service Extension (for that particular trunk group) whenever Night Service is activated. The COR of the trunk group is used for authorization checks. Stations with Fully Restricted Service should not be assigned as the Night Service Extension for public network trunks, to ensure it does not receive calls from the public network. Single Night Service Extension redirects

attendant-seeking calls to a designated extension number whenever Night Service has been activated and the Night Console is either not provided or not available. The COR assigned in the 'Console Parameters' screen is used for authorization checks. A station with Fully Restricted Service should not be assigned as the single night service extension.

- Features that have access to an outgoing trunk with an associated queue. When a station is in an outgoing trunk queue for a public network trunk and Fully Restricted Service is activated for its COR, it is still called back when a trunk becomes available and enabled to place a call over the public network. However, any further attempts to access a public network trunk are redirected to intercept treatment.

Accessible

- Station number represents a loudspeaker paging zone - Each Loudspeaker Paging Access zone is assigned a COR, which is used to check authorization on all calls to Loudspeaker Paging. Code Calling Identification is part of Loudspeaker Paging which allows different Code Calling identifications (chime signals) to be assigned to different Access zones. Each Code Calling Access zone is assigned a COR, which is used to check authorization on all incoming calls to Loudspeaker Paging.
- Station denied for Authorization Codes - An authorization code allows, in certain cases, a terminal user (including an attendant) to dial a four to seven digit code which changes to a new COR that overrides the COR associated with the class of user. The Authorization Code feature can be used with the Automatic Route Selection (ARS), the Automatic Alternate Routing (AAR), certain incoming trunks and/or the Remote Access features. Authorization codes do not override Fully Restricted Service.
- Centralized Attendant Service (CAS)-This involves two switches with calls extended by the attendant on switch A to B. Since COR information is not passed over Release Link Trunks (RLTs), Fully Restricted Service allows all CAS calls. Therefore, this feature allows a public network call to be completed to a station with Fully Restricted Service.
- Distributed Communication System (DCS) - Since COR information is not transparent for DCS, Fully Restricted Service allows all DCS calls. Therefore, this feature can allow a public network call to be completed to a station with Fully Restricted Service.
- Emergency Transfer - All authorization features are bypassed when an Emergency Transfer station is connected to an Emergency Transfer trunk in the Emergency Transfer Mode.
- Hunt Group - The COR assigned to the Hunt Group is checked on calls redirected by either Direct Department Calling (DDC) or Uniform Call Distribution (UCD) of the hunt group feature. Extensions in the hunt group that have Fully Restricted Service can receive calls from the public network (via the hunt group), if the COR of the Hunt Group does not have Fully Restricted Service. Stations with Fully Restricted Service should not be assigned to Hunt Groups without Fully Restricted Service.

- Stations assigned a Personal Central Office Line (PCOL) - If a station with Fully Restricted Service is assigned a PCOL, calls can still be placed to/from the PCOL. Stations with Fully Restricted Service should not be assigned PCOL.
- In a system that associates a barrier code with remote access - If a barrier code is entered on connection to remote access, the barrier code's associated COR is used for authorization checks. If remote access does not require a barrier code (because the barrier code length is blank on the 'Remote Access' screen), then the default barrier code's COR is used ('none' is entered as the only barrier code with an associated COR on the 'Remote Access' screen). Remote Access can require an authorization code instead of or in addition to the barrier code. If an authorization code is required, the authorization code's associated COR overrides the barrier code's COR. CORs assigned to remote access barrier codes and authorization codes, should have the 'CALLING PERMISSION' field marked ***n*** for CORs with Fully Restricted Service, so that remote access cannot call stations with Fully Restricted Service.
- Stations in a non-restricted Terminating Extension Group - A Terminating Extension Group (TEG) is assigned a COR for authorization checks on calls to the TEG extension. TEGs can only receive calls, they cannot originate calls.
- Stations that may use Through Dialing - On Through Dialing calls, the attendant's COR applies (not the COR assigned to the calling party). Transfer and conference checks would still be done for extensions that have Fully Restricted Service active.

Also see Class of Restrictions for detailed feature interactions of restrictions.

Administration

Administrable items are:

- COR identifiers and their associated definitions.
- Each class of user is assigned a COR.

Hardware/Software Requirements

There is no special hardware required for this feature.

Restriction — Miscellaneous Terminal

Feature Availability

This feature is available with all Generic 3 releases.

Description

Restricts callers at specified voice terminals from accessing certain other voice terminals.

Miscellaneous Terminal Restrictions can be used whenever it is undesirable for users at certain voice terminals to access other specific voice terminals.

Considerations

The Miscellaneous Terminal Restriction is controlled by the COR assigned to the calling voice terminal user and to the voice terminal being called. Any COR can be administered to allow or deny access to any other COR. Restricted calls are routed to intercept tone.

Interactions

A voice terminal user with authorization to access an Abbreviated Dialing Privileged Group Number List or Privileged System Number List can place calls to any number on that list. COR assignments are not checked.

Administration

Miscellaneous Terminal Restriction is administered via the COR feature by the System Manager. The only administration required is the permission for each COR to access other CORs.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Miscellaneous Trunk

Feature Availability

This feature is available with all Generic 3 releases.

Description

Restricts users at specified voice terminals from accessing certain trunk groups, such as WATS.

For a detailed description of Miscellaneous Trunk Restrictions, see the Class of Restriction (COR) description.

Considerations

Miscellaneous Trunk Restriction can be used whenever it is necessary to restrict users at certain voice terminals from accessing specific trunk groups.

The Miscellaneous Trunk Restriction is controlled by the COR assigned to the calling voice terminal user and to the trunk group being accessed. Any COR can be administered to allow or deny access to any other COR. Restricted calls are routed to intercept tone.

Interactions

The following features interact with the Restriction — Miscellaneous Trunk feature.

- Abbreviated Dialing

  A voice terminal user with authorization to access an Abbreviated Dialing Privileged Group Number List or a Privileged System Number List can place calls to any number on that list. COR assignments are not checked.

- ARS

  This feature overrides the Miscellaneous Trunk Restriction feature. Permission or denial of ARS calls is determined by the FRL.

Administration

Miscellaneous Trunk Restriction is administered via the COR feature by the System Manager. The only administration required is the permission for each COR to access other CORs.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Toll

Feature Availability

This feature is available with all Generic 3 releases.

Description

Restricts users at specified voice terminals from placing calls that have been designated as toll calls by system administration.

With the Toll Restriction feature, a Toll Analysis table can be administered to assign certain dialed digit strings to a ‘‘toll list.’’ A call containing one of the dialed digit strings assigned to the toll list is designated as being a ‘‘toll call.’’

When a user attempts to dial a toll call, as defined in the Toll Analysis table, he or she may or may not be able to place the call, depending on his or her assigned COR:

a. If the user has a COR with a calling party restriction of ‘‘All-Toll,’’ that user is prevented from making ARS or CO/FX trunk access calls to certain toll areas as defined in the Toll Analysis table, unless the number is on an UCL associated with the user’s COR. (Information on Restricted and Unrestricted Call Lists is provided in the COR feature description elsewhere in this manual.)

b. If the user has a COR with a calling party restriction of ‘‘TAC-Toll,’’ that user is prevented from making trunk access calls to certain toll areas as defined in the Toll Analysis table, unless the number is on an UCL associated with the user’s COR. TAC-Toll restrictions are included with the All-Toll calling party restriction. The only difference is that All-Toll also applies to ARS calls.

If a user is restricted from making an attempted toll call, the user receives intercept treatment.

If the system is connected to a CO that uses a step-by-step switch, all seven digits which are normally dialed for a local call may not be required by the CO to route the call. For example, the central office may only require the last five of the normally dialed seven digits. If all seven digits are dialed, the step-by-step switch uses digit absorption to absorb the unneeded digits. Digit absorption can be provided within the system to emulate the absorption at the central office. This prevents users from bypassing code and toll restriction by dialing unneeded digits. For example, assume that the central office absorbs leading sevens before processing a number and that a toll-restricted user wants to dial a toll code of the form 1-201 plus seven more digits. The user can dial 77-1-201 plus seven more digits. The Toll Restriction feature does not recognize the call as a toll call and the central office routes the call. With digit absorption, the 77 is absorbed by the system before Toll Restriction is used. Thus, the call is denied, as intended. Up to five digit absorption lists can be assigned.

If digit absorption is administered, 0/1 toll restriction is used to determine whether or not the call is allowed. The Toll Analysis table is not checked.

Considerations

Toll Restriction is used whenever it is necessary to restrict users at certain voice terminals from making toll calls. The customer can define what numbers are to be considered toll calls.

One toll list can be assigned per system.

Dialed Digit Strings administered as toll calls in the Toll Analysis table can be a maximum of 18 digits in length.

The Toll Analysis table can have a maximum of 1,000 dialed string entries. Each entry can be assigned to the toll list. However, the 1,000 entries in this table are also used to assign restricted and unrestricted call lists.

Interactions

None.

Administration

Toll Restriction is administered by the System Manager, and is normally enabled for each of your foreign exchanges (FX) and central office (C0) trunks. This means that a toll-restricted user is not able to make a toll call using the trunk group's TAC. If the System Manager wishes to allow toll-restricted users to place toll calls over a particular CO/FX trunk group, the "Toll Restricted" field should be set to ''n.'' The "Toll Restricted" field is ignored if the originator of the call is not toll-restricted.

Toll Restriction is also administered by the System Manager to the following by the COR:

- Attendant consoles as a group
- Incoming tie trunks on a trunk group basis
- Voice terminals on a per-terminal basis
- Data modules on a per-module basis

The Toll Analysis table must be administered by the System Manager to define which calls are toll calls.

The System Manager may also administer digit absorption lists containing absorption treatment of each digit zero through nine.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Voice Terminal — Inward

Feature Availability

This feature is available with all Generic 3 releases.

Description

Restricts callers at specified voice terminals from receiving public network, attendant-originated, and attendant-extended calls. A denied call is routed to intercept tone, a recorded announcement, or the attendant.

Calls can redirect to an inward-restricted voice terminal. The COR of the originally called extension number is the only one checked unless the three-way COR check on Conference and Transfer calls is provided (see Interactions).

Considerations

Inward Restriction is used whenever it is necessary that users at certain voice terminals receive only internal calls from other voice terminals.

Interactions

The following features interact with the Restriction — Voice Terminal — Inward feature:

- Controlled Restriction

  When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

- Night Service

  The Trunk Answer From Any Station and Night Station Service features, if assigned to an inward-restricted voice terminal, override the Inward Restriction.

- Tie Trunk Access

  Incoming dial repeating tie trunk calls can be completed directly to an inward-restricted extension number. However, such calls cannot be extended by an attendant to an inward-restricted voice terminal.

- Transfer

  Incoming trunk calls can be transferred from an unrestricted extension number to an inward-restricted extension number unless the three-way COR check on Conference and Transfer (G3i-global and G3V2 only) is not overridden.

  NOTE:
  With G3i, G3vs/G3s, G3r, and the default operation on G3i-global, G3V2, and later releases, no 3-way COR check is made. In G3i-global, G3V2, and later releases, the 3-way COR check can be optionally allowed for all stations and attendants, or for all stations only. When a 3-way COR check is made, Incoming Trunk calls cannot be transferred to an inward-restricted extension number.

Administration

Inward Restriction is administered by the System Manager to voice terminals by the COR feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Voice Terminal — Manual Terminating Line

Feature Availability

This feature is available with all Generic 3 releases.

Description

Restricts callers at specified voice terminals from receiving calls other than those from an attendant. All other calls are routed to intercept tone, a recorded announcement, or an attendant. The voice terminal user can originate calls and activate features.

Calls can redirect to a voice terminal assigned this feature. The COR of the originally called extension number is the only one checked.

Considerations

Manual Terminating Line Restriction is used whenever it is necessary to have users at certain voice terminals receive only calls from or extended by an attendant.

Interactions

The following features interact with the Restriction — Voice Terminal — Manual Terminating Line feature.

- Controlled Restriction

  When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

- Night Service

  The Trunk Answer From Any Station or Night Station Service feature, if assigned to a restricted voice terminal, overrides Manual Terminating Line Restriction.

Administration

The Manual Terminating Line Restriction feature is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Voice Terminal — Origination

Feature Availability

This feature is available with all Generic 3 releases.

Description

Restricts callers at specified voice terminals from originating calls. Voice terminal users can receive calls.

If a voice terminal user attempts to place a call, intercept tone is received. A voice terminal can, however, activate certain features by dialing the assigned feature access codes.

Considerations

Origination Restriction is used whenever a voice terminal is to be used only for answering incoming calls.

Interactions

When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

Administration

The Origination Restriction feature is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Voice Terminal — Outward

Feature Availability

This feature is available with all Generic 3 releases.

Description

Prevents specified voice terminal users from placing calls to the public network. Calls can be placed to other voice terminal users, to the attendant, and over tie trunks.

Considerations

Outward Restriction is used whenever it is desired that a voice terminal make only internal calls.

The attendant or an unrestricted voice terminal user can extend a call to an outside number for the outward-restricted voice terminal user.

Interactions

When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

Administration

The Outward Restriction feature is administered by the System Manager to voice terminals by the COR feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Voice Terminal — Public

Feature Availability

This feature is available with G3i-Global and all Generic 3 V2 and later releases.

Description

Restricts callers at specified voice terminals from receiving public network calls. A denied call is routed to intercept tone, a recorded announcement, or the attendant.

Calls can redirect to an public-restricted voice terminal. The COR of the originally called extension number is the only one checked.

Considerations

Public Restriction is used whenever it is necessary that users at certain voice terminals receive only internal calls from other voice terminals or calls extended from the attendant.

Interactions

The following features interact with the Restriction — Voice Terminal — Public feature:

- Controlled Restriction

  When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

- Night Service

  The Trunk Answer From Any Station and Night Station Service features, if assigned to a public-restricted voice terminal, override the Public Restriction.

- Tie Trunk Access

  Incoming dial repeating tie trunk calls can be completed directly to an public-restricted extension number.

- Transfer

  Incoming trunk calls can be transferred from an unrestricted extension number to an public-restricted extension number.

Administration

Public Restriction is administered by the System Manager to voice terminals by the COR feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Voice Terminal — Termination

Feature Availability

This feature is available with all Generic 3 releases.

Description

Restricts voice terminal users on specified extension numbers from receiving any calls. The restricted users can, however, originate calls.

Considerations

Termination Restriction is used whenever a voice terminal is to be used only for making calls.

Interactions

When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

Administration

The Termination Restriction feature is administered by the System Manager to voice terminals by the COR.

Hardware and Software Requirements

No additional hardware or software is required.

Ringback Queuing

Feature Availability

This feature is available with all Generic 3 releases.

Description

Places outgoing calls in an ordered queue (first-in, first-out) when all trunks are busy. The voice terminal user is automatically called back when a trunk becomes available. The voice terminal receives a distinctive three-burst alerting signal (Priority Calling) when called back.

If a multiappearance voice terminal user has an idle Automatic Callback button and tries to access an all-trunks-busy trunk group, Ringback Queuing is automatically activated, the lamp associated with the Automatic Callback button lights, and confirmation tone is heard. The multiappearance voice terminal user must also be authorized to make the call, Ringback Queuing must be allowed on the trunk, and the trunk queue must not be full.

Ringback Queuing is automatic for a single-line voice terminal. After dialing is complete, the user hears confirmation tone if the queue is available. No action is required by the voice terminal user. The user hangs up and waits for callback.

The callback call is automatically placed to the terminal when a trunk becomes available. When the user answers the callback call, the original call automatically continues. Redialing is not required.

Queuing can be specified for any non-DCS outgoing only trunk group, or for the outgoing direction of a non-DCS 2-way trunk group.

Considerations

With Ringback Queuing, users do not have to keep trying to access a trunk group when all trunks in the group are busy. This feature provides for the caller of a busy trunk group to automatically be called back when a trunk becomes available.

Queuing can reduce the number of trunks required.

A single-line voice terminal can have only one call waiting at a time; therefore, Ringback Queuing is denied to these voice terminals if a call is already waiting.

A multiappearance voice terminal can have one callback call associated with each Automatic Callback button assigned to the terminal.

A queue request is canceled for the following reasons:

- A trunk is not available within 30 minutes.
- The voice terminal user does not answer the callback call within the administered interval (two to nine ringing cycles).
- The voice terminal is busy when the callback call is attempted.
- The voice terminal user dials the Ringback Queuing cancellation code or presses the Automatic Callback button associated with the queued call.

Incoming tie trunk calls cannot queue on an outgoing trunk group. The system does not know the calling number and cannot originate the callback call.

The system checks the busy/idle status of the trunk group just once (immediately after the trunk access code is dialed). If at this time all of the trunks are busy, the call is put into queue, even if a trunk has become available by the time the caller has completed dialing the number. This occasionally results in the caller being called back immediately after receiving confirmation tone and going on-hook.

At times, a trunk appears to be available, but outgoing calls are still placed in a queue. In this case, a trunk is not free, but is being reserved for a previous Automatic Callback request.

Interactions

If Ringback Queuing is provided, Automatic Callback must also be provided. Automatic Callback is administered through the Class of Service.

Ringback Queuing affects the following features:

- ARS

  If a multiappearance voice terminal user has an Automatic Callback button, makes an ARS call, and all trunks are busy, Ringback Queuing is activated automatically.
- Bridged Call Appearance

  Ringback Queuing is not provided on calls originated from a bridged call appearance.
- Call Coverage

  Callback calls do not redirect even if Send All Calls is activated.
- Call Forwarding All Calls

  Callback calls are not forwarded.
- Call Pickup

  Callback calls cannot be picked up.

- Conference or Transfer

  A single-line voice terminal cannot receive a callback call while it has a call on hold and can have only one active call at a time.

- Internal Automatic Answer (IAA)

  Automatic calls generated by Ringback Queuing are not eligible for IAA.

- Remote Access

  A callback call cannot be made to a Remote Access user because the system does not know the calling number.

Administration

Ringback Queuing is administered by the System Manager. The following items require administration:

- Callback call no-answer time-out (from two to nine ringing cycles)
- Automatic Callback button (per multiappearance voice terminal)
- Ringback Queuing cancellation code
- Queue length (per outgoing trunk group)

Hardware and Software Requirements

No additional hardware or software is required.

Ringer Cutoff

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the user of a multiappearance voice terminal to turn certain audible ringing signals on and off. Visual alerting is not affected by this feature.

When this feature is enabled, only Priority ring (three-burst ringing), Intercom ring, and Manual Signaling rings at the voice terminal. One-burst, two-burst, and redirection notification does not ring. When this feature is disabled, the voice terminal has normal ringing.

Table 3-74 summarizes the call types affected by the activation of Ringer Cutoff. The type of ring depends on whether Redirect Notification is active.

Table 3-74. Summary of Call Types Affected by Ringer Cutoff

		Will The Voice Terminal Ring If Ringer Cutoff Is Active? Redirect Notification =	
Call Type	**Ring Type**	**No**	**Yes**
Voice Terminal to Voice Terminal	1-burst	no	ring ping
Attendant to Voice Terminal	2-burst	no	ring ping
Internal Tie to Voice Terminal	1-burst	no	ring ping
APLT Trunk to Voice Terminal	1-burst	no	ring ping
Trunk to Voice Terminal	2-burst	no	ring ping
Priority Call to Voice Terminal	3-burst	yes	yes
Intercom Call to Voice Terminal	Intercom	yes	yes
Manual Signaling	Manual Signal	yes	yes

NOTE:
If Call Coverage is set to ‘‘Cover All’’ and Ringer Cutoff and Redirect Notification are both active, then there is no audible alerting.

There are occasions when a user does not wish to be disturbed by the arrival of incoming calls, and does not want the call to be immediately redirected to coverage. For example, an executive may have a secretary who has bridged call appearances of his or her extension. If the executive does not wish to be disturbed, this feature can be used to allow the secretary a chance to answer the incoming call before it redirects to coverage. The bridging user (the secretary) is not affected by the executive's activation of Ringer Cutoff.

If a bridging user has Bridged Call Alerting administered for his or her voice terminal (and does not have Ringer Cutoff administered), the bridging user receives ringing for the principal's call.

If a primary extension and all other users with bridged appearances of the primary extension activate Ringer Cutoff, an incoming call silently alerts all of those voice terminals before the call redirects to coverage.

To activate Ringer Cutoff, the user pushes the voice terminal's Ringer-Cutoff button. The associated green status lamp then lights. This lamp remains lighted until the feature is deactivated. If the feature is activated while the voice terminal is ringing with a one-burst ring, two-burst ring, or redirection notification, the ringer is silenced.

To deactivate Ringer Cutoff, the user pushes the active Ringer-Cutoff button on his or her voice terminal. The green status lamp associated with the button then goes dark. If the selected call is in any ringing state, the ringer returns to the proper audible ring. If there are no calls ringing at the voice terminal, the ringer remains silent.

Considerations

The Ringer Cutoff feature allows a user to turn off audible ringing on his or her voice terminal.

Each multiappearance voice terminal user may have one Ringer Cutoff button on his or her voice terminal

Interactions

The following features interact with the Ringer Cutoff feature.

- Automatic Callback

  Even if the Ringer Cutoff feature has been activated, the Automatic Callback call returns to the user's voice terminal with the normal three-burst ring.

- Bridging

  A bridging user is not affected by a primary extension's activation of Ringer Cutoff; nor is the primary extension affected by the activation of Ringer Cutoff by a bridging user.

- Call Forwarding All Calls

  If Ringer Cutoff and Call Forwarding All Calls are active, the user does not receive redirect notification, even if the ‘‘redirection notification’’ is administered for that extension.

- Distinctive Ringing

  Activation of Ringer Cutoff only turns off the ringing of internal and external calls. Intercom ringing, Priority ringing and Manual Signaling are not turned off by the feature.

- Intercom (Automatic and Dial)

  Even if the Ringer Cutoff feature has been activated, Intercom calls still ring the user’s voice terminal.

- Manual Signaling

  Even if the Ringer Cutoff feature has been activated, Manual Signaling still rings the user’s voice terminal.

- Ringback Queuing

  Even if the Ringer Cutoff feature has been activated, the return call for Ringback Queuing still rings the user’s voice terminal.

- Priority Calling

  Even if the Ringer Cutoff feature has been activated, Priority Calls still ring at the user’s voice terminal.

- Send All Calls

  When Ringer Cutoff and Send All Calls are both active, the user does not receive redirect notification, even if the redirection notification is administered for that extension.

- Call Received with Active Call in Progress

  If a new call comes in on a station with an active call on it, and ring cutoff is enabled on the set (with single-ring or continuous-ring enabled for Active Station Ringing), ringing will stop. If the call is still alerting visually and the new call is still active when the ringer cutoff is disabled, ringing will *not* resume.

Administration

The Ringer Cutoff feature is administered on a per-voice terminal basis by the System Manager. The only administration required is a Ringer Cutoff button, which can be assigned to any multiappearance voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

Ringing — Abbreviated and Delayed

Feature Availability

This feature is available with G3V4 and later releases. It provides ringing options identical to those available with DEFINITY switches G2.

Description

What Is Ringing — Abbreviated and Delayed?

Ringing — Abbreviated and Delayed allows the System Administrator to assign one of four ring types to each primary or bridged call appearance on a voice terminal. Ring types fall into two categories:

- Those that alert consistently either by ringing or silently alerting, and
- Those that transition from ringing to silent alerting or vice versa. The transition, known as the Abbreviated/Delayed Transition, can occur automatically based on a system-wide administered interval. Or, it can be manually forced, by pressing an administered Abbreviated Ringing button. The Abbreviated Ringing button will only force the transition for calls that are alerting for the extension number assigned to the button. Calls to other extensions ringing at the terminal will not be affected.

For example, a call appearance can be administered so that a call will initially alert the terminal by ringing. Then, after the Abbreviated/Delayed Transition, it will alert the station silently without ringing. The Abbreviated/Delayed Transition occurs either when the Abbreviated Ringing button is pressed, or optionally when the administered Abbreviated/Delayed Transition Interval is reached.

For Ringing — Abbreviated and Delayed each call appearance must:

- Be assigned a ring type
- Be administered to transition either:
  - — When the Abbreviated/Delayed Transition Interval is reached or when the Abbreviated Ringing Button is pressed
  - — Only when the Abbreviated Ring Button is pressed regardless of the Abbreviated/Delayed Transition Interval.

See the *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for complete instructions for administering Ringing — Abbreviated and Delayed.

What Ring Options Are Available?

One of the following ring types can be assigned to each line button.

Abbreviated Ring

A call will ring the terminal until the automatic or manual Abbreviated/Delayed Transition occurs. The call will then silently alert the terminal.

Delayed Ring

A call will silently alert the terminal until the automatic or manual Abbreviated/Delayed Transition occurs. The call will then ring the terminal.

No Ring

A call will silently alert the terminal regardless of the Abbreviated/Delayed Transition.

Ring

A call will ring the terminal regardless of the Abbreviated/Delayed Transition.

Table 3-75. Ring Types

Ring Type	When Alerting Starts	After Abbreviated/Delayed Transition
Abbreviated Ring	Ringing	Silent Alerting
Delayed Ring	Silent Alerting	Ringing
No Ring	Silent Alerting	Silent Alerting
Ring	Ringing	Ringing

Sample Operation

Consider the following scenario. The system is administered with an Abbreviated/Delayed Transition Interval of two rings and with a coverage Number of Rings interval of four rings. Two terminals on the system are set up as follows.

Terminal 1

Primary Appearance Ext 2375

Automatic Abbreviated/Delayed Ringing is set to yes
Ring Type is set to Ringing

Primary Appearance Ext. 2375

Automatic Abbreviated/Delayed Ringing is set to yes
Ring Type is set to Abbreviated

Bridged Appearance Ext. 2376

Automatic Abbreviated/Delayed Ringing is set to yes
Ring Type is set to Abbreviated

Abbreviated Ringing Button is Assigned to Ext. 2375

Terminal 2

Primary Appearance Ext 2451

Automatic Abbreviated/Delayed Ringing is set to no
Ring Type is set to Abbreviated

Bridged Appearance Ext. 2375, button 1

Ring Type is set to Delayed

Bridged Appearance Ext. 2375, button 2

Ring Type is set to Delayed

Abbreviated Ringing Button is Assigned to Ext. 2451

Figure 3-29. Ringing—Abbreviated and Delayed Sample Terminal Configuration

When a call arrives at extension 2375 at the first call appearance it will begin ringing at Terminal 1 and will silently alert Terminal 2. If it is not answered after 2 rings (the Abbreviated/Delayed Transition Interval) it will continue ringing at Terminal 1 and will begin ringing at Terminal 2. If it is not answered after 4 rings (the coverage Number of Rings interval) it will route to coverage.

When a call arrives at extension 2375 at the second call appearance it will begin ringing at Terminal 1 and will silently alert Terminal 2. If it is not answered after 2 rings (the Abbreviated/Delayed Transition Interval) it will stop ringing at Terminal 1 but will continue to alert silently. It will begin ringing at Terminal 2. If it is not answered after 4 rings (the coverage Number of Rings interval) it will route to coverage.

If a call arrives at extension 2375 at the second call appearance and the Abbreviated ringing button is pressed, the call will immediately begin ringing at Terminal 2 and will silently alert Terminal 1. If a call were to arrive at extension 2376 at Terminal 1 at the same time, it would continue to ring Terminal 1 until the Abbreviated/Delayed Transition Interval was reached despite the fact that the Abbreviated Ringing Button was pressed. The ringing for the call to extension 2376 would not change because the Abbreviated Ringing Button is assigned only to extension 2375 calls.

If a call arrives at extension 2451 at Terminal 2 it will begin ringing. Because Automatic Abbreviated/Delayed is set to no, the call will continue to ring at Terminal 2 until the Abbreviated Ringing Button is pressed.

End-User Operation

As stated, Ringing — Abbreviated and Delayed can be administered per call appearance to transition automatically or manually. Automatic transition is caused by the administered Abbreviated/Delayed Transition Interval and requires no user action.

To cause a manual transition the user presses the Abbreviated Ringing button on their terminal. Pressing the button forces the Abbreviated/Delayed Transition to occur for all calls to the administered extension. Calls to other extensions alerting the terminal are not affected.

Sample Application

In a typical arrangement, for example with an executive and secretary, the primary call appearance is assigned a ring type of abbreviated ringing while the bridged call appearance is assigned a ring type of delayed ringing. In this way, if the primary user does not answer the call after a certain number of rings, the bridged appearance user is audibly alerted to answer the call.

In the same scenario, if the primary user is on a call, other calls to that same extension may alert on other call appearances and disturb the conversation taking place. If an Abbreviated Ringing button is administered for the extension, the primary user can press the button and in so doing force the alerting calls to stop ringing at his or her station and start ringing at the bridged appearance.

The feature also allows certain bridged appearances to audibly alert at a terminal while other bridged appearances only alert visually.

Considerations

If a call appearance is administered for delayed or abbreviated dialing and is not administered for the automatic Abbreviated/Delayed Transition, an Abbreviated Ringing button must be assigned to the call appearance extension on that terminal.

Ringing — Abbreviated and Delayed can be assigned to analog stations whether they are administered as primary or bridged call appearance sets. However, since analog stations cannot visually alert, a user may unexpectedly answer an incoming call while intending to originate a call.

Interactions

With the following features, when the feature attempts to alert a station, the ringing presented to the user is influenced by the ring type translations made for the Ringing — Abbreviated and Delayed feature. Undesirable feature operation may result if these translations are set inappropriately. For example, a ring type of no ringing would result in the station not being alerted audibly.

- Automatic Call Distribution (ACD)/Hunting
- Automatic Callback
- Automatic Circuit Assurance (ACA)
- Busy Verification of Terminals
- Consult
- Night Service (Unattended Console Service)
- Personal Central Office Line (PCOL)
- Priority Calling
- Ringback Queuing
- Security Violation Notification
- Terminating Extension Group

The following features have additional interactions with Ringing — Abbreviated and Delayed.

- Active Station Ringing

  Ringing — Abbreviated and Delayed administration determines when a station is audibly alerted. Active Station Ringing administration determines if the ring is to be permitted, shortened to a single ring cycle, or suppressed. Single Ring timing begins when the call begins to ring at each station that has access to the appearance. In this way, even though a call has been alerting other ringing or abbreviated ringing stations for more than one ring cycle, the delayed ringing stations will have a full cycle of ring in cases where full ringing would normally be applied to an idle station.

- Administration Without Hardware

  It is possible to assign Ringing — Abbreviated and Delayed to stations administered without a specified port.

- Attendant Console

  Ringing — Abbreviated and Delayed cannot be assigned to an attendant console.

- Call Coverage

  When an alerting call redirects to coverage, Call Coverage may cause the call appearance to no longer alert the call appearance. In this case the call would not be affected by the Ringing — Abbreviated and Delayed feature. However, timing continues for the Automatic Transition Interval in case no coverage point is available and the call continues to alert at the station.

  If the coverage Number of Rings interval is shorter than the Automatic Transition Interval, the call will redirect to coverage before ringing a station with a ring type of delayed ringing.

  When a call is immediately redirected to coverage (due to Cover Active, Cover All, Cover Busy or Send All Calls), the Ringing — Abbreviated and Delayed feature has no affect.

- Call Forwarding—Busy/Don't Answer

  When a call is forwarded because it is not answered in the specified time, the call stops alerting the station and so is not affected by the Ringing — Abbreviated and Delayed feature. However, timing continues for the Automatic Transition Interval in case forwarding fails and the call continues to alert at the station.

  If the Call Forward Don't Answer Interval is shorter than the Automatic Transition Interval, the call will redirect to the forwarded-to extension before ringing a station with a ring type of delayed ringing.

- Call Vectoring—Expert Agent Selection—Logical Agents

  Calls routed to a Logical Agent will use the translations for the Ringing — Abbreviated and Delayed feature of the station being used by the agent.

- Data Extension Calls

  Data Extension calls are not affected by the "Ring" values, but will continue to be directed by the "Bridged Call Alerting?" administration.

- OPS/OPX Lines

  The ring type assigned to an Off-Premises Extension is forced to a value of ring.

- Distinctive Alerting (Ringing)

  The Ringing — Abbreviated and Delayed feature will affect all audible alerting using the 1-burst, 2-burst or 3-burst pattern.

- Hospitality Features—Do Not Disturb

  The Do Not Disturb feature takes precedence over the Ringing — Abbreviated and Delayed feature in blocking ringing to the station.

- Integrated Services Digital Network (ISDN)—World Class Core BRI

Several of the protocol variations supported by the World Class BRI feature do not permit the messaging required for control of the station's ringer by Ringing — Abbreviated and Delayed. In this case, ring type is forced to a value of ring.

- Multi-Appearance Preselection and Preference

  The system will automatically select any alerting call on a station whether or not it is ringing if the "Per Button Ring Control?" field is set to n. If the field is set to y, it will select only ringing calls.

- PCOL Calls

  PCOL calls are not affected by the "Ring" values, but will continue to be directed by the "Bridged Call Alerting?" administration.

- Redirection Notification

  If Redirection Notification is enabled, terminals will only receive redirection notification if the alerting button or first call appearance has an assigned ring value of ring or abbreviated ring. Bridged terminals will only receive redirection notification for a forwarded call if the "Per Button Ring Control?" field is set to y and the call appearance has an assigned ring value of ring or abbreviated ring.

- Ringer Cutoff

  Ringer Cutoff takes precedence over Ringing — Abbreviated and Delayed. Pressing the Ringer Cutoff button will turn off audible alerting at all call appearances and bridged appearances on the terminal.

- TEG Calls

  TEG calls are not affected by the "Ring" values, but will continue to be directed by the "Bridged Call Alerting?" administration.

- Temporary Bridged Appearances

  Temporary Bridged Appearances are not affected by the Ringing — Abbreviated and Delayed feature. They always silently alert a station.

- Voice Mail Systems

  Voice mail systems may look for ringing applied to a port to trigger call answer. Undesirable adjunct operation may result if ring type translations are inappropriately set for ports serving these adjuncts.

Administration

See "Ringing — Abbreviated and Delayed" in the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for complete instructions for administering this feature.

Hardware and Software Requirements

Ringing — Abbreviated and Delayed is available for both multi-appearance and analog terminals.

Rotary Dialing

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows rotary dialing voice terminals to be used with a system.

When a number is dialed at a rotary dialing voice terminal, the voice terminal outpulses at a rate of 10 pulses per second. Each digit dialed sends out the corresponding number of pulses. For example, dialing a seven results in seven pulses being sent from the voice terminal. The DEFINITY system Generic 3 software recognizes that the voice terminal is rotary when the user lifts the handset, and expects to receive dial pulses instead of tones.

Considerations

With Rotary Dialing, existing rotary dialing voice terminals can be used in situations where very simple call processing functions are required.

Any functions requiring the * and # symbols cannot be performed on a rotary dialing voice terminal.

Interactions

None.

Administration

Rotary Dialing voice terminals must be administered as a 500 set.

Hardware and Software Requirements

No additional hardware is required.

Security Violation Notification (SVN)

Feature Availability

This feature is available on all versions of the DEFINITY Communications System.

Description

The Security Violation Notification (SVN) feature notifies a designated referral point of a security violation. A designated referral point can be an attendant console, display equipped voice terminal, or voice terminal without display requiring the notification to be by announcement. The SVN feature provides the capacity to disable a valid login ID or remote access following a security violation. The SVN feature also provides an audit trail containing information about each attempt to access the switch. If disabled, the login ID, or remote access feature remains disabled until re-enabled by a login ID with correct permissions.

Sequence of events with the SVN feature enabled and a security violation occurs:

1. SVN parameters are exceeded (the number of invalid attempts permitted in a specified time interval is exceeded).
2. A SVN referral call with announcements (announcement message identifying the violation) is placed to a designated point and the SVN feature provides an audit trail containing information about each attempt to access the switch.
3. The SVN feature disables a valid login ID or remote access barrier code following the security violation.
4. The Login ID or Remote Access remains disabled until re-enabled by an authorized login ID, with the correct permissions.

SVN Enhancements

Referral Call Activation/Deactivation

Referral Call Placement is automatic with G3V3, and later releases. SVN referral calls are placed by the system any time a security threshold violation occurs. To stop placement of referral calls, activate these buttons.

NOTE:
Calls are placed if these buttons are not activated.

- The login security violation feature button ‘‘lsvn-halt.’’
- The remote access security violation feature button ‘‘rsvn-halt.’’

- The authorization code security violation feature button ‘‘asvn-halt.’’

Repeated security violations can result in numerous referral calls being made in a short period of time.

Login ID Kill After ‘‘N’’ Attempts

The Login ID Kill After ‘‘N’’ Attempts feature provides the ability to disable a login ID when a login security violation is detected for a valid login ID (the number of invalid login attempts permitted in a specified time interval is exceeded). If the login security violation parameters are exceeded, the login ID is disabled until re-enabled by a login ID with re-activation permissions. This feature is controlled by an administrable parameter and is optional on a per-login ID basis. The system default value is to disable a login ID if the SVN feature is active and a security violation occurs. Any attempt to access the switch using a login ID that has been disabled by the Security Violation Notification (SVN) feature fails, even if the correct login ID and password is entered. If the login ID is disabled while logged in on another session, once that session is terminated any subsequent attempt to log in using that login ID is prohibited. SVN referral calls are placed by the system each time a login security violation occurs. A disabled login ID remains disabled until it is re-enabled by a login ID with reactivation permissions.

A major alarm is logged whenever a security violation is detected involving an AT&T services login ID and that login ID has been disabled as a result of the security violation. AT&T is responsible for retiring the alarm.

Remote Access Kill After ‘‘N’’ Attempts

The Remote Access Kill After ‘‘N’’ Attempts SVN feature provides the ability to disable the Remote Access feature when a remote access barrier code security violation is detected (the number of invalid Remote Access attempts permitted in a specified time interval is exceeded), and the “Disable Following a Security Violation” field is enabled. Any attempt to use the Remote Access feature once it has been disabled fails, even if a correct barrier code or barrier code/authorization code combination is given. SVN referral calls are placed by the system any time a Remote Access security violation occurs. The Remote Access feature remains disabled until re-enabled by a login ID with re-activation permissions.

Authorization Code Security Violation

The Authorization Code Security Violation feature generates a referral call upon detection of a violation. An audit trail containing relevant information about each attempt is registered.

SVN Referral Call With Announcement

The SVN Referral Call with Announcement option has the capacity to provide a recorded message identifying the type of violation accompanying the SVN referral call. Using Call Forwarding, Call Coverage, or Call Vector Time-of-Day Routing (to route to an extension or a number off the switch), SVN referral calls with announcements can terminate to a point on or off the switch.

Use of other means to route SVN referral calls to alternate destinations are not supported at this time. An attempt to use an alternate method to route SVN referral calls may result in a failure to receive the call or to hear the announcement.

Monitor Security Violations Report

The security violations reports provide current status information for invalid Login or Remote Access (barrier code) or Authorization Code attempts. The data displayed by these reports is updated every 30 seconds. A total of 16 entries is maintained for each type of violation. The oldest information is overwritten by the new entries at each 30-second update. When a login is added or removed, the Security Measurements reports are not updated until the next hourly poll, or a **clear measurements security-violations** command is entered. The security violations report is divided into three distinct reports:

- Login Violations
- Remote Access Barrier Code Violations
- Authorizations Code Violations

To access Monitor Security Violations reports, enter the command interface command **monitor security-violations <report name>**. The report names are ‘‘login,’’ remote-access,’’ and ‘‘authorization-code.’’

The following fields are displayed on the Login Violation report:

- **Date:** The date the attempt occurred.
- **Time:** The time the attempt occurred.
- **Login:** The login string entered as part of the login violation attempt. An invalid password may cause a security violation. If a valid login ID causes a security violation by entering an incorrect password, the Login Violation report displays the valid login ID.
- **Port:** The port on which the failed login session was attempted. The following abbreviations are used for G3i:
  - — **MGR1:** The dedicated management terminal connection (the EIA connection to the maintenance board).
  - — **NET-N:** A network controller dialup port (1-4).
  - — **EPN:** The EPN maintenance EIA port.
  - — **INADS:** The INADS port (Initialization and Administration System).

— **EIA:** Other EIA ports.

The following abbreviations are used for G3r:

— **SYSAM-LCL:** Local administration to Manager 1.

— **SYSAM-RMT:** Dial up port on SYSAM board, typically used by services for remote maintenance, and used by the switch to call out with alarm information.

— **SYS-PORT:** System ports accessed through TDM bus.

— **MAINT:** Ports on expansion port network maintenance boards, used as a local connection for onsite maintenance.

- **Ext:** The extension assigned to the network controller board that the failed login session was attempted on. This field is present only in the case where the System Manager's SAT is administered through a network controller port.

The following fields are displayed on the Remote Access Violations report:

- **Date:** The date that the attempt occurred.
- **Time:** The time that the attempt occurred.
- **TG No:** The trunk group number associated with the trunk where the remote access attempt terminated.
- **Mbr:** The trunk group member number associated with the trunk where the remote access attempt terminated.
- **Ext:** The extension used to interface with the Remote Access feature.
- **Barrier Code:** The incorrect barrier code that resulted in the invalid attempt.

The following fields are displayed on the Authorization Code Violations report:

- **Date:** The date that the attempt occurred.
- **Time:** The time that the attempt occurred.
- **Originator:** The type of resource originating the call that generated the invalid authorization code access attempt. Originator types include:

  — Station.

  — Trunk (other than a trunk assigned to a remote access trunk group).

  — Remote Access (when the invalid authorization code is associated with an attempt to invoke the Remote Access feature).

  — Attendant.
- **Auth Code:** The invalid authorization code entered.
- **TG No:** The trunk group number associated with the trunk where the attempt terminated. It appears only when an authorization code is used to access a trunk.

- **Mbr:** The trunk group member number associated with the trunk where the attempt terminated. It appears only when an authorization code is used to access a trunk.
- **Barrier Code:** The incorrect barrier code that resulted in the invalid attempt. It appears only when an authorization code is entered to invoke Remote Access.
- **Ext:** The extension associated with the station or attendant originating the call. It appears only when authorization code is entered from the station or attendant console.

Administering SVN System Parameters

To activate SVN system features, three sets of system level parameters must be administered:

- SVN Login Violation Notification
- SVN Remote Access Violation Notification
- SVN Authorization Code Violation Notification

Refer to the SVN Referral Call With Announcement section on page 3-1077.

Administering the SVN Login Security Violation Notification Feature

To administer the login component of the SVN feature, enter the `change system-parameters security` command.

To administer system parameters for the login component of the SVN feature violation notification:

1. Access the "System Parameter Security" form by entering the `change system-parameters security` command from the command line interface.
2. When the "SVN Login Violation Notification Enabled" field is enabled, the following fields appear on the "Security-Related System Parameters" form:
   - Originating Extension

     Requires the entry of an unassigned extension local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for login security violations.

     The originating extension initiates the referral call in the event of a login security violation. It also sends the appropriate alerting message or display to the referral destination.

- Referral Destination

  This field requires an entry of an extension, assigned to a station, attendant console, or vector directory number (VDN) that receives the referral call when a security violation occurs. If a VDN is assigned the Time-of-Day routing capability, Call Vectoring may be used to route the referral call to different destinations based on the time of day or the day of the week. The referral destination must be equipped with a display module unless the Announcement Extension is assigned. Administration of the Announcement Extension is also required if the referral destination is a VDN.

- Login Threshold

  This field requires an entry of the minimum number of login attempts that are permitted before a referral call is made. The value assigned to this field, in conjunction with the “Time Interval” field, determines whether a security violation has occurred. The system default is 5.

- Time Interval

  This field requires the entry of the time interval in which a login security violation must occur. The range for the time interval is one minute to eight hours (0:01 to 7:59), and is entered in the form ‘‘xx:xx.’’ For example, if you want the time interval to be one minute, you enter 0:01. If you want the time interval to be seven and one-half hours, you enter 7:30. The system default is 0:03.

- Announcement Extension

  This field requires an entry of a extension that is assigned to an SVN announcement.

3. Administer an ‘‘lsvn-halt’’ button on any station/attendant console (maximum 1 per system). The SVN button location can be determined by entering the command **display svn-button location.**

Enable/Disable a Login ID

The “Disable a Login ID Following a Security Violation” field on the “Login Administration” form is used to set the SVN parameters for a single login. When set to ‘‘y’’ (yes) this SVN disables the specified login ID (system default is y). When set to ‘‘n’’ the SVN feature does not disable the specified login ID if a security violation is detected for the login ID. The “Disable a Login ID Following a Security Violation” field is dynamic and only appears on the “Login Administration” form when the login component of the SVN feature is enabled.

To enable a login ID that has been disabled by a security violation, or disabled manually with the command **disable login** the user must:

1. Log in to the switch using a login ID with the correct permissions.
2. Enter the command **enable login <login ID>.**

To disable a login ID, the user must:

1. Log in to the switch using a login ID with the correct permissions.
2. Enter the command **disable login <login ID>**.

List the Status of a Login ID

To list the status of a login ID, the user must:

1. Log in to the switch using a login ID with the correct permissions.
2. Enter the command **list login.**

You see a display indicating the status of the specified login ID. A login ID status can be listed as:

- Login ID status equals **disabled** indicating that the login ID was disabled manually using the **disable login** command.
- Login ID status equals **svn-disabled** indicating that a security violation was detected for that login ID and the login was disabled by the SVN feature.
- Login ID status equals **active** indicating that the login ID is currently logged in.
- Login ID status equals **inactive** indicating that the login ID is not logged in.

Administering Remote Access Security Violation Notification Parameters

To administer the Remote Access component of the SVN feature:

1. Access the "System Parameter Security" form by entering the `change system-parameters security` command from the command line interface.
2. Enable the Remote Access component of the feature by entering a ''y'' in the "SVN Remote Access Violation Notification" field on the "System Parameters Security" form.
3. When the "SVN Remote Access Violation Notification Enabled" field is enabled, the following additional fields appear on the "Security-Related System Parameters" form:
   - Originating Extension

     This field requires the entry of an unassigned extension that is local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for remote access barrier code violations.

The originating extension initiates the referral call in the event of a Remote Access security violation. It also sends the appropriate alerting message or display to the referral destination.

- Referral Destination

  This field requires an entry of an extension, assigned to a station, attendant console, or vector directory number (VDN) that receives the referral call when a security violation occurs. If a VDN is assigned the Time-of-Day routing capability, Call Vectoring may be used to route the referral call to different destinations based on the time of day or the day of the week. The referral destination must be equipped with a display module unless the Announcement Extension is assigned. Administration of the Announcement Extension is also required if the referral destination is a VDN.

- Barrier Code Threshold

  This field requires an entry of the minimum number of remote access barrier code attempts that are permitted before a referral call is made. The value assigned to this field, in conjunction with the “Time Interval” field, determine whether a security violation has occurred. The system default for Barrier code threshold is 10.

- Time Interval

  This field requires the entry of the time interval in which the remote access barrier code attempts must occur. The range for the time interval is one minute to eight hours (0:01 to 7:59), and is entered in the form ‘‘xx:xx.’’ For example, if you want the time interval to be one minute, you enter ‘‘0:01.’’ If you want the time interval to be seven and one-half hours, you enter ‘‘7:30.’’ The system default is 0:03.

- Announcement Extension

  This field requires an entry of a extension that is assigned to the SVN remote access barrier code violation announcement.

4. Administer an ‘‘rsvn-halt’’ button on any station or attendant console (maximum 1 per system). The SVN button location can be determined by entering the command **display svn-button-location.**

Enable/Disable Remote Access Code

To enable remote access that has been disabled following detection of a remote access security violation, or disabled manually with the command **disable remote access**, the user must:

1. Log in to the switch using a login ID with the correct permissions.
2. Enter the command **enable remote access.**

To disable Remote Access, the user must:

1. Log in to the switch using a login ID with the correct permissions.
2. Enter the command **disable login.**

Administering Authorization Code Security Violation Parameters

To administer the Authorization Code component of the SVN feature, the user must:

1. Access the “System Parameter Security” form by entering the `change system-parameters security` command from the command line interface.
2. When the “SVN Authorization Code Violation Notification Enabled” field is enabled, the following additional fields appear on the “Security-Related System Parameters” form:
   - Originating Extension

     This field requires the entry of an unassigned extension that is local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for authorization code security violations.

     The originating extension initiates the referral call in the event of a authorization code security violation. It also sends the appropriate alerting message or display to the referral destination.
   - Referral Destination

     This field requires an entry of an extension, assigned to a station, attendant console, or vector directory number (VDN) that receives the referral call when a security violation occurs. If a VDN is assigned the Time-of-Day routing capability, Call Vectoring may be used to route the referral call to different destinations based on the time of day or the day of the week. The referral destination must be equipped with a display module unless the Announcement Extension is assigned. Administration of the Announcement Extension is also required if the referral destination is a VDN.
   - Authorization Code Threshold

     This field requires an entry of the minimum number of invalid authorization code security violations attempts that are permitted before a referral call is made. The value assigned to this field in conjunction with the “Time Interval” field, determines whether a security violation has occurred. The system default for authorization code security violations threshold is 10.

- Time Interval

  This field requires the entry of the time interval in which the authorization code security violations must occur. The range for the time interval is one minute to eight hours (0:01 to 7:59), and is entered in the form ‘‘x:xx.’’ For example, if you want the time interval to be one minute, you enter ‘‘0:01.’’ If you want the time interval to be seven and one-half hours, you enter ‘‘7:30.’’ The system default is 0:03.

- Announcement Extension

  This field requires an entry of a extension that is assigned to an SVN authorization code announcement.

3. The SVN button location can be determined by entering the command **display svn-button-location**

```
--------------------------------------------------------------------------
monitor security-violations login
--------------------------------------------------------------------------
                      SECURITY VIOLATIONS STATUS
                                      Date:  NN:nn DAY MON nn 199n


                               LOGIN VIOLATIONS


                  Date     Time     Login     Port      Ext
                 01/08    07:51    root      NET-1     4030
                 01/08    07:51    admin     NET-1     4030
                 01/07    07:52    system technician   MGR1
--------------------------------------------------------------------------
--------------------------------------------------------------------------
```

Screen 3-40. Monitor Security Violations Report (Login)

```
--------------------------------------------------------------------------
monitor security-violations remote-access
--------------------------------------------------------------------------
                      SECURITY VIOLATIONS STATUS
                                      Date:  NN:nn DAY MON nn 199n


                  REMOTE ACCESS BARRIER CODE VIOLATIONS
                  Date     Time     TG No   Mbr   Ext   Barrier Code
                  01/08    10:55    31      5     4050  1030
                  01/08    10:54    31      1     4050  2345













--------------------------------------------------------------------------
--------------------------------------------------------------------------
```

Screen 3-41. Monitor Security Violations Report (Remote Access)

```
--------------------------------------------------------------------------
monitor security-violations authorization-code
--------------------------------------------------------------------------
                      SECURITY VIOLATIONS STATUS
                     Date:  NN:nn DAY MON nn 199n

                     AUTHORIZATION CODE VIOLATIONS


Date   Time    Originator     Auth Code  TG No   Mbr   Barrier Code   Ext
01/07  08:33   STATION        1234567                                 84321
01/06  07:32   TRUNK          1233555    35      14
01/03  14:22   REMOTE ACCESS  2222222    31       3    3295912
12/25  16:45   ATTENDANT      1212111                                 84000



--------------------------------------------------------------------------
--------------------------------------------------------------------------
```

Screen 3-42. Monitor Security Violations Report (Authorization Code)

```
 --------------------------------------------------------------------------
 --------------------------------------------------------------------------
Switch Name:______________        Date: xx:xx am  DAY MON xx, 19xx

                 SECURITY VIOLATIONS SUMMARY REPORT
             Counted Since: xx:x am   DAY MON xx,19xx

        Barrier Codes                    Authorization Codes
                 Security                                 Security
 Valid  Invalid  Violations   Originator  Valid  Invalid  Violations
xxxxxxx  xxxxxxx  xxxxxxx     Station    xxxxxxx  xxxxxxx  xxxxxxx
                              Trunk      xxxxxxx  xxxxxxx  xxxxxxx
                           Remote Access xxxxxxx  xxxxxxx  xxxxxxx
         Attendant   xxxxxxx  xxxxxxx  xxxxxxx
          Total      xxxxxxx  xxxxxxx  xxxxxxx

           Successful  Invalid  Invalid   Forced    Login Security  Trivial
 Port Type Logins      Attempts IDs     Disconnects  Violations     Attempts
 SYSAM-LCL xxxxx       xxxxx    xxxxx   xxxxx        xxxxx          xxxxx
 SYSAM-RMT xxxxx       xxxxx    xxxxx   xxxxx        xxxxx          xxxxx
 MAINT     xxxxx       xxxxx    xxxxx   xxxxx        xxxxx          xxxxx
 SYS-PORT  xxxxx       xxxxx    xxxxx   xxxxx        xxxxx          xxxxx

 Total     xxxxx       xxxxx    xxxxx   xxxxx        xxxxx          xxxxx
 --------------------------------------------------------------------------
 --------------------------------------------------------------------------
```

Screen 3-43. Security Violations Traffic Summary Report

Send All Calls

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows users to temporarily direct all incoming calls to coverage regardless of the assigned Call Coverage redirection criteria. Send All Calls also allows covering users to temporarily remove their voice terminals from the coverage path.

Send All Calls is activated by pressing the Send All Calls button or by dialing the Send All Calls access code. It is deactivated by pressing the button a second time or by dialing the deactivate access code.

Details of how Send All Calls is used in conjunction with Call Coverage are given in the "Call Coverage" feature description, elsewhere in this section.

Considerations

Send All Calls provides the option for all incoming calls to be sent to coverage. This is useful when a user needs to be away from the desk temporarily.

Interactions

Send All Calls is used only in conjunction with the Call Coverage feature.

- ACD

  Activation of Send All Calls does not make an ACD agent unavailable for ACD calls.

- Automatic Callback

  Send All Calls does not work with Automatic Callback calls.

- DDC/UCD

  Activation of Send All Calls makes a member of a DDC/UCD hunt group unavailable for calls.

- Internal Automatic Answer (IAA)

  IAA does not apply to calls to the original called extension when the called voice terminal has selected Send All Calls.

- Console Night Service

  With the console in Night Service and the Send All Calls activated, the call covers *after* the “no answer” ring interval is reached.

Administration

Send All Calls is administered by the System Manager. The following items require administration:

- Send All Calls button (per voice terminal)
- Activate and Deactivate access codes for Send All Calls (per system)
- Send All Calls coverage criteria (per coverage path)

Hardware and Software Requirements

No additional hardware or software is required.

Senderized Operation

Feature Availability

This feature is available with all Generic 3 releases.

Description

Reduces the time necessary to place calls to distant locations equipped to receive touch-tone signals (or DTMF) and allows end-to-end signaling to remote computer equipment.

The number dialed and end-to-end signaling digits from voice terminals and trunks are detected by the system and regenerated for transmission over outgoing trunks. The distant end associated with the trunk must be equipped to receive touch-tone signals.

Considerations

This feature provides quicker service to remote touch-tone receiving facilities.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Service Observing

Feature Availability

This feature is available with all Generic 3 releases. Service Observing Logical Agent IDs, Service Observing VDNs, and Service Observing Remotely/By FAC are available with G3V3 and later releases. Vector Initiated Service Observing is available with G3V4.

Description

Allows a specified user, such as a supervisor, to observe a call that involves other users while the call is in progress. The call can be observed on a listen-only or listen-and-talk basis.

In this feature description, the term observer refers to the supervisor or other user who is observing calls. The observer can service observe a station, attendant, logical agent ID, or VDN. See "Service Observing Logical Agent IDs"and "Service Observing VDNs" later in this chapter for more information about these options.

In this feature description, the term “agent” refers to the station, attendant, or logical agent being observed.

Service Observing can be activated in one of two ways: the Service Observe button; or Service Observing (SO) feature access codes (FAC.) See "Service Observing Remotely/By FAC" later in this section for more information about using SO FACs to access Service Observing.

When using a Service Observe button, Service Observing can be activated by any user with access to the voice terminal. To activate Service Observing, the observer presses the Service Observe button followed by the extension number of the agent whose calls are to be observed. To deactivate Service Observing, the observer can either hang up, select another call appearance, or press the Disconnect or Release button.

When Service Observing is activated with the Service Observe button, the observer is in the listen-only mode. Each additional press of the Service Observe button causes the observer to toggle between a listen-only and a listen/talk connection to the call. The lamp signal indicates which mode the observer is in.

The observer can observe consecutive calls without reactivating Service Observing. In other words, as long as an observer has activated Service Observing for a specific agent, he or she can observe that agent’s calls until Service Observing is deactivated.

An optional warning tone can be administered (on a per-system basis) to let the agent and the calling party know that someone is observing the call. The warning tone is a 440 Hz tone. A two-second burst of this tone is heard before the split supervisor is connected to the call. A half-second burst of this tone is heard every 12 seconds while a call is being observed. The warning tone is heard by all parties on the observed call.

It is possible for an observer to activate Service Observing for an agent’s calls even though the agent is not active on a call. In this case, the observer enters the ‘‘waiting’’ mode until the agent receives a call. When the agent receives a call, the observer is bridged onto the call.

If an agent makes an outgoing trunk call and is being observed, Service Observing begins when dialing is completed. For CO trunks with answer supervision, dialing is considered completed when answer supervision is returned. For CO trunks without answer supervision, dialing is considered completed when the answer supervision timeout occurs.

With G3V3 and later releases, when the observer attempts to activate Service Observing, the system returns different tones to indicate if he or she was successful or not. If activation of Service Observing was successful, the system returns confirmation tone. If activation was denied, the system returns intercept, reorder, or busy tone depending upon the reason for denial. If a call becomes ineligible while being observed, the system returns "ineligible" tone. If a logical agent logs out while being observed, the system returns busy tone to indicate that the observing connection has terminated. Tones are provided whether Service Observing is accessed by a Service Observe button or Service Observing FAC. For more information, see "Service Observing User Interface" later in this feature description.

NOTE:
The use of Service Observing features may be subject to federal, state, or local laws, rules or regulations and may be prohibited pursuant to the laws, rules, or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable law, rules and regulations before using these features.

Service Observing Logical Agent IDs

When EAS is optioned, an observer can observe agents based on their logical agent ID rather than by the physical voice terminal where they are located. The observer can enter the logical agent ID extension number of a logged-in agent to set up the observing connection. When the agent is being observed (based on the logical agent ID), the observer is able to monitor every ACD, personal, and Direct Agent call delivered to or placed by the agent, including calls placed to the physical extension.

When an observer attempts to observe a Logical Agent ID, the “Can Be Service Observed?” field on the “Class of Restriction” form is used to determine if the agent can be observed. If the “Can Be Service Observed?” field is set to n, the observer receives the denial indication of a broken flutter and intercept tone. If the agent is not logged in, the observer receives the denial indication of a broken flutter lamp indication and busy tone.

Only one observer is allowed to observe a physical station at one time. If an observer requests to observe a login agent ID extension that is at a station already being observed, the system denies the second observer. Likewise, if an observer requests to observe a physical extension that is already being observed as a logical agent ID extension, the system denies the second observer.

If the logical agent logs out while being observed, the monitoring connection is dropped and the observer receives busy tone.

Service Observing VDNs

The Service Observing feature provides the option of being able to observe VDNs. This option allows you to select a specific VDN and be bridged on to calls (one call at a time) that have just started vector processing for that VDN. The observer hears all tones, announcements, music, and speech that the caller and the agent hear and say, including Call Prompting and caller dialing. Also, the observer hears VDN of Origin announcements, and can replay VDN of Origin announcements if that observer is at a terminal equipped with a VOA Repeat feature button, and they have a COR that allows the VOA announcement to be heard.

Once the system makes an observing connection to a call in vector processing, it maintains the connection throughout the life of the call until the call is disconnected or until the observer hangs up. This is the case even if the call is routed or transferred externally. If the observer does not disconnect after one observed call is disconnected, the system connects the observer to another new call on the same VDN. This type of Service Observing is listen-only as long as the call is in vector processing. Once the call is out of vector processing, observers with listen/talk capability [activated by the listen/talk Service Observing (SO) feature access code (FAC) or the SO button] are able to talk as well as listen. More than one observer can observe the same VDN simultaneously.

If the optional warning tone is administered, the warning tone is heard by the caller and the observer only when the system connects the call to the answering or routed-to internal or external destination (after vector processing is finished). The system continues to apply the periodic tone throughout the duration of the call even if the call is transferred off-switch. Consider using a warning announcement at the first step of the vector to inform the caller of possible monitoring since warning tone cannot be given until after the call is out of vector processing.

The ability to service observe VDNs is enabled on the “System-Parameters Customer-Options” form. As with the other Service Observing capabilities, the “Service Observing” field on the “Class of Restrictions” form must be set to y for both the observer and for the agents that received observed calls.

If an agent or extension to which the call is connected is already being observed, the call cannot be observed.

Multiple users can observe a single VDN simultaneously. However, only one observer can be connected to a given call on the VDN.

To service observe a VDN, the observer specifies the VDN extension in the same manner as he or she would specify a physical extension or logical agent ID extension. However, if the option is not enabled on the “System-Parameters Customer-Options” form, the observer is denied and receives a broken flutter with intercept tone.

NOTE:
When a VDN is observed, the name of the VDN, agent, or trunk appears on the observer’s display as each is accessed by the VDN. For example, during vector processing the VDN name is displayed, but when the call connects to an agent, the agent name is displayed.

Service Observing Remotely/By FAC

This Service Observing feature allows users to observe calls on a listen-only or a listen/talk basis from a remote location or a local station using Service Observing (SO) feature access codes (FAC.) This gives observers dial-up access to Service Observing from any telephone set without a Service Observe button. Different FACs are required for listen-only and listen/talk observing. When observing remotely by FAC is used, the observer cannot toggle between listen-only and listen/talk observing. Service Observing remotely can be used to observe physical extensions, logical agent ID extensions, and VDNs.

Observers who activate Service Observing by way of a FAC will hear tones to indicate the status of the observing in place of lamp signals on the voice terminal. For example, observers will receive confirmation tone when Service Observing is activated.

Once connected to a call, remote observers have all the same features and functions as local observers, with the exception of not being able to replay VDN of Origin announcements if the observing station does not have a VOA Repeat feature button, or if their COR denies them access to the VOA announcement.

Remote Service Observing can be initiated either through Remote Access to PBX services or Call Vectoring. When Service Observing occurs through the Remote Access feature, a remote user gains access to the switch via either a trunk group dedicated to Remote Access or DID to the Remote Access extension. Remote Service Observing operates with all types of DID trunks, including ISDN-PRI, tie trunks, and DCS over analog, T1 or over ISDN PRI.

When Service Observing is initiated through the Call Vectoring feature, a remote user gains access by dialing the VDN extension or a CO trunk that has as its incoming destination a VDN extension. Using route-to commands, the Service Observing vector associated with the VDN can be designed to provide direct access to a specific extension to be observed. Or, it can be designed to access Service Observing dial tone. If Service Observing dial tone is accessed, the observer has the option of entering the number of any extension they are authorized to observe. Call Prompting and Call Vectoring can be combined to provide adequate security and to limit observing capabilities.

The following is a simple example of a Service Observing vector.

```
1.wait-time 0 seconds hearing ringing
2.collect 5 digits announcement 2300
  ("please dial your 5- digit security code")
3.goto step 5 if digits = 12345
4.disconnect after announcement 2000
5.collect 1 digits announcement 2310
  ("enter 1 to observe sales, 2 to observe billing")
6.route-to number 113001 with cov n if digit = 1
  (11=listen-only observe, 3001="Sales" VDN)
7.route-to number 113002 with cov n if digit = 2
  (11=listen-only observe, 3002="Billing" VDN)
8.goto step 5 if unconditionally
```

See *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 555-230-520, for additional information about creating a Service Observing vector.

The ability to service observe remotely and by FACs is enabled on the “System-Parameters Customer-Options” form. As with the other Service Observing capabilities, the “Service Observing” field on the “Class of Restrictions” form must be set to y.’

Service Observing User Interface

The following table shows the Service Observing condition indicators that the observer will receive.

Table 3-76. General Indications to Observer

Condition	Button Lamp	Tone
Not Active	Dark	None
Denied Activation	Broken Flutter	Intercept/Busy/Reorder
Activated	Steady/Winking	Confirmation tone followed by silence or connection to call.
Observing (listen only)	Steady	Hear call
Observing (listen/talk)	Winking	Hear/talk on call
In "wait" state	Flash	None
Denied observing of call	Flash (wait state)	Silence/Ineligible Tone followed by silence.

The following tables show the feedback that the observer can receive during activation or operation of Service Observing. In these tables:

- Wait State refers to the state where the observer is in the Service Observing mode but there is not a call to observe or the call cannot be observed. Under this state a call appearance is not reserved. An idle call appearance must be available for the user to go to the observing state when a new eligible call arrives.
- Ineligible Tone is a tone given to the observer when the call being observed becomes ineligible. A call becomes ineligible when it encounters an ineligible condition as described in "Considerations" later in this feature description. The ineligible tone is the "hold confirmation

tone"— a rapid series of five short beeps (50 msec on/off) of 440 hz. This tone is not given to the observer if the agent receiving the call (who is ineligible for observing) hears zip tone.

Table 3-77. Feedback When Activation Denied

Condition	State	Lamp	Tone
No Such Extension	denied	broken flutter	intercept
Extension Not Observable	denied	broken flutter	intercept
Not Allowed COR[1]	denied	broken flutter	intercept
Extension Has Data Restriction	denied	broken flutter	intercept
Extension Has Exclusion Active	denied	broken flutter	busy
Extension Has Data Privacy Active on Call	denied	broken flutter	busy
Extension Already Observed	denied	broken flutter	busy
Extension is an Observer	denied	broken flutter	busy
Extension Being Busy Verified	denied	broken flutter	reorder
Extension Has a 6-Party Conference	denied	broken flutter	reorder
COR Doesn't Allow SO Activation	denied	broken flutter	intercept
Observe VDN Not Optioned	denied	broken flutter	intercept
Logical ID Not Logged In	denied	broken flutter	busy
Activation to Logical with Physical Observed	denied	broken flutter	busy
Activation to Physical with Logical ID Observed	denied	broken flutter	busy
Maximum VDNs Being Observed	denied	broken flutter	reorder

1. Extension COR cannot be observed or COR for observer calling permission does not allow observing the COR of extension to be observed.

Table 3-78. Feedback When Activation Allowed — At Time of Activation

Condition	State	Lamp	Tone
Active-Eligible Call	observing	steady/ winking	confirmation tone followed by connection to call
No Active Call	wait state	flash	confirmation tone followed by silence
Call Not Eligible	wait state	flash	confirmation tone followed by silence
Call Has "No Observe" COR	wait state	flash	confirmation tone followed by silence
VDN Call Already Being Observed	wait state	flash	silence

Table 3-79. Feedback When Activation Allowed — After Observe Activated

Condition	State	Lamp	Tone
No Active/Eligible Call	wait state	flash	silence
Call in 6-Party Conference	wait state	flash	silence
Call Already Being Observed	wait state	flash	silence
Call Is Being Busy Verified	wait state	flash	silence
Call Has Data Privacy Active	wait state	flash	silence
Call Has Data Restriction	wait state	flash	silence
Call Has Exclusion Active	wait state	flash	silence
Active-Eligible Call (in listen-only mode)	SO listen	steady	hear call

Continued on next page

Table 3-79. Feedback When Activation Allowed — After Observe Activated — *Continued*

Condition	State	Lamp	Tone
Active-Eligible Call (in listen/talk mode)	SO listen/talk	winking	hear/talk on call
Press Button While Observing in Listen-only Mode	SO listen/talk	winking	hear/talk on call
Observer Presses Release	not observing	dark	none
Call Has "No Observe" COR	wait state	flash	silence
VDN Call Already Being Observed	wait state	flash	silence
No Active Eligible Call	wait state	flash	silence
Eligible VDN Call	observing	steady/ winking	hear call
Eligible VDN Call (in vector processing)	SO listen	steady	hear call
Eligible VDN Call (out of vector processing in listen-only	SO listen	steady	hear call
Eligible VDN Call (out of vector processing in listen/talk)	SO listen/talk	winking	hear/talk on call
Press Button While Observing in Vector Processing	SO listen	steady	no change to mode
Press Button While Not in Vector and in listen-only	SO listen/talk	winking	hear/talk on call
Call Being Observed Becomes Ineligible	wait state	flash	ineligible tone followed by silence
Active Call Disconnects	wait state	flash	silence
Logical Agent Logs Out	denied	broken flutter	busy
Observer (without button) Hangs Up	deactivates observing	n/a	n/a

Security

The following sections describe the Service Observing security restrictions.

General Service Observing Security

The following COR restrictions guard against unauthorized observing.

- The observer must have a COR with the “Can Be An Observer” field set to yes.
- The extension to be observed must have a COR with the “Can Be Observed” field set to yes.
- The observer must be assigned calling permissions to all CORs to be observed on the Calling Permissions COR Table.

In addition:

- A logical agent must be logged in to be observed.
- An optional warning tone informs the person being observed and the caller that Service Observing is in effect.

Vector Initiated Service Observing Security

The following COR restrictions guard against unauthorized VDN call observing.

- The VDN extension COR must have an observable COR (“Can Be Observed” field set to yes).
- The VDN extension must be in the Calling Permissions COR Table of the observer.
- Each observer is restricted to observing only those VDN CORs in the Calling Permissions COR table assigned to the observer.
- The VDN destination must have an observable COR.

In addition:

- When the VDN call is connected to its destination, an optional warning tone informs the destination user and the caller that Service Observing is in effect.

Remote/FAC Observing

In addition to the general Service Observing restrictions:

- Separate FACs are required for listen-only and listen-talk observing.

Remote Access Feature Access to Service Observing

Service Observing is activated remotely by a Service Observing FAC that is entered after hearing the PBX dial tone. Barrier Codes and Authorization Codes limit the use of Remote Access to authorized users. Seethe "Remote Access (with Security Measures)" feature for additional information about these codes and other Remote Access security measures. To increase security, use different Authorization Codes for different Service Observing permissions.

In addition:

- Use Call Prompting to create additional access security.
- Use Facility Restriction Levels (FRLs) and other restrictions such as the Authorization Code COR to restrict Remote Access service observer access to other destinations (for example, stations or trunks).

COR “can be observer” and Calling Permissions can be assigned to the VDN, Barrier Code and Authorization Code. The last COR encountered is used to determine observer permissions.

VDN Call Observing Security

The following restrictions can be used with vector initiated Service Observing (G3V4 and later releases) to guard against unauthorized use.

- Call prompting commands can be used in Service Observing vectors to provide passcode protection, and to limit access to observing specific destinations or verified caller entered digits.
- Time of Day/Day of Week checks can be incorporated in Service Observing vectors.
- A vector can be created to be used exclusively for Service Observing.
- For a VDN to be observed as the result of a route-to command, the VDN must have a COR that allows it to be observed.
- The calling permissions of the COR assigned to the Service Observing VDN in conjunction with the “can be observed” settings of the COR assigned to the destination determine what agents, stations, or VDNS can be observed.

Considerations

With Service Observing, a supervisor can monitor an agent while the agent is active on an ACD call. This allows the supervisor to ensure that calls are being handled properly. The supervisor can also assist the agent with the call if necessary.

Although an agent can be a member of more than one split or skill, an agent can only be observed by one supervisor at a time.

Each observer will have only one Service Observe button.

If the agent whose calls are to be observed has a COR that does not permit Service Observing, the observer cannot observe that agent’s calls.

In vector initiated Service Observing, the COR assigned to the VDN used to access Service Observing, the COR assigned to the internal caller extension, and the COR assigned to the logical agent or extension to be observed are used to determine if Service Observing will be allowed. If Service Observing is denied by the called agent/extension COR, Service Observing is denied regardless of the VDN or caller COR. If Service Observing is denied by the caller extension COR, Service Observing is denied regardless of the VDN or called party COR. When the access call routes through multiple VDNs, the COR of the last VDN is used for calling/observing permissions regardless of VDN Override settings.

A maximum of fifty observers can be observing VDNs simultaneously. There is no limit to the number of observers observing the same VDN so long as the total number of observers actively observing VDNs does not exceed fifty.

A call to an observed VDN or agent/extension is ineligible to be service observed when:

- It is already being service observed
- It is conferenced with a station/call that is already being service observed
- It is being busy verified
- It has Data Privacy active (i.e., the call was placed with Data Privacy)
- It has Data Restriction (conferenced with an extension with Data Restriction or a VDN call that reaches an extension with Data Restriction)
- It has Exclusion active (conferenced with an extension with Exclusion active or a VDN call that reaches an extension with Exclusion active)
- It is in a conference or is connected to a station that is in a conference and the addition of the observer will result in more than six parties (including caller, extension/agent and observer)
- It is being observed and is added to a conference where the observer remaining bridged on the call will result in more than six parties (including caller, extension/agent and observer)
- It is a VDN observed call that reaches an extension or VDN with the "can not be observed" COR restriction (The COR of the hunt group, split or skill used to distribute the call to the station/agent will not be checked. Also the COR of stations/agents conferenced with the call will not be checked.)

If two agents with different supervisors are being service observed, and one agent calls the other, the originator’s supervisor observes the call, and the other supervisor is in the waiting mode.

An attendant *cannot* be a service observer. However, an attendant can be observed.

While service observing the only buttons that can be pressed are:

Call Appearance	Bridged Appearance
Position Busy	Auxiliary Work
Auto-ckt Assure	Queue Status (NQC, OQT, AQC, and AQT)
Release (ACD) (This will end Service Observing)	System Night Service
Service Observing	Hold (ignored)

Interactions

The following features interact with the Service Observing feature:

- Agent Login/Logout

  If a Logical Agent being observed logs out, service observing of that ID will be deactivated.

- ASAI

  A call to a VDN being service observed will continue to be monitored after adjunct routing. A call can be routed to a Service Observing FAC by the adjunct routing command in the same way that it can be with the route-to command.

- Assist

  A VDN observer follows a call during an assist operation. The observer monitors the caller while on hold and monitors the agent-supervisor conference after the agent conferences the assist call with the VDN call.

- Attendant Access

  The PBX attendant cannot activate Service Observing by way of the Service Observing FACS. This is because an attendant cannot be a service observer.

- BCMS

  When observing a physical agent (non-EAS), the Report By Login ID shows the physical extension with the login ID. BCMS does not report on Service Observing. The BCMS reports show the normal measured call and/or agent activity related to the Service Observing calls. Neither the activating call or observed call activity is identified as activity for Service Observing.

- Bridged Appearances

  Service Observing is set up to observe all calls to an extension, not all calls to the particular station. If the observer is observing agent extension 3082, the observer will only be bridged onto calls to extension 3082. If the agent with extension 3082 also has a bridged appearance for extension 3282, the calls to extension 3282 will not be observed. The fact that extensions 3082 and 3282 have a call appearance on the same station does not mean that the observer will be able to observe both extensions at the same time.

- Busy Verification of Terminals and Trunks

  An observer cannot service observe an agent's call that is being bridged onto by busy verification. Also, an agent's call that is being bridged onto by Service Observing cannot be busy verified.

- Call Coverage

  An observer cannot service observe a call that has been answered by a covering user until the called agent bridges onto the call.

  A call to an observed VDN will continue to be monitored if the call is routed to a destination that forwards the call.

- Call Forwarding

  A call to an observed VDN will continue to be monitored if the call is routed to a destination that forwards the call.

- Call Park

  A split supervisor cannot park a call while service observing the call. The VDN observer monitoring connection will go with the call.

- Call Pickup

  An observer cannot service observe a call that has been answered by a member of a pickup group until the called agent bridges onto the call. The VDN observer monitoring connection will go with the call.

- Call Waiting

  A call cannot wait on a single-line voice terminal that is being service observed.

- Call Work Codes/Integrated Directory

  The observer will not hear the dialing with these features because the digits are passed to the switch in S-Channel messages.

- CMS

  The system sends an indication to CMS when an observer is bridged on to a VDN call.

- Conference

  The observer cannot use this feature when Service Observing is activated.

A call in a conference can be monitored up to the limit of a 6-party conference. If an agent conferences a call while being observed and the number of parties in the call is less than six, the observer is put into the waiting mode. The observer is bridged onto any call the agent becomes active on before the conference is complete. When the conference is complete, the observer is again bridged onto that call.

If an agent conferences a call while being observed and the number of parties in the call is six, including the observer, the conference is denied. A call to an observed VDN cannot be monitored if bridging on the observer along with the caller and other parties to the call adds up to more than six parties.

If a conference is being service observed as a result of observing an agent who enters the conference, when the agent hangs up the conference will no longer be observed. If a conference is being observed as a result of observing a VDN call, Service Observing continues until the call is transferred to an unobservable destination.

All members of a conference are observed during the conference regardless of whether they would normally be eligible for Service Observing.

If a VDN call that is being observed is conferenced with a call or station that is already being observed, the VDN call becomes ineligible for observing. The observer receives ineligible tone and is then put in the wait state or begins observing another call.

- Converse Command

  Converse split extension ports can be observed as physical extensions. A call to an observed VDN continues to be monitored if the call is answered by a VRU through the converse command.

- Data Restriction/Data Privacy

  A Data Restricted call or a call placed with Data Privacy to an observed VDN will not be monitored. If an observed VDN call is routed to a destination with Data Restriction, the call becomes ineligible and is no longer observed.

- Dialed Number Identification Service

  Observing by VDN provides monitoring by DNIS since the VDNs represent the DNIS of the service dialed.

- Direct Agent Calling

  A direct agent call (placed by an adjunct over ASAI, a voice terminal, or a route-to command) to a Logical Agent ID is monitored by observing the Logical Agent.

- DCS

  A DCS station extension (on another node) is not a valid Service Observing extension. However, a user can observe stations on another node by remote observing that node. A DCS station can only service observe another node via remote Service Observing. Service observing displays are not supported across DCS. A remote observer on another DCS node will hear the same feedback as any other remote service observer.

- Hold

  The observer cannot use this feature when Service Observing is activated. If the hold button is pressed, the action is ignored.

  If an agent places a call on hold while being observed, the observer is put into the waiting mode. A VDN observer will continue to monitor the caller placed on hold.

- Look Ahead Interflow

  If an observed VDN call routes to another location as a result of Look Ahead Interflow, the call will continue to be observed. Warning tone, if optioned for the sending switch, will be applied when the call is accepted by the receiving switch. The periodic tone will continue to be applied while the call is monitored by the VDN observer.

- LWC

  Leave Word Calling cannot be used by any party on a call that is being service observed.

- Move Agent/Change Skills While Staffed

  If move or change skills is requested for a physical or logical agent being service observed, the move or change will take place according to the move or change rules. Observing will continue.

- Multiple Call Handling

  When observing an agent extension or logical ID, only the call that the agent is active on will be monitored. If all calls are put on hold, the observer will hear silence. When observing a VDN call, the monitoring connection stays with the call when the call is put on hold.

- Music on Delay/Music-on-Hold

  A VDN observer hears the music provided to the caller

- Night Service

  A VDN observer monitoring connection will go with the call.

- No Disconnect Supervision Trunk Operation

  Service observing cannot be activated over no disconnect supervision trunks. Denial indication is given to the caller.

- Privacy — Manual Exclusion

  A observer cannot service observe an agent that has activated Privacy — Manual Exclusion.

- Recorded Announcement

  An observed VDN call will continue to be monitored while the call is connected to a recorded announcement. A Verify Announcement call placed by an observed physical or logical agent will be monitored by the observer.

- Redirection on No Answer (RONA)

  An observed VDN call will continue to be monitored when the call is redirected or rings “in limbo” due to the RONA feature.

- Route-to Number/Digits with Coverage

  A route-to number or route-to digits step to a Service Observing FAC or Service Observing FAC + extension fails when Service Observing activation is not allowed or when the extension cannot be observed. In this case, the destination is considered invalid and vector processing goes to the next step regardless of whether coverage is set to y or n.

- Transfer

  The observer cannot use this feature when Service Observing is activated.

  If an agent transfers a call while being observed, the observer is put into the waiting mode. The observer is bridged onto any call that the agent becomes active on before the transfer is complete.

  A VDN observer monitoring connection will go with the call. A VDN call transferred to a nonobservable destination becomes ineligible for observing.

- VDN in a Coverage Path

  A call that covers to a VDN being service observed can be monitored by the VDN observer. An observed VDN call that covers to another VDN will continue to be monitored.

- VDN of Origin Announcements

  Service observers will hear the VDN of origin announcement given to the answering station user and the caller.

- VDN Return Destination

  When Service Observing is activated through a prompting VDN that has return destination assigned, the activation call could return back to the prompting VDN to allow retry of the Service Observing activation if the first attempt fails and the denial indication times out. This is true only when the disconnect due to denial indication occurs after the call leaves vector processing. Failure detected while in vector processing will execute the next vector step. Disconnect due to the disconnect or busy commands will drop the call and not trigger return destination.

  When return destination is triggered, the call is monitored through each return destination operation until the caller disconnects.

- Voice Terminal Display

  If an internal (not remote) observer has a display voice terminal, the display will mirror exactly what is being displayed on the observed physical or logical agent's voice terminal display. For example, **a="3035001234 to Sales SO"**

  When the observer is observing a VDN, the display will show the name of the VDN being observed while in vector processing. After the call leaves vector processing, the display will show the name of the agent or trunk group the call connected to.

- VuStats

  An internal service observer using a 2-line display can activate VuStats for the agent to be observed. The stats for that agent appear on the second line of the observer's display during Service Observing. The VuStats monitoring must be activated before activating Service Observing.

- Zip Tone

  VDN service observers will not hear the zip tone given to the answering agent.

Administration

Service Observing is administered by the System Manager. See *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for complete instructions for administering Service Observing.

Hardware and Software Requirements

No additional hardware or software is required.

Single-Digit Dialing and Mixed Station Numbering

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows easy access to internal hotel/motel services and provides the capability to associate room numbers with guest room voice terminals.

The following dial plan types are provided:

- Single-Digit Dialing
- Prefixed Extensions
- Mixed Numbering

Single-Digit Dialing

A single-digit extension number can be assigned to internal hotel/motel services such as room service. These single-digit extension numbers can be assigned to an individual voice terminal or to a group of voice terminals used, for example, to service the front desk.

Prefixed Extensions

A prefixed extension is made up of a prefix (or first digit) and an extension number with up to five digits. The prefix identifies the call type. The switch collects the dialed digits, removes the prefix digit, and uses the extension number for any further processing.

Assume that the dial plan listed in the following table is administered for a hotel/motel system.

Table 3-80. Hotel/Motel Dial Plan — 1

First Digit	Length 1	2	3	4	5	6
1		TAC				
2	EXT					
3	EXT					
4	EXT					
5			EXT			
6				PEXT		
7					PEXT	
8	TAC					
9	TAC					
0	ATTD					
*		FAC				
#		FAC				

This example dial plan allows the following call types:

- Single-digit access to the hotel/motel attendant (0)
- Ten TACs beginning with the digit 1 (10 through 19)
- Single-digit access to three hotel/motel services using the digits 2, 3, and 4
- Nonprefixed access to as many as 100 hotel/motel staff extensions (500 through 599)
- Room extensions for as many as 100 floors
  - — Access to floors 1 through 9 (prefix digit 6 + [100 through 999])
  - — Access to floors 10 through 99 (prefix digit 7 + [1000 through 9999])
- Toll calling access by dialing TAC 8
- Toll calling access by dialing TAC 9
- Two-digit feature access codes (FACs) beginning with * and # and followed by a second digit

The system identifies a Prefixed Extension number through translation processing. Without the prefix digit, the same group of digits could belong to any call type. In the preceding dial plan example, the digits 71234 are identified as extension 1234 preceded by the prefix 7. If 1234 is dialed, the system interprets it as the two-digit trunk access code 12 because a four-digit extension number beginning with a 1 is not defined.

Mixed Numbering

A dial plan with mixed numbering has the following characteristics:

- Extension numbers can have from one to five digits and can begin with any digit from 0 to 9.
- The first digit, in combination with the number of digits dialed, defines the call type that corresponds to the dialed numbers.

The flexibility of mixed numbers, administrative staff extension numbers, service extension numbers (Single-Digit Dialing), TACs, and FACs may have common leading digits. To differentiate between two numbers with the same leading digit but with different lengths, the system applies a 3- to 9-second interdigit time-out administered on the “System Features” form (Short Interdigit Timer).

Assume that the dial plan listed in Table 3-81 is administered.

Table 3-81. Hotel/Motel Dial Plan — 2

First Digit	Length 1	2	3	4	5	6
1	EXT	EXT	EXT[1]			
2	EXT	EXT	EXT[1]			
3	EXT	EXT	EXT[1]			
4	EXT	EXT	EXT[1]			
5	EXT	EXT	EXT[1]			
6	EXT	EXT	EXT[1]			
7	EXT	EXT	EXT[1]			
8	TAC					
9	TAC					
0	ATTD					
*		FAC				
#		FAC				

1. Time-outs are applied after the first, second, and third digits.

This dial plan example allows the following dial access:

- Single-digit access to the hotel/motel attendant (0)
- Single-digit access to seven hotel services (extensions 1 through 7)
- Two-digit access to 70 hotel/motel services (extensions 10 through 70)
- Guest room extensions for floors 1 through 7 (extensions 100 through 799)
- Toll calling access by dialing TAC 8
- Toll calling access by dialing TAC 9
- Two-digit FACs by dialing * or # plus another digit

Using the preceding dial plan example, the digit 2 can be assigned as the extension number for a hotel/motel service, 22 as an extension number for an administration staff member, and 222 as the extension number for guest room

222. Interdigit time-outs are used by the DEFINITY switch after the first and second digits.

Time-out intervals can be canceled if the user dials # after dialing all required digits.

Considerations

Single-Digit Dialing allows easy access to hotel/motel services.

Mixed Station Numbering allows guest room numbers and room extensions to be the same. Dialing time is a little longer, however, because of the required interdigit time-out interval.

Prefixed extensions greater than five digits in length (including the prefix) cannot be assigned to intercom lists.

A trunk access code and an extension number can only share a first digit if the extension number is shorter than the trunk access code.

Although extensions with the same first digit can have different lengths, data channel extensions should have the maximum number of digits possible in order to avoid timeout problems for data calls that the DEFINITY switch automatically sets up (for example, the CDR link).

Extension numbers and feature access codes can share the same first digit with the extension number being longer (as long as they are not used for AAR/ARS faxs), but these extension numbers only work within the switch. They do not work as remote UDP extensions.

Interactions

The following features interact with the Single-Digit Dialing and Mixed Station Numbering feature.

- Attendant Display and Voice Terminal Display

  If prefixed extensions are used in the system’s dial plan, the prefix is not displayed when the extension is displayed. The Return Call button can be used to dial prefixed extensions, because the system dials the prefix, even though it is not displayed.

- Property Management

  If Prefixed Extensions are assigned in the system, the prefix digit is removed before messages containing the extension number are sent to the PMS.

  Five-digit extensions cannot be exchanged with a PMS until modifications are made to the PMS interface.

- UDP

  The following limitations apply to a DCS environment:

  — Extension numbers that differ in length from the UDP cannot be distributed to other switches.

  — If the first two digits of an extension number correspond to the floor number, floors cannot be serviced by more than one switch.

Administration

The System Manager defines the dial type (extensions, prefixed extensions, trunk access codes, and feature access codes) when the dial plan is administered for the system. The timeout affecting this feature (short interdigit timer) is administrable for G3rV1, G3i-Global, G3V2, and later releases on the “System Parameters — Features” form.

For each first digit (1 through 9, *, and #), a dial type can be defined for each length up to six digits.

Hardware and Software Requirements

No additional hardware or software is required.

Straightforward Outward Completion

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows an attendant to complete an outgoing trunk call for a voice terminal user, without requiring the voice terminal user to hang up.

Considerations

With Straightforward Outward Completion the attendant determines which calls should be allowed and can select the trunk group used for the call.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Subnet Trunking

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides modification of the dialed number so an AAR or ARS call can route over trunk groups that terminate in switches with different dial plans.

Subnet Trunking provides digit insertion, deletion, pauses, and/or wait for dial tone in digit outpulsing, as required, to permit calls to route:

- To or through a remote switch
- Over tie trunks to private network switch
- Over CO trunks to the serving central office

All AAR and ARS calls ultimately reach a point where they can no longer route on a private network. That is, the call reaches a point where another on-network switch is not available for the call. (In an ARS stand-alone configuration, this is the originating switch.) Assuming the call is not denied at this point, then the call must route to one of the following:

- Directly to a party at the local switch.
- To a party at a remote switch (without accessing the public network).
- Through a remote switch to a party at a subtending location (without accessing the public network).
- Directly to a WATS serving office.
- Directly to a local CO or a FX CO, both of which may or may not provide dial access to a long-distance carrier. (The alternative to dial access is for the central office to automatically provide access to a single long-distance carrier of the subscriber's choice.)
- Through a remote switch to the local or FX CO serving the remote switch.
- Through a remote switch to the WATS office serving the remote switch.
- To an EPSCS, Common CCSA office, or ETN office.

Subnet Trunking is not required on calls terminating directly to a party at the local switch. AAR handles these calls.

Subnet Trunking is required on calls routing to or through a remote switch, regardless of the call's destination.

With direct access to a WATS, EPSCS, CCSA, or ETN office, Subnet Trunking is not normally used. The called number on these types of calls is not normally modified. Subnet Trunking is needed only if the number is modified or if the call passes through some intermediate switch, such as a main.

Calls accessing a local or FX CO directly from the terminating switch normally require Subnet Trunking only if access to a long-distance carrier is other than the carrier automatically provided by the CO. In this case, the appropriate dial access code is inserted into the digit string by the system.

Aside from the normal cases, Subnet Trunking can be used to provide added functionality to the system, for example, to convert an AAR number into an international number. Also, Subnet Trunking can modify a digit string so that a Remote Access trunk group can be used on calls. This capability is called equivalent DID and may be useful when a location has Remote Access but does not have DID or NID. (NID is the private network equivalent of DID.)

Addition or deletion of an Area Code on an ARS call does not require Subnet Trunking. ARS handles it via code conversion, as required.

With AAR, an on-network number can be converted into a public network number. In this case, the conversion may include an Area Code insertion via Subnet Trunking.

Any of three special characters may be used with Subnet Trunking:

- Pause — Delays outpulsing of subsequent digits for 1.5 seconds.
- Wait — Can be administered in one of two ways. In the first way, delays outpulsing of subsequent digits for a preprogrammed interval (from 5 to 25 seconds) or, if tone detectors are provided, until dial tone is received from the distant switch or the interval expires, whichever occurs first. In the second way, dial tone must be received before any outpulsing is done.
- Convert-to-tone — Causes all remaining digits to be outpulsed using tone signaling.

Use of these special characters is discussed in the following paragraphs.

During outpulsing of a digit string, it may be necessary to pause or wait for the distant switch to act upon the digits already sent. A programmed pause (a '','' symbol) is used when the required action by the distant switch occurs within 1.5 seconds. Multiple pauses can be used. A ''wait for dial tone'' character (a ''+'' symbol) is used to specify a longer interval with the option of sending or dropping after waiting a period of time for dial tone. The ''time to wait'' is the ''off premises dial tone detect'' time on the "System Parameters — Features" form. Receipt of dial tone automatically cancels the remainder of an interval when tone detectors are provided. If a dial tone detector is not available on a given call, the system uses the wait interval to determine when to resume outpulsing. Multiple waits can be used. If ''outpulse without tone'' on the "System Parameters Country

Options" form is set to no, the trunk is dropped and intercept tone is returned to the calling party.

Dial tones will also be heard if Network Feedback During Tone Detect is set to no on the "System Parameters Country Options" form. Silence will be heard otherwise.

The type of outpulsing, either dial pulse (rotary) or tone, used on a call is specified by the trunk group selected for the call. In some cases, it may be necessary to assure that a portion of the digits are sent using tone signaling. The convert-to-tone character (a ''%'' symbol) is used to indicate that all digits remaining in the string to be outpulsed will use tone signaling.

Digit deletion always begins with the first digit. Subnet Trunking can delete up to 11 digits and can insert up to 36 digits. The last four digits dialed are normally retained. Thus, the new digit string can be up to 40 digits long. Typical uses of digit insertion are the conversion of an AAR call to an international call and the insertion, in the US, of a long-distance carrier code, 10xxx, on a domestic call.

The insertion of a long-distance carrier access code in the string of digits to be outpulsed does not usually require a pause or wait symbol. Interconnecting offices, other than crossbar offices, can handle the code and the called number as a single string. However, a crossbar office returns dial tone after receiving the long-distance carrier code. Thus, a pause or wait is required between the long-distance carrier code and the called number. Likewise in some countries, access to international trunks returns dial tone after dialing the international access code (for example, 00 in Belgium). Note that the user will not typically hear this dial tone, especially if Network Feedback is set to no on the "System Parameters — Features" form.

Considerations

Subnet Trunking allows AAR and ARS calls to access the public network. With AAR, the major advantage is that the call continues although no on-network routes are available to handle the call. With ARS, the major advantage is that calls destined for the public network can route partially over the private network, if there is one. This saves toll charges for a portion of the call.

It is not necessary to include the trunk access code for the trunk group connecting to the distant switch in the string of digits to be outpulsed. In fact, such inclusion must be avoided. Access to the interconnecting trunk group is automatic. Outpulsing the access code, therefore, serves no purpose, and will cause mishandling of the call at the distant end.

The wait interval is a System Parameter. This interval can be from 5 to 25 seconds (in increments of one second).

Up to four special characters can be included in a string of digits to be outpulsed. Each special character counts as two digits.

Interactions

Subnet Trunking is a function associated with the AAR and ARS features. Interactions are the same as those given for AAR and ARS.

Administration

Subnet Trunking is set by the System Manager as a part of AAR and/or ARS administration. The following items require administration:

- Wait — Specify the wait interval used with Subnet Trunking.
- Routing Pattern — Specify the number of digits to delete (beginning with the first digit) and the specific string of digits to insert. Special characters, if any, are included in the inserted string.

Hardware and Software Requirements

Additional tone detectors such as TN744C-Tone Detector/Call Classifier, TN748C-Tone Detector, TN420C-Tone Detector, or TN2182-Tone Clock/Detector/Generator may be required if the special “wait” character is used frequently. TN420C, TN744C, and TN2182 support A-law.

Private Network Access or ARS software is required for Subnet Trunking.

Switch Based Bulletin Board

Feature Availability

This feature requires DEFINITY Communications System G3V3 or later release software. (G3V3 does not have to be enabled on the “System-Parameters Customer-Options” form.) High priority messaging requires G3V4 or later release software.

Description

This feature provides a switch based bulletin board that allows customers and AT&T technical personnel to communicate with each other. A SAT user who has appropriate permissions is able to leave and receive messages on the Switch Based Bulletin Board. In addition, with G3V4 and later release software, AT&T services’ personnel can leave high-priority messages that are displayed on the first ten lines of the bulletin board for easy recognition.

This feature is not intended as a substitute for existing escalation procedures but as an aid to the existing process.

Access to the Switch Based Bulletin Board

When the user logs in to the system, they are notified of any messages in the bulletin board and the date of the ‘‘last entered’’ message. If the bulletin board is at 80% or more capacity, an additional message is displayed indicating how full the bulletin board is (for example, 86%).

If a high-priority message exists, the user is also notified at login that a high-priority message exists and when it was entered. In addition, all logged in users receive a message on the SAT prompt line indicating that high-priority messages have been entered on the bulletin board, but only after a command has been executed on these SATs. In other words, if the SAT is idle, the message will not appear until someone executes a command on that SAT.

The users are responsible for maintaining the bulletin board.

Users who have ‘‘Administer Features’’ permissions can enter or change a message except in the first ten lines on the bulletin board. Users who have ‘‘Display Admin and Maint Data’’ permissions can display, print, and schedule to print the contents of the bulletin board. Users with “init” and “inads” logins can edit the first ten lines of page one of the bulletin board. This area is reserved for high-priority messages.

Bulletin Board Commands

System Commands Used to Access the Switch Based Bulletin Board

Two commands **change bulletin board** and **display bulletin board** were added to access the Switch Based Bulletin Board.

To enter a message into the Switch Based Bulletin Board:

1. Log in to the system with ‘‘Administer Features’’ permissions.
2. Enter the command **change bulletin-board.**
3. Enter your message.
4. Commit your message by pressing “Enter.’’

To display messages in the Switch Based Bulletin Board:

1. Log in to the system with ‘‘Display Admin’’ and ‘‘Maint Data’’ permissions.
2. Enter the command `display bulletin board`**.**

Commands Used to Change Switch Based Bulletin Board Messages

The control keys used for changing existing feature forms are used to change messages on the Switch Based Bulletin Board. To save changes to the “Bulletin Board Message” field, execute the `save translation` command. This also ensures that messages will not be lost during a reset level 3, 4, or 5.

The following characters are allowed in the “Text” field:

- Uppercase and lowercase letters, spaces, numerals, and #$%^ &*()_ -+=[]{};’"<>./?@!~|‘\:.

Tab characters are not allowed. If invalid characters are entered in the “Message” field, a system message is displayed that prompts the user to enter the correct characters for the “Message” field.

Entering Messages into the Switch Based Bulletin Board

After entering the change bulletin board command into the system, the user can change any message in the “Message” field with the exception of high-priority messages. The bulletin board can contain up to three pages of text. The first ten lines of the first page are reserved for high-priority messages that can be entered only by AT&T service personnel. Each page of the “Message” field contains up to 20 lines of text with 40 characters per line. Each line of the bulletin board has an

“Associated Date” field that is populated automatically by the system each time the form is submitted.

Displaying Messages on the Switch Based Bulletin Board

While entering the **display bulletin-board** command into the system, the user can request, print, or schedule to print any message on the Switch Based Bulletin Board.

Considerations

User considerations:

- Only one user can change a bulletin board message at a time.
- The user must have the proper permissions to access the bulletin board.
- The user must maintain the information stored in the bulletin board and is the responsibility of the user to delete old messages. The user can delete information by pressing the ‘‘space’’ bar as the first character in a line, and then pressing the return key. If the bulletin board is full, any new messages overwrite old messages.

System considerations:

- The switch must have G3V3 or later software. (G3V3 does not have to be enabled on the “System-Parameters Customer-Options” form.)
- Two new commands, **change bulletin-board** and **display bulletin-board** are added to the system commands.
- The bulletin board provides three pages for each message, each page (except page 1) has 20 lines of text, and each line can have as many as 40 characters. The first line of the first page is reserved for future enhancements.

Administration

To use the Switch Based Bulletin Board feature, a user must have the correct permissions administered in the “Command Permission Categories” screen.

- To display messages, a user must have the “Display Admin and Maint Data” field set to ***yes*** in the “Command Permission Categories” screen.
- To change messages, a user must have the “Administer Features” field set to ***yes*** on the ‘Command Permission Categories’ screen.

Feature Initialization and Recovery

Feature recovery from a reset condition:

- The bulletin board does not lose information during a reset level 1 or level 2 and is read in from tape during reset levels 3, 4, and 5.

System Measurements

Feature Availability

This feature is standard with all Generic 3 releases except G3vs/G3s ABP. To receive this feature with G3vs/G3s ABP, you must purchase the System Measurements Option package or the Basic Call Center Option Package.

NOTE:
Security Violations is always provided, even with G3vs/G3s ABP when neither the System Measurements Option package nor the Basic Call Center Option Package is purchased.

Description

Provides reports on items such as trunk group usage, hunt group usage and efficiency, attendant group activity and efficiency, and security violations.

Individual reports are available for each of the following:

- Attendant Groups
- Attendant Positions (G3rV1, G3V2, and later releases)
- Automatic Circuit Assurance
- Blockage Study (G3rV1, G3V2, and later releases)
- Call Rate
- Call Summary
- Call By Call Trunk Group
- Coverage Paths
- Coverage Principals
- DS1 Link Performance Measurements
- DS1 Facility Link Performance Measurements (G3rV1, G3V2, and later releases)
- Hunt Groups
- Lightly Used Trunks G3vsV1/G3sV1, G3iV1, G3i-Global, G3V2, and later releases)
- Load Balance Study (G3rV1, G3V2, and later releases)
- Modem Pool Groups
- Outage Trunks
- Performance Attendant Group
- Performance Hunt Group

- Performance Trunk Group
- Performance Summary
- Processor Occupancy and Communications Links
- Route Patterns
- Security Violations
- System Status
- Tone Receiver
- Traffic Summary (G3rV1, G3V2, and later releases)
- Trunk Group Hourly (G3rV1, G3V2, and later releases)
- Trunk Group Summary
- Wideband Trunk Group Hourly (G3V2 and later releases)
- Wideband Trunk Group Summary (G3V2 and later releases)

All reports are on-demand reports. None are given automatically. Reports are available on the G3-MT or a remote administration terminal. The reports can be printed if a printer is associated with the terminal. The reports can also be scheduled to print at the system printer via the Report Scheduler and System Printer feature.

Considerations

Reports provided by System Measurements contain data that is useful to determine group efficiency. Details of specific items on the reports, as well as guidelines to use the data provided, are given in *DEFINITY Communications System Generic 3 Version 4 Traffic Reports*, 555-230-511.

Traffic measurements are automatically accumulated by the system and are available on demand. However, reports are not archived. If needed, reports must be requested periodically. Obtaining a printed copy can aid in maintaining a history of the system traffic.

Detailed information of each call handled by a specific trunk group, if required, must be provided by the CDR feature. Processed CDR data can also provide detailed information on trunk group usage. However, if individual call details are not required for bill-back or cost-allocation, System Measurements should be considered as the means to determine and maintain trunk group efficiency.

Interactions

None.

Administration

Measurements for the Coverage Paths, Coverage Principals, Route Patterns, Trunk Group Hourly, and Wideband Trunk Group Hourly reports are only collected for objects that have been administered using the *change meas-selection* commands.

Hardware and Software Requirements

System administration terminals are required to monitor system measurements. A system printer is required to generate paper copies of the reports. No additional software is required.

System Status Report

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the user to view data associated with attendants, major and minor alarms, and traffic measurements. The information is displayed on the Management Terminal, and presents a basic picture of the System condition. The report can only be displayed by the System Manager and maintenance personnel.

The Status Report is displayed by entering one of the commands listed below. Once the command is entered, the system continually displays the report until it is canceled.

NOTE:
Cancelling a **monitor system view1** or **monitor system view2** report results in automatically logging the user off the system.

- **monitor system view1**

  This command displays the following information:

  — Activation status of all attendants (updated every minute)

  — Maintenance status which includes major and minor alarms for trunk ports, terminal ports, and all maintained objects in the system except terminals and trunks (updated every minute)

  — Traffic measurements for trunk groups, hunt groups, and attendant groups (updated every hour)

- **monitor system view2**

  This report is a subset of the view1 report and displays the same information listed for the view1 report except the last hour's measurement for the hunt groups.

- **monitor traffic trunk-groups** (updated every minute)

  This command displays the following information:

  — Trunk group number

  — Number of members in each trunk group

  — Number of members in each trunk group that are active on a call

  — Length of group queue

  — Number of calls waiting in the group queue

- **monitor traffic hunt-group** (V3 and later releases only) (updated every minute)

  This command displays the following information:

  — Hunt group number

  — Number of members in each hunt group

  — Number of members in each hunt group that are active on a call

  — Length of group queue

  — Number of calls waiting in the group queue

  — Length of time the oldest call in queue has been waiting to be serviced

When a CO call enters a full ACD split queue, CDR and the CMS may show different measurements. CDR measurements indicate the maximum number of calls allowed in the queue, whereas the CMS measurements indicate all calls in the queue plus any call on the CO trunk waiting to enter the split queue.

Considerations

In addition to providing status reports, this feature also provides an indication that the administration terminal is functioning. Any attempt to stop the ‘‘monitor system view1/view2’’ reports logs the administration terminal off the system. Therefore, no unauthorized administration can be performed.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Temporary Bridged Appearance

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows multiappearance voice terminal users in a Terminating Extension Group (TEG) or Personal Central Office Line Group (PCOLG) to bridge onto an existing group call. If a call has been answered using the Call Pickup feature, the originally called party can bridge onto the call. Also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it.

A call incoming to a TEG or PCOLG is not a call to an individual, although one particular member of the group may be the most qualified person to handle the given call. If this individual did not answer the call originally, then he or she can simply bridge onto the call. The call does not have to be transferred.

A call to an individual can be answered by a Call Pickup group member. If the called person returns while the call is still connected, he or she simply bridges onto the call and the answering party hangs up.

Call Coverage provides redirection of calls to alternate answering positions (covering users). A Temporary Bridged Appearance is maintained at the called voice terminal.

The called party can answer the call at any time, even if already answered by a covering user. If the called party does not bridge onto the call, the covering user can use the Consult function of Call Coverage to determine if the called party wants to accept the call. The Consult function uses the Temporary Bridged Appearance maintained on the call. When the consult call is finished, the Temporary Bridged Appearance is removed.

Stations that normally have a temporary bridged appearance with their coverage point, do not have a temporary bridged appearance if the coverage point is AUDIX.

Considerations

Temporary Bridged Appearance permits the desired party to bridge onto a call without manually transferring the call. This provides convenience of operation and also saves time.

Temporary Bridged Appearance does not provide any call originating capability or the capability to answer another party's calls. These capabilities are provided by the Bridged Call Appearance feature.

If two parties are bridged together on an active call with a third party, and if the conference tone feature is enabled, the conference tone is heard.

The Bridged Call Appearance feature enhances Temporary Bridged Appearance by allowing more than one call to an extension to be bridged and by allowing calls to be originated from bridged appearances.

Interactions

Privacy — Manual Exclusion, when activated, prevents other users from bridging onto a call. A user who attempts to bridge onto a call with the Privacy — Manual Exclusion feature active is dropped.

Calls redirected to Call Coverage maintain a Temporary Bridged Appearance on the called voice terminal if a call appearance is available to handle the call. The called party can bridge onto the call at any time. The system can be administered to allow a temporary bridged appearance of the call to either remain at or be removed from the covering voice terminal after the principal bridges onto the call. If two parties are bridged together on an active call with a third party, and the bridging tone is administered to yes, all three parties hears the bridging tone.

Consult calls use the Temporary Bridged Appearance maintained on the call. At the conclusion of a consult call, the bridged appearance is no longer maintained. If the principal chooses not to talk with the calling party, the principal cannot bridge onto the call later.

If a call has, or has had, a Temporary Bridged Appearance, is conferenced or transferred, and redirects to coverage again, a Temporary Bridged Appearance is not maintained at the conferenced-to or transferred-to extension.

Administration

The only required administration is to administer whether or not a temporary bridged appearance is maintained by the covering user after the principal bridges onto the call. (‘‘Keep Held SBA at Coverage Point’’ field on the “Feature-Related System Parameters Screen” form.)

Hardware and Software Requirements

No additional hardware or software is required.

Tenant Partitioning

Feature Availability

Tenant Partitioning is available with G3V4 and later releases.

Description

Provides telecommunications services to multiple independent groups of users through a single PBX. Most commonly, Tenant Partitioning is used to provide telecommunications services from a single provider to multiple tenants of an office complex. It eliminates the need for each tenant to purchase services separately, while giving each tenant the appearance of a dedicated PBX. The feature can also be used to provide group services, such as departmental attendants, on a single-customer PBX.

Services that can be provided to tenants include: telephone equipment; building wiring; public and private network access; and attendant services. In addition, a full range of PBX features can be provided to even the smallest tenant office, including Call Coverage, Call Forwarding, Remote Access, Night Service Routing, Listed Directory Numbers (LDNs), Trunk Answer Any Station (TAAS), Call Center features, distinctive music on hold and others. Tenants can also purchase PBX adjunct features if they are available on the switch, such as voice mail, or CMS activity reporting.

Tenant Partitioning provides advantages to both the telecommunications service provider and to the individual tenants.

- Shared resources offer enhanced services at lower cost to the tenant, with increased profit for the service provider.
- The tenant is provided with the appearance of a dedicated PBX without the expense.
- Attendant services can be provided to all tenants.
- Installation, administration, and maintenance of the PBX can be delegated to a trained, full-time staff.
- With proper administration, tenant resources, including trunking facilities, and all other switch endpoints can be protected from access by other tenants.

Partitioning Tenants

The Tenant Partitioning feature is delivered with one universal tenant. This tenant partition, partition 1, is usually reserved for the service provider. By default it has access to all facilities and can be accessed by any other tenants.

It is the function of the service provider to create additional partitions based on tenant requirements. When deciding which tenant partitions to create, remember:

- Each switch endpoint can be assigned to one and only one tenant partition. And, each switch endpoint must be assigned to a partition. For example, each voice terminal, attendant console, trunk, and virtual endpoint, such as an LDN or VDN, must be assigned to a tenant partition.
- Most tenant partitions are discrete, separate units. By default, all tenants except partition 1 are prevented from accessing stations or trunking facilities belonging to other tenants.
- However, the system administrator can change this default to permit or forbid any individual tenant to access any other specified tenant. For example, tenant 6 could be given permission to call tenants 9 and 16 only.

  Note that if a tenant has permission to call another tenant, it has access to every endpoint belonging to that tenant. For example, if tenant 6 has permission to call tenant 9, tenant 6 can also use any trunking facilities present in tenant partition 9.

  NOTE:
  Even if two extensions are blocked from calling each other by Tenant Partitioning restrictions, either extension can still reach the other by dialing the extension's Direct Inward Dialing (DID) number via the public network.

- If any facilities are to be shared among tenants who do not want complete access to each other's facilities, then the shared facilities must be grouped in a separate partition. For example, if two tenants are going to share a trunk but not have direct access to each other's voice terminals, then that trunk will need to be placed in its own partition so that both tenants can be given access to it.

NOTE:
Tenant Partitioning restrictions do not override COR restrictions. COR restrictions are independent of tenant partitions.

It is also important to consider the following constraints and requirements of access control, attendant services, music sources on hold and route selection when establishing or assigning partitions.

Access Control

By default, tenant partitions are prevented from accessing or being accessed by other partitions. Explicit permission must be given for one tenant to access another. The exception is the "universal" tenant who has permission to call or be called by any tenant.

PBX features, such as call coverage, are limited by tenant-to-tenant access restrictions. For example, suppose tenant 1 included a voice terminal from tenant

2 in its coverage path. If tenant 3 had permission to call tenant 1 but not tenant 2, a call from tenant 3 to tenant 1 would skip the tenant 2 coverage point.

It may also be desirable to set up tenants with special access privileges. For example, a restaurant in an office complex could be given permission to be called by any other tenant. Likewise, permission to call or be called by other tenants would be helpful for building security or PBX administration or troubleshooting.

Another example would be to assign all central office trunks to one tenant partition that could then be accessed by all other tenants.

Attendant Services

One of the advantages of Tenant Partitioning is being able to provide personalized attendant services to each tenant.

There is one Principal and one Night or Day/Night Attendant per attendant group. Each tenant is assigned an attendant group for service. Each attendant group has a separate queue. Queue warning lamps remain dark when Tenant Partitioning is active. However, the information displayed when a queue status button is pressed reflects the status of the attendant group queue. The total number of calls queued for all tenants cannot exceed the system limit.

Attendant groups may serve more than one tenant. In this case, the attendant group is not able to extend a call from one tenant via facilities belonging to another tenant unless the former tenant is permitted to access the other's facilities.

Each tenant may have a designated night service station. Calls to an attendant group in night service are directed to the night service station of the appropriate tenant (when a Night Attendant is not available). When an attendant group is put into night service all trunk groups and hunt groups that belong to tenants served by that attendant group are placed into night service and incoming calls are routed to the night service destination of the appropriate tenant. Each tenant can have its own Listed Directory number (LDN) night destination, Trunk Answer on Any Station (TAAS) port, or Night Attendant.

An attendant can specify that access to a trunk group is under attendant control *if* the trunk group is assigned to a tenant served by that attendant's group. In this case, any valid user attempt to access the trunk group is directed to the attendant group serving the tenant to which the trunk group is assigned.

Multiple Music on Hold

With Tenant Partitioning, each tenant can be assigned a unique source for music to be heard when a call is placed on hold. Customers hear music most appropriate to the business of the tenant, including type of music and special advertisements.

With the exception of calls placed to the attendant group, the caller hears the music source of the tenant partition that he or she called. The caller continues to hear this same music source for the duration of the call regardless of transfers, call forwarding, or call coverage.

One of the following music on hold types can be assigned to each tenant partition:

none	A caller placed on hold hears silence
tone	A caller placed on hold hears a system-wide administered tone.
music	A caller placed on hold hears the music associated with the administered port. The number of possible music sources equals the number of possible tenant partitions. Each partition can have its own music source.

For information on administering Multiple Music on Hold see “Tenant Partitioning” in the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

NOTE:
If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from or pay fees to a third party such as the American Society of Composers, Artists and Producers (ASCAP) or Broadcast Music Incorporated (BMI). You can purchase a Magic on Hold® system, which does not require such a license, from AT&T.

Network Route Selection

Network route selection takes account of the tenant partition number assigned to the caller and to each trunk group. This means that trunk groups belonging to different tenants can be included in the same route pattern. Calls routing to that pattern will select the first trunk group in that pattern that is permitted access by the calling tenant (subject to normal constraints).

Sample Tenant Partitioning

The following is a simple example of how Tenant Partitioning could work in an office complex.

Tenant partition 1, the universal tenant, is assigned to the service provider. All other tenants can call and be called by the service provider. Extensions assigned to this partition are used for a variety of purposes including PBX troubleshooting and administration, and building security.

Tenant partitions 2-15 are each assigned to individual businesses located in the complex. Default restrictions are left in place for these tenants. That is, tenants are prevented from accessing voice terminals, trunking facilities or other switch endpoints belonging to other tenants.

Tenant partition 16 is assigned to the restaurant located in the building complex. All tenants are permitted to call this tenant. However, to prevent the restaurant from accessing trunks and other facilities belonging to tenants, the restaurant is not given permission to call any other tenants.

Tenant partition 17 is assigned to all Central Office (CO) trunk groups. All tenants are given permission to call this tenant.

Tenant partition 18 is assigned to a trunk group that tenants 3 and 7 want to share. Tenants 3 and 7 are given access to this partition, all other tenants are denied access. To prevent toll fraud, tenant 18 is not allowed to call itself.

The ARS route pattern can be the same for all tenants. In this example, the trunk(s) for tenant partition 18 (the private trunk shared by tenants 3 and 7), is placed first in the route pattern. Tenant partition 17 is placed second in the pattern. Tenants 3 and 7 route first to partition 18 and then as a second choice to partition 17. All other tenants are denied access to partition 18 and so are routed directly to partition 17.

All facilities that are not shared, including trunk groups, VDNs, voice terminals, attendant consoles and other endpoints, are assigned to the tenant partition they serve.

The following table summarizes the calling permissions for the different partitions. Yes indicates that the partitions have permission to call and be called by each other, no indicates that partitions cannot call or be called by each other.

Table 3-82. Sample Calling Permissions

Calling Tenant Partition Number	**Called Tenant Partition Number**					
	1	**2, 4-6, 8-15**	**3,7**	**16**	**17**	**18**
1	yes	yes	yes	yes	yes	yes
2, 4-6, 8-15	yes	Each partition can call itself but not the others	no	yes	yes	no
3,7	yes	no	Each partition can call itself but not the others	yes	yes	yes
16	yes	no	no	yes	yes	no
17	yes	yes	yes	yes	yes	no
18	yes	no	yes	no	no	no

Considerations

For a comparison of the DEFINITY system G2 Tenant Services feature with G3V4 Tenant Partitioning, see the *DEFINITY Communications System Generic 2 to Generic 3 Version 4 Transition Reference*, 555-230-636.

Capacities

Table 3-83. Tenant Partitioning Capacities

	G3vs/G3s	G3i	G3r
Tenant Partitions	20	20	100
Attendant Groups	7	16	28

Interactions

Tenant partition identification is not passed between switches. A network of DEFINITY switches will not enforce Tenant Partitioning restrictions without special administration. For example, Tenant Partitioning on a network of DEFINITY switches will not enforce tenant specific tie trunks.

Administration of the following features will require special care to avoid undesired inter-tenant access.

- Bridging
- Call Pickup
- Controlled Restriction
- Facility Busy Indication
- Facility Test Calls
- Integrated Directory
- Inter-PBX Attendant Calls
- Main/Satellite/Tributary
- Malicious Call Trace
- Personal CO line
- Private Networking (AAR)
- Uniform Dial Plan

The function of any feature that specifies a tenant partition will be affected by tenant-to-tenant restrictions. The function of the following features is directly affected by Tenant Partitioning.

- AAR/ARS

  Tenant partitions should not be confused with Time of Day Plan Numbers and Partition Groups in AAR/ARS. Time of Day Plan Numbers and Partition Groups can still be used to select one of eight route patterns for AAR/ARS routing when Tenant Partitioning is in effect

- Attendant and Attendant Group Features

  Tenant Partitioning creates multiple attendant groups. Attendant operations such as direct station or trunk-group select (DCS/DTGS) are subject to tenant-to-tenant restrictions both at selection time and at split time.

  All calls put on hold by an attendant from the attendant group will hear the music source from the attendant group.

- Attendant Control of Trunk Group Access

  An attendant group can only control access to trunk groups that belong to tenants that are served by that attendant group.

- AUDIX, Intuity AUDIX, DEFINITY AUDIX, and AUDIX Voice Power

  AUDIX voice and data ports will be subject to the same tenant-to-tenant restrictions as any other endpoint.

  AUDIX has the capability to restrict one group of subscribers from sending voice mail to another group of subscribers. The tenant partitioning provider will be able to create up to 10 different communities within each AUDIX that have/do not have permission to send voice messages across community boundaries.

- Automatic Wakeup

  The wakeup music will be the music source assigned to the wakeup station's tenant partition.

- Bridged Call Appearance

  All stations with bridged call appearances should be administered to be under control of the same tenant.

- Call Coverage

  Tenant-to-Tenant access restrictions will apply to coverage paths. If a tenant cannot access a particular tenant, it will not be able to access that tenant as part of another tenant's coverage point.

  When an attendant is specified as part of a coverage path, the attendant group of the called tenant, not the calling tenant, is accessed.

  When a call goes to coverage, is answered and then put on hold, the music on hold will be the music source assigned to the tenant partition of the terminal that was originally called.

- Call Detail Record (CDR)

  CDR does not report the Tenant Partition Number of the extension or trunk group used. The Tenant Partition Number must be inferred from the extension or trunk group number.

- Call Pickup

  All stations in a call pickup group should be administered to be under control of the same tenant.

- Call Vectoring/VDN

  A caller routed to a new destination by a vector step hears the music assigned to the new destination.

- CMS

  CMS can be administered to provide CMS reports to each tenant. Each login to CMS can be restricted to control, on a permission basis, only those entities that are assigned to a particular tenant. Outputs to separate printers will allow any tenant to print their own CMS reports. The tenant partitioning provider will be responsible for the administration of the CMS to provide this separation of tenant permissions.

- Dial Access to Attendant

  When a tenant dials an attendant it will receive access to its assigned attendant group.

- Emergency Access to the Attendant

  When a tenant dials emergency access, it will receive access to its assigned attendant group.

- Intercept Treatment

  When access to the attendant is designated as intercept treatment, the caller will receive access to their assigned attendant group.

- Malicious Call Trace

  By default, Malicious Call Trace extensions are assigned to Tenant Partition 1. Therefore, if the Malicious Call Trace feature is enabled, it can be used by any voice terminal with permission to call Tenant Partition 1.

- Multiple Listed Directory Numbers

  Each Listed Directory Number is assigned to a tenant partition.

- Multiple Audio/Music Sources for Vector Delay

  When music is administered on the wait-time vector step, the music played will be the music source of the VDN's tenant number.

- Night Service

  Each tenant can have its own Listed Directory Number (LDN) night destination, Trunk Answer on Any Station (TAAS) port, or Night Attendant.

- PC Interfaces

  Each PC Interface must be assigned to a tenant partition.
- PC/PBX Connections

  Each PC/PBX Connection must be assigned to a tenant partition.
- PC/ISDN

  Each PC/ISDN must be assigned to a tenant partition.
- Remote Access

  Each Remote Access barrier code is assigned to a tenant.
- Traffic Studies

  Traffic Studies do not report the Tenant Partition Number of the extension or trunk group used. The Tenant Partition Number must be inferred from the extension or trunk group number.
- Uniform Dial Plan

  If a Uniform Dial Plan is in place between switches, Tenant Partitioning identification will not be passed between the switches, and so Tenant Partitioning restrictions will not be enforced between the switches without special administration.

Administration

Because some features are not partitioned, special care must be exercised during administrative assignment of those features to prevent inter-tenant access. See the “Interactions” section of this description for a list of these features.

Careful administration by the service provider will be necessary to ensure that:

- All stations in a call pickup group are under control of the same tenant
- All stations with bridged appearances are under control of the same tenant
- Stations in different departments (for the purposes of attendant services) can call each other.

Each object (endpoint, virtual endpoint or other entity) that has an assigned COR must also be assigned a tenant partition number. The exceptions are authorization codes and fixed-assignment virtual endpoints.

A “Tenant Partition” form will need to be filled out for each tenant partition. For complete instructions for administering Tenant Partitioning see "Tenant Partitioning" in the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

Hardware and Software Requirements

No additional hardware is required. Multiple Music on Hold will require separate music sources. Tenant Partitioning requires a separate software right to use fee.

Terminal Translation (with Security Measures)

Feature Availability

This feature is available with Generic 3rV1 and all Generic V2 and later releases.

Description

The TTI Feature allows a user to associate a terminal administered-without-hardware translation to a valid port address by dialing a special digit sequence (feature access code, TTI security code, and extension) from a terminal connected to the port. It also allows a user to disassociate a terminal from its port location by dialing a similar ‘‘disassociate’’ digit sequence.

The feature also includes the administration necessary to change unadministered ports in the switch to ‘‘TTI Ports’’ — or ports from which the TTI association sequence can occur. The TTI Feature, when used with the Administration Without Hardware (AWOH) feature, is expected to reduce costs in the initial provisioning of a system, and also aids in station and data module moves from office to office.

Terminal Association from a Voice TTI Port

When you go offhook, you receive dial tone and the selected call appearance light is lit on multifunction sets. Then you dial the feature access code for station TTI activation. If you are dialing from a set connected to a TTI Port and TTI is turned on for VOICE on the “System Parameters — Features” form you get a second dialtone. Then you dial the TTI security code. If you dial the correct code, you are allowed to continue and get a third dial tone. (If you dial the wrong code, you get intercept treatment and are dropped from the call. If you dial a valid extension and if the extension is being changed on the ‘admin G3-MT’ or ‘G3-MA’ screen, then you get reorder treatment.) You then dial a valid station number and receive confirmation tone. Reorder tone is also an invitation to try the association again. Reorder tone could be given if many users are trying to associate at once in which case trying again immediately probably succeeds. (The extension is valid in the system if it is within the dialplan, is administered without a hardware location specified, and the set type of the extension matches the port type of the TTI Port making the merge request. For example, a digital set type can only be merged to a port on a digital board). As a result, the TTI Port is associated with the extension, creating a fully translated station.

If there is a problem you receive intercept treatment and are dropped from the call. One reason for this might be that you are not dialing from a TTI port or the TTI is not set to VOICE.

Terminal Disassociation from a Voice Station

When you go offhook, you receive dialtone. The selected call appearance light is lit on multifunction sets. You dial the feature access code for station TTI disassociation. If you are dialing from a translated voice extension then you get a second dialtone. You dial the TTI security code. If you dial the correct code, you are allowed to continue and are given a third dialtone.

> **NOTE:**
> If the code is dialed incorrectly, you get intercept treatment and are dropped from the call. If you dial the extension of the set you are originating from, and if the extension is being changed on the 'admin G3-MT' screen, you get reorder treatment. If not, the station's port translation is disassociated from the station, you would get confirmation tone, and it becomes a station administered without hardware.

Then dial the extension and get the confirmation tone.

Other Types of Terminals

The examples given so far have been specific for voice terminals. The TTI feature also supports attendants, data modules, voice/data, and ISDN-BRI terminals.

Attendants

The TTI port translations are the same for digital voice terminals and attendant consoles. To perform an association between a digital TTI voice port and an attendant AWOH extension, a digital voice station must be plugged into the jack assigned to the attendant console, and the TTI association digit sequence must be entered on the digital voice station. Once the TTI association has been completed for the attendant, the digital station must be unplugged and the attendant plugged into the jack.

An attendant can only be separated from its extension translation via the 'Attendant Administration' screen. A TTI disassociate request from an attendant console gives the user intercept treatment.

Analog Queue Warning Ports and External Alert Ports

The analog queue warning ports (used for hunt groups) and the external alert port may be administered with an extension but without hardware. These types of extensions can be associated to an analog port via the TTI feature. The association must be done by an analog set, and then the analog set is unplugged from the port. These extensions cannot be disassociated from their port location with the TTI feature. A TTI disassociate request from one of these ports gives you intercept treatment.

Data Modules

There are tone and digit-sequence differences with data modules.

Tone differences

In the association/disassociation control flow, different tones are given to the voice terminal user to give the current status of the TTI operation. Instead of audible tones, status messages are displayed on a terminal connected to a data module when activating the TTI sequence through keyboard dialing. Table 3-84 lists a comparison of messages you get instead of tones. TTI must be turned on for data in the "Change System Features Parameter" form. These status messages are:

Table 3-84. Voice Terminal Tones vs. Data Terminal Display

Voice Tone	Data Display
dial tone	DIAL:
confirmation	CONFIRMED
reorder	TRY AGAIN
intercept	DENIED
busy	BUSY

Digit sequence differences

For a stand alone data module, the TTI associate/disassociate digit sequence is entered in one line at a dial prompt:

- DIAL: <TTI feature access code><TTI security code><AWOH ext>

Separate prompts are not given for the TTI security code and extension.

Voice/Data terminals

A station with a data terminal (DTDM) is treated the same as voice terminals in the TTI association/disassociation procedures. The DTDM is associated/disassociated from its hardware translation at the same time the station is associated/disassociated. The TTI associate/disassociate sequence can only be initiated through the voice station for DTDM's; it cannot be initiated through the data port.

ISDN BRI Terminals

The TTI *disassociation* sequence for Automatic TEI SPID initializing BRI terminals is identical to the sequence used for voice terminals. However, the *association* sequence is different.

- Disassociation sequence:
  - Feature Access Code
  - Security Code
  - Extension
- Association sequence:
  - Connect the station to any port to get power
  - Program the SPID to the extension with which it will be associated
  - Unplug the station (this is necessary even if the station is connected to its intended port)
  - Connect the station to its intended port (this port should indicate Equipment Type: TTI Port)
  - Receive dial tone:
    - Dial tone - association is complete
    - No dial tone - terminal's SPID is not an available extension number

For BRI sets, dialing the TTI association sequence can be used if a user disassociates a BRI extension from its set and then wants to undo the process by reassociating the set to the same extension.

Security Measures

This feature may be subject to unauthorized use. A person could disrupt normal business functions by disassociating voice or data terminals. The administrator or system manager can help protect against this action by changing the security code often. Another step is to remove the Feature Access Code from the system when it does not need to be used (for example, there are no moves going on at present). Consult the *BCSystems Security Handbook*, 555-025-600, for additional steps to secure your system and find out about obtaining information regularly about security developments.

Considerations

The total number of translated voice terminals and Voice TTI ports in a system is limited to the maximum number of administered stations supported in the system. The total number of translated data terminals and Data TTI ports in a system is limited to the maximum number of administered data modules allowed in the system. Each time a station is unmerged from a port, two station records are created. When the maximum number of administered stations is reached, they are no longer available for TTI.

The provisioning procedures indicate that TTI should be set to TTI Voice first and then to TTI Data. The reason for the preference in order is to reduce the chance of a user trying to use TTI on a data-only terminal that does not have TTI Port Translation. This can happen because the number of voice terminals allowed by the system is twice the number of data terminals. For example, if the system limit for voice terminals is 15,000 and 7,500 for data, then when TTI was turned on for Data first, only the first 7,500 unadministered ports would get TTI port translations. These 7,500 ports may not be the ‘‘correct’’ 7,500 ports.

The TTI Feature, when used with the Administration Without Hardware feature, is expected to reduce costs in the initial provisioning of a system, and also aids in station and data module moves from office to office.

Interactions

The following describes how association/disassociation affects other features in use or active at the time the association/disassociation is performed. Some functionality is limited to multibutton sets.

- Automatic Callback

  If a station becomes disassociated while a normal station has automatic callback activated for that station, the normal stations auto-callback light is turned off and the auto-callback sequence is broken.

- Bridged Call Appearance

  If a station has a bridged call appearance of another station, that is off hook, the station administered with the bridged call appearance can disassociate at any time and not disrupt the call in progress on the bridge.

  If a station with a bridged appearance of a normal station associates itself while the extension for the bridged appearances involved in the call, that station associating itself can join the call after it has completed the association sequence.

  Disassociation cannot be performed from a bridged call appearance. Disassociation (TTI) of a station must be performed from the port on which the set resides.

- Call Coverage

  If a station disassociated while send-all-calls or goto-coverage is active, then these features remains active while the station has no hardware translation.

- Call Coverage Answer Group

  Members in a group cannot be disassociated while there is still an incoming call to the group. In this case all sets are seen as active and therefore cannot be disassociated.

  If any endpoint previously administered without hardware translation is inserted into translation via TTI or station administration, that endpoint is excluded from all transactions taking place in the call coverage answer group.

- Call Forwarding

  A station can disassociate while call forwarding is active. If a destination extension for call forwarding disassociates, call forwarding to that extension remains active.

- Call Park

  A terminal is not allowed to be disassociated while placed in a parked state by another set. It is virtually put on hold.

- Call Pickup

  The primary extension of a call pickup group cannot disassociate while a call is attempting to terminate. All secondary members are allowed to disassociate while a call is terminating to a primary extension. If a call has terminated to any extension within a group, any member of the group may disassociate. If a call has or has not terminated, and a member of a group disassociates, that member does not join the group for the call that is currently in progress, but is available for all subsequent calls to that group.

- Conference

  See "Hold" below.

- Customer Provided Equipment (CPE) Alarm

  If a station that has been administered with a customer provided equipment alarm, enters the system translation while an alarm is active, then the newly associated station receives the alarm indication upon entering the system.

- Hunt Group UCD/DDC

  Any member of a hunt group that is not the target of the current call (not ringing), can disassociate itself. The primary (ringing) extension cannot.

  If any endpoint previously administered without hardware translation is inserted into translation via TTI or station administration, that endpoint is excluded from all transactions taking place in the hunt group. This means that a member does not join the other members of an incoming call for one that is taking place during the ''binding'' of the station to the port location. The member is able to join in all subsequent calls.

- Hold

  Neither party, the one that puts another on hold, nor the one that is being held can perform disassociation.

- Incoming Destination

  See the “station-to-Station Call” section.

- Intercom Group

  See the “station-to-Station Call” section.

- Message Light

  All messages need not be deleted prior to disassociation. If a station should receive messages while it is in the untranslated state, when the terminal receives hardware translation, the message light is updated.

- Send All Calls

  Send all calls remains active when a station becomes disassociated.

- Station-to-station Call

  No disassociation is performed until a call is established. No disassociation can occur while a set is ringing.

- Terminating Extension Group

  Members in a group cannot be disassociated while there is an incoming call to the group. All members of the group are seen as active, since they are ringing and busy indicator lights are flashing.

  If any endpoint previously administered without hardware translation via TTI or station administration, that endpoint is excluded from all transactions taking place in the termination extension group. This means that a member does not join the other members of an incoming call for one that is taking place during the binding of the station to the port location. The member is able to join in all subsequent calls to the group.

- Transfer

  After a connection has been established from the first to the third party, the second party, the one who performed the transfer, can be disassociated. Parties one and three are treated as a station-to-station call.

- Attendant

  Because the attendant is a central focus for incoming calls, it is advisable to have the attendant in Position Busy Mode, so as to route incoming calls away from the attendant attempting to disassociate. It is possible to have the attendant in Position Available Mode and still disassociate. Any calls queued, held, or seen as active for the attendant, prevents disassociation.

- Incoming Destination

An incoming destination is not allowed to be disassociated while a call is in progress to that extension, regardless of whether the extension is a station or an attendant.

- Attendant Night Service

  The night service station cannot be disassociated while in night service. For the removal of any endpoint administered without hardware translation while night service is activated, see that particular section.

- Attendant Release Loop Operation

  All calls held with the release loop operation by the attendant are reclassified as attendant group calls if the attendant disassociates before the attendant timed reminder interval expires.

- Attendant to Attendant

  To remove an attendant see the station-to-station section.

- Night Service-Trunk Group

  If the night service destination is a station, disassociation operates the same way as disassociation for stations.

Administration

The following provides information on the administration of TTI.

TTI Port Assignment

A TTI Port may be assigned to other resources through the appropriate administration screen, just as if the port was unadministered. A TTI Port address may be assigned to an extension on the “Station” form, for example.

Terminal Moves and Site Data

TTI can be used for implementing station moves. That is, the user disassociates the station from its port with TTI, unplugs the station from the jack, plugs the station in a another jack in a different location, and associates the station to its new port location with TTI. Certain translations is associated with the port and other translations is associated with the extension during a TTI station move.

Table 3-85 lists these associations.

Table 3-85. Site Data Association

Port	Extension
Building	Color
Floor	Speakerphone
Room	Headset
Jack number	Cord length
Cable	Mounting

The ability to automatically move appropriate translation with the station is needed to prevent time-consuming manual translation changes.

Administration without Hardware

If a fully translated station, attendant, or data module is removed via administration, or if it is disassociated from its port by putting an ***X*** in the “Port” field on the ‘Station’ or ‘Data Module’ screen, the port becomes a Voice or Data TTI Port if TTI is set to ***voice*** or ***data*** in the system. If TTI is ***off*** in the system, then a port disassociated from its station or data module becomes an unadministered port.

Enabling/Disabling TTI

The following is information regarding the enabling and disabling of TTI.

System Parameters Customer Option Screen

A “TTI” field on the “System Parameters Customer-Option” screen gives Services Personnel the option to turn on the TTI feature for a customer. A ***y*** (yes) after this field denotes that TTI is available; a ***n*** (no) denotes that TTI is not available on the switch.

System Parameters Features Screen

If TTI is allowed on the switch via the customer option screen, a field on the ‘System Parameters Features’ screen is used to enable or disable TTI on a system wide basis.

System Parameter Operation

The “TTI System Parameter” field may have one of five values:

- TTI Voice — TTI is turned on, the default translation generated for a TTI Port is for a voice or voice/data terminal, and the activation/deactivation sequence comes from a station keypad.
- TTI Data — TTI is turned on, and the default translation generated for a TTI Port is for a stand alone data module. The activation/ deactivation sequence comes from a data terminal.
- Off — TTI is turned off for the system.
- Suspended — TTI was previously on for TTI Voice or TTI Data but TTI port translation generation is temporarily prevented. Existing TTI port translations are not removed but no further translations are allowed (including via board insertions) until this flag is manually set to TTI Voice or TTI Data or Resume.
- Resume — When TTI is manually suspended, the system administrator may set the "TTI System Parameter" field to `Resume`. The state of TTI is then set back to what it was before TTI was manually suspended.

When TTI is turned on for the system (the first two cases above), the following actions takes place:

- If TTI was previously turned on but in a different state for example, a voice to data state or vice versa, the old TTI translations are removed and the new ones added on a board by board basis.
- If TTI is set to TTI Voice mode, then default TTI translations are generated for every unadministered port on all digital, hybrid, and analog boards.
- If TTI Data translations exist for a digital board when TTI is set to Voice, the TTI Data translations are removed and TTI Voice translations are added in their place.
- If TTI is set to TTI Data mode, then default TTI translations are generated for every unadministered port on all digital and data line boards in the system.
- When TTI is set to Data, the TTI Voice translations are removed and TTI Data translations added in their place.
- Whenever a new digital board is inserted when the system is in TTI Data mode, or when a digital, hybrid, or analog board is inserted when the system is in TTI Voice mode, the unadministered ports on the board becomes TTI ports.
- When TTI is turned off, all translation for the TTI ports are removed in the system; these ports return to an ‘‘unadministered’’ state.

TTI Status Form

This form checks on the status of TTI port translation generation after TTI has been turned on for the system.

Save Translation Operation

All TTI port translations are saved if a ‘‘save translation’’ administration operation is requested. This lessens the impact to booting and board insertion by reading the translations from disk or tape instead of recreating them.

If a ‘‘save translation’’ operation is requested while the system is actively generating TTI port translations as a result of turning the TTI system parameter on, the TTI port generation process is automatically suspended while the ‘‘save translation’’ operation executes. When the operation finishes, the TTI port generation is automatically resumed.

Transaction Logging

When a set is associated or disassociated through the use of TTI, the transaction appears in the system transaction log. This log can be displayed by the ‘‘list history’’ command on the administration console.

TTI Security Code

A single 1 to 7 digit TTI security code is administrable on the ‘System Parameters Features’ screen. This code is entered during the TTI associate/disassociate sequence after the TTI associate/disassociate feature access code. It is not an optional field.

Feature Access Codes

The TTI Association and Disassociation Feature Access Codes must be administered on the ‘Feature Access Code’ screen before any TTI transactions can occur.

Hardware/Software Requirements

None.

Terminating Extension Group (TEG)

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows an incoming call to ring (either audible or silent alerting) as many as four voice terminals at one time. Any user in the group can answer the call.

Any voice terminal can be administered as a Terminating Extension Group (TEG) member; however, only a multiappearance voice terminal can be assigned a TEG button with associated status lamp. The TEG button allows the user to select a TEG call appearance for answering or bridging onto an existing call but not for call origination.

When an incoming call is answered by a TEG member, a Temporary Bridged Appearance is maintained at the multiappearance voice terminals in the group. However, the Temporary Bridged Appearance is not visible on a call appearance. Any of the TEG members can bridge onto the call by pressing the TEG button, if assigned. For example, suppose an incoming call has been answered by a certain TEG member, and this TEG member does not have the needed information. If another member has the needed information, that member needs only to bridge onto the call to provide the information.

The Privacy — Manual Exclusion feature can be assigned to any or all of the multiappearance voice terminals in a TEG. This allows the answering TEG member, by pressing the Exclusion button, to prohibit bridging by other group members. Pressing the button again reestablishes the bridging capability.

A single-line voice terminal administered as a TEG member is rung for a TEG call if it is idle.

A TEG is established by associating the individual member's extension number with a TEG extension number. The members have call placing and receiving privileges for their individual extension numbers, as defined by the assigned COR. Each TEG is also assigned a COR. The group COR overrides an individual member's COR on calls to the group. Thus, the members could be Termination Restricted, but still receive TEG calls.

Considerations

TEGs are useful when it is desirable to have incoming calls to a specific extension number ring more than one voice terminal simultaneously. For example, the appliance department of a large retailer might have three voice terminals. Anyone in the department can answer the call. The salesperson most qualified to handle the call can bridge onto the call from either of the other two voice terminals.

A voice terminal user can be a member of more than one TEG, but can have only one TEG button for each group.

A TEG can only handle one TEG call at a time. If any member of a TEG is active on a call to the TEG, a second call to the TEG waits until the first call is terminated before it rings the TEG. The TEG members have no way to know when a TEG call is waiting. If a coverage path is assigned to the TEG, the waiting call routes accordingly.

Interactions

The following features interact with the TEG feature.

- Automatic Callback

  This feature cannot be activated for a TEG.

- Bridged Call Appearance

  Calls to a TEG cannot be bridged, except via a Temporary Bridged Appearance.

- Call Coverage

  Calls to a TEG can be redirected to alternate answering positions whenever the Call Coverage feature is assigned and no group member is available to answer the call. If any member of a TEG is active on a TEG call, all subsequent TEG calls redirect to coverage. However, a TEG cannot serve as an alternate answering position. In other words, a TEG can have a Call Coverage path assigned, but cannot be a point in a Call Coverage path.

  A Send Term button for the TEG can be assigned to any or all group members who have multiappearance voice terminals. When the Send Term button is pressed, all calls to that TEG redirect to coverage. The associated status lamp lights on the activating voice terminal and all other voice terminals with a Send Term button. Any member with a Send Term button can deactivate Send Term by pressing the button. The Send Term status lamp then goes dark on all voice terminals. Incoming calls are again directed to the group.

- Call Park

  A TEG call cannot be parked on the group extension number; however, a group member answering a call can park such a call on his or her own extension number.

- DDC and UCD

  A TEG cannot be a member of a DDC or UCD group.

- Internal Automatic Answer (IAA)

  Calls to a Terminating Extension Group extension are not eligible for IAA; however, calls placed to the individual extension are eligible.

- LWC

  LWC messages can be stored for a TEG and can be retrieved by a member of the group, a covering user of the group, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization can be assigned to the message retriever. Also, a remote Automatic Message Waiting lamp can be assigned to a group member to provide a visual indication that a message has been stored for the group. One indicator is allowed per TEG.

- Temporary Bridged Appearance

  At multiappearance voice terminals in the TEG, a Temporary Bridged Appearance is maintained after a call is answered. This allows other members of the group to bridge onto the call.

The Privacy — Manual Exclusion feature, when activated, prevents other TEG members from bridging onto a call. A TEG member who attempts to bridge onto a call with Privacy — Manual Exclusion activated is dropped.

Administration

TEGs are administered by the System Manager. The following items require administration for each group:

- Group number
- Extension number for the group
- Group name (for display purposes)
- Call Coverage path number
- Group COR
- Up to four group member extension numbers

The following items can be administered to multiappearance voice terminal TEG members:

- TEG button with associated status lamp.
- Exclusion button associated with the TEG extension number. (Keeps other group members from bridging onto an existing call.)
- Send Term button for the TEG extension number.
- Remote Automatic Message Waiting lamp (one per TEG extension number).
- Audible or silent alerting.

Hardware and Software Requirements

No additional hardware or software is required.

Time of Day Routing

Feature Availability

This optional feature is available with all Generic 3 releases if you select Private Networking (AAR) or ARS.

Description

Provides the most economical routing of ARS and AAR calls, based on the time of day and day of the week that each call is made.

With Time of Day Routing, a company can take advantage of lower calling rates during specific times of the day and week. In addition, companies with locations in different time zones may be able to maximize the use of facilities by utilizing those in a location that has a lower rate at different times of the day or week. This feature can also be used to change the patterns during the times an office is closed in order to reduce or eliminate unauthorized calls.

Time of Day Routing uses the Time of Day Plan Number assigned by the COR feature. A Time of Day Routing Plan (see Screen 3-44). can be administered for each of the eight Time of Day Plan Numbers. When a user makes an AAR/ARS call, the call is routed according to the Time of Day Routing Plan associated with that user's Time of Day Plan Number.

```
                                                               Page 1 of 1

                               TIME OF DAY ROUTING PLAN 1
      Act    PGN    Act    PGN    Act    PGN    Act    PGN    Act    PGN    Act    PGN
      Time    #     Time    #     Time    #     Time    #     Time    #     Time    #
Sun   00:01   1     __:__   _     __:__   _     __:__   _     __:__   _     __:__   _
Mon   00:01   1     __:__   _     __:__   _     __:__   _     __:__   _     __:__   _
Tue   00:01   1     __:__   _     __:__   _     __:__   _     __:__   _     __:__   _
Wed   00:01   1     __:__   _     __:__   _     __:__   _     __:__   _     __:__   _
Thu   00:01   1     __:__   _     __:__   _     __:__   _     __:__   _     __:__   _
Fri   00:01   1     __:__   _     __:__   _     __:__   _     __:__   _     __:__   _
Sat   00:01   1     __:__   _     __:__   _     __:__   _     __:__   _     __:__   _
```

Screen 3-44. Time of Day Routing Plan 1

Time of Day Routing provides the flexibility to change the routing of outgoing calls as many as six times a day, each day of the week. Each of the six possible routing plans (from a pool of eight) is assigned an activation time and a RPN (shown as `PGN #` on the "Management Terminal Screen" form). (***PGN #*** is the Partition Group Number.) The RPN selects the ARS or AAR Digit Analysis Table associated with the RPN to be used to route the calls. When a particular RPN is activated, it remains in effect until the activation time of the next RPN or until it is overridden. Activation of a new RPN does not affect calls in progress.

When a user dials the AAR or ARS feature access code followed by the desired number, and the system has collected enough digits to search for the routing pattern, the system will then select one of the eight Time of Day Routing Plan tables based on the Time of Day Plan Number assigned to the user’s COR. Then, depending on the day of the week and the time of the day the call is being made, the system selects the RPN (***PGN #*** on the ‘Management Applications’ screen). The system then uses the Digit Analysis Table (G3i) associated with this RPN to select the Routing Pattern to be used on the call.

When Time of Day Routing is activated, it applies to all outgoing calls from voice terminals, attendants, data terminals, remote access users, incoming tie trunks, ISDN-PRI trunks, and trunks used for call forwarding to external numbers.

For additional information on "Automatic Alternate Routing (AAR)" and "Automatic Route Selection (ARS)", see the respective feature descriptions elsewhere in this chapter.

Time of Day Routing Example

Assume the following:

- Jim is the user at extension 1234.
- Extension 1234 is assigned a COR of 2.
- COR 2 is assigned a Time of Day Plan Number of 3.
- The Time of Day Routing Plan table for Time of Day Plan Number 3 is administered as shown in Screen 3-45.

When Jim comes into work on Monday morning at 8:30 and at that time makes an ARS call (dials the ARS access code followed by the number of the person he is calling), the system will look at the Time of Day Plan Number assigned to Jim’s COR to determine which Time of Day Routing Plan table is to be used.

Since Jim has a COR of 2 and COR 2 has a Time of Day Plan Number of 3, the system will use Time of Day Routing Plan 3 to route the call.

According to Time of Day Routing Plan 3, all calls made between 8:00 a.m. and 12:00 p.m. route according to the Digit Analysis Table associated with RPN 2. Therefore, these tables will be used to find a Routing Pattern for the call.

If Jim makes a call between 12:00 p.m. and 1:00 p.m. on Monday, the same Time of Day Routing Plan table (number 3) is used and the call is routed according to RPN 1. See Screen 3-45.

```
                                                      Page 1 of 1

                         TIME OF DAY ROUTING PLAN 3

    Act   PGN Act       PGN   Act     PGN   Act      PGN    Act      PGN  Act
    Time   #  Time       #    Time     #    Time      #     Time      #   Time
Sun 00:01  1 __:__       _    __:__    _     __:__    _     __:__   _  __:__  _
Mon 00:01  1 __08:__00  _2  __12:__00  _1   __13:__00  _2  __17:__00 _1 __:__  _
Tue 00:01  1 __08:__00  _2  __12:__00  _1   __13:__00  _2  __17:__00 _1 __:__  _
Wed 00:01  1 __08:__00  _2  __12:__00  _1   __13:__00  _2  __17:__00 _1 __:__  _
Thu 00:01  1 __08:__00  _2  __12:__00  _1   __13:__00  _2  __17:__00 _1 __:__  _
Fri 00:01  1 __08:__00  _2  __12:__00  _1   __13:__00  _2  __17:__00 _1 __:__  _
Sat 00:01  1   __:__    _     __:__    _     __:__    _     __:__   _  __:__  _
```

Screen 3-45. Time of Day Routing Plan 3

Overriding the Time of Day Routing Plan

An attendant or a voice terminal user with console permission and a display can temporarily override the activating user's current routing plan. This can be accomplished by either of two methods:

- Immediate Manual Override
- Clocked Manual Override

Both types of override are discussed in detail in the following paragraphs. It should be noted that both types of override cannot be activated simultaneously. If either type is activated while the other is still in effect, the newly activated override goes into effect and the other override is automatically deactivated.

There is no indication via the Management Terminal that either type of override has been activated. Also, since these overrides are temporary, they are not saved to translations during a "save translations." This way, the overrides are not reactivated at a later time when the system reboots. Therefore, in the event of a system reset, these overrides are deactivated.

Immediate Manual Override

This type of override is button activated, takes place immediately upon activation, and remains in effect for the activating user's Time of Day Plan Number (and all who share this Time of Day Plan Number) until it is manually deactivated or the next scheduled change in the Time of Day Routing Plan takes place.

When a user presses an idle Immediate Manual Override button, the associated lamp flashes (unless the user is an attendant, in which case, the lamp will not flash) and the display shows:

OLD ROUTE PLAN: x   ENTER NEW PLAN:

> **NOTE:**
> x is the number of the routing plan currently in effect.

The user then enters an RPN (one to eight) using the dial pad and the display updates to:

OLD ROUTE PLAN: x   NEW PLAN: y

The user then presses the flashing Immediate Manual Override button or the Normal button. The Immediate Manual Override button lamp then lights steadily and the display is returned to the NORMAL mode. At this time, the old RPN (x) is deactivated and the new RPN (y) is in effect.

The override attempt is denied if any of the following occurs:

- The activating user is not an attendant or a voice terminal user with console permission and a display.
- The activating user enters anything other than 1 through 8 when prompted by the display for the new RPN.
- The activating user presses the flashing Immediate Manual Override button before entering a new RPN.
- The activating user enters another display mode (that is, ‘‘Normal’’) before completing the attempt.

When Immediate Manual Override is activated by a user, the override is also in effect for all other users with the same COR Time of Day Plan Number.

A user can deactivate the override by pressing the steadily lighted Immediate Manual Override button. When the override is deactivated, the scheduled routing plan goes into effect. The on/off status of the button lamp is tracked by all other users with the same COR Time of Day Plan Number who have Immediate Manual Override buttons. Therefore, a user other than the activating user can deactivate the override.

Clocked Manual Override

This type of override requires the user to manually enter a specific day and time for activation and deactivation of the override. The override occurs at the specified day and time of activation and remains in effect until the specified day and time of deactivation. When the override is deactivated, the normally scheduled routing plan goes into effect.

When a user presses an idle Clocked Manual Override button, the associated lamp flashes (unless the user is an attendant, in which case, the lamp will not flash) and the display shows:

ENTER ACTIVATION ROUTE PLAN, DAY & TIME

The user then uses the dial pad to enter an RPN (1 to 8), followed by the day

(1 to 7, where **1** is for Sunday and **7** is for Saturday) and the activation time (0000 to 2359, military time). The display then updates to:

ROUTE PLAN: x FOR: yyy ACT-TIME: zz:zz

In the above display, x is the new RPN, yyy is a three-letter abbreviation for the day of the week, and zz:zz is the activation time for the override.

The user then presses the flashing Clocked Manual Override button again, the lamp continues to flash, and the display shows:

ENTER DEACTIVATION DAY AND TIME

The user then uses the dial pad to enter the day (1 to 7, where **1** is for Sunday and **7** is for Saturday) and the activation time (0001 to 2400, military time). The display then updates to:

ROUTE PLAN: x FOR: yyy DEACT-TIME: zz:zz

In the above display, x is the new RPN, yyy is a three-letter abbreviation for the day of the week, and zz:zz is the deactivation time for the override.

The user then presses the flashing Clocked Manual Override button or the Normal button. The Clocked Manual Override button lamp then lights steadily and the display is returned to the NORMAL mode. At the entered times and days, the new RPN (x) is activated and then deactivated.

The override attempt is denied if any of the following occurs:

- The activating user is not an attendant or a voice terminal user with console permission and a display.
- The activating user enters anything other than valid information when prompted by the display.

- The activating user enters another display mode (that is, normal) before completing the attempt.

When Clocked Manual Override is activated by a user, the override is also in effect for all other users with the same COR Time of Day Plan Number.

A user can deactivate the override by pressing the steadily lighted Clocked Manual Override button. The on/off status of the button lamp is tracked by all other users with the same COR Time of Day Plan Number who have Clocked Manual Override buttons. Therefore, a user other than the activating user can deactivate the override.

Considerations

Time of Day Routing enhances AAR and ARS by allowing companies to choose more economical call routing based on the day of the week and the time of the day.

Time of Day Routing provides up to eight different routing plans. The routing plan can be changed as many as six times a day, each day of the week. At least one time period and RPN must be assigned to each day of the week.

A maximum of 10 Immediate Manual Override buttons and 10 Clocked Manual Override buttons is allowed per Time of Day Plan Number. These buttons can only be assigned to and used by attendants and voice terminal users with both console permission and a display. Each attendant console or voice terminal can be assigned a maximum of one Immediate Manual Override button and one Clocked Manual Override button.

Time of Day Routing can only be used if AAR/ARS Partitioning and AAR and/or ARS are provided.

Interactions

The following features interact with the Time of Day Routing feature.

- Abbreviated Dialing

  For Time of Day Routing purposes, a user’s own COR Time of Day Plan Number is used when accessing an Abbreviated Dialing privileged list. The call is processed the same as if the call had been dialed directly using AAR or ARS.

- Attendant Extended Calls

  When an attendant extends (places) a call for a station user or trunk and that call uses AAR or ARS to process the call, the call is routed according to the Time of Day Plan Number of the attendant’s COR.

- Authorization Codes

  If a user's FRL has been changed through the use of an Authorization Code, the COR FRL associated with the entered Authorization Code will be used in routing pattern selection.

- AAR

  When Time of Day Routing is assigned, all AAR calls use the Time of Day Routing Plans for routing calls.

- ARS

  When Time of Day Routing is assigned, all ARS calls use the Time of Day Routing Plans for routing calls.

- Bridged Call Appearance

  The COR Time of Day Plan Number of the primary extension applies to calls originated from a bridged call appearance of the primary extension.

- Call Forwarding All Calls

  If a user has activated Call Forwarding All Calls, and AAR or ARS is used to route an incoming call to the forwarded-to number, the COR Time of Day Plan Number of the calling party is used to route the call.

- DCS

  Care should be taken when making Time of Day Routing assignments in a DCS environment. Depending on a user's Time of Day Plan Number, a user may or may not be routed to a DCS trunk group. If a user is not routed to a DCS trunk group, feature transparency will be lost.

  When a call routes over a DCS trunk, the switch at the far end will route the call according to the COR Time of Day Plan Number of the incoming trunk.

- Individual Attendant Access

  When an AAR/ARS call is made from an individual attendant (that is, not extending a call), the individual attendant's COR Time of Day Plan Number is used for routing the call.

- Recent Change History

  Changes made to Time of Day Routing Plan charts, routing plans, routing patterns, trunk groups, and CORs are recorded by the Recent Change History feature.

- Remote Access

  When an AAR or ARS call is made via Remote Access, the COR Time of Day Plan Number of the Barrier Code and/or Authorization Code that was entered is used for routing the call.

- CDR

  Normal CDR records are generated for AAR/ARS calls on trunks administered for CDR. However, information about the Time of Day Plan Number used to route the call is not provided.

- UDP

  Since the dialed digits of UDP calls are expanded into an RNX digit string, the AAR feature creates a potential for the use of different routing patterns. Once the call begins to be routed by AAR, the originating user's COR Time of Day Plan Number is used to route the call.

Administration

Time of Day Routing must be activated on the 'System Parameter-Customer Options' screen. It can then be administered by the System Manager. The following items require administration:

- AAR/ARS Partitioning must be administered as well as AAR and/or ARS.
- A different Digit Analysis Table must be administered for each Time of Day Plan Number.
- A Time of Day Routing Plan must be administered for each Time of Day Plan Number.
- A Time of Day Plan Number must be assigned to each COR table. Up to eight Time of Day Plan Numbers can be used.
- Immediate Manual Override and Clocked Manual Override buttons may be administered in order to manually override the Time of Day Routing Plan. These buttons can only be assigned to and used by attendants and voice terminal users with both console permission and a display.

Hardware and Software Requirements

No additional hardware or software is required.

Timed Reminder and Attendant Timers

Feature Availability

This feature is available with all Generic 3 releases.

Description

The Timed Reminder feature automatically alerts the attendant after an administered time interval for the following types of calls:

- Extended calls waiting to be answered or waiting to be connected to a busy single-line voice terminal
- One-party calls placed on hold on the console
- Incoming calls answered by a voice terminal user, but which are unanswered after being transferred.

The attendant can reenter the call and decide whether to terminate the call or permit the waiting to continue.

Like the numerous new trunk timers (these timers existed but were not administrable as they are now), some attendant timers were brought in from the international arena. Attendant timers are important for attendant-intensive settings, like those without Direct Inward Dialing. The attendant timers are:

- Unanswered DID Call Timer - routes the call to the administered ‘‘DID/TIE/ISDN Intercept Treatment’’ if the DID call goes unanswered for the length of the timer. This timer is available in G3i-Global, G3rV1, G3V2, and later releases.
- Attendant Return Call Timer — Unanswered calls extended and then released from an attendant console return to the same attendant position that released it if the attendant is available. If the attendant is not available, it goes to the attendant group queue. (This applies to G3i-Global, G3rV1, and all G3V2 and later releases.) The Attendant Return Call Timer is *not* set for calls extended from one attendant to another individual attendant.

  NOTE:
  For G3vsV1/G3sV1 and G3iV1, these calls return to the same attendant group, but not necessarily the same attendant.

- Attendant Timed Reminder of Held Call Timer — Allows the administration of an interval to be used when determining whether a held call has held too long. When this timer expires the held call alerts the attendant. The message hc appears on the attendant display. In G3iV1 and G3vsV1/G3sV1, the call alerts the attendant with a high pitched ring.

In G3i-Global and GrV1, the call alerts the attendant with primary alert. G3V2 and later releases can be administered to use either type of alert for held calls that time out.

- Attendant No Answer Timer (G3i-global, G3rV1, G3V2, and later releases) — Calls that terminate at an attendant console ring with primary alerting until this administrable timeout value is reached. When this timeout value is reached, the call rings with a secondary, higher pitch ring. If the Attendant No Answer Timer is disabled, the ringing pattern does not change over from the primary to the secondary pattern.
- Attendant Alerting Interval Timed Reminder (G3i-global, G3rV1, G3V2, and later releases) — This timer notifies, via secondary alerting, attendants in an attendant group of an unanswered call. The ‘‘Attendant Alerting Interval Timed Reminder’’ starts when a call reaches the ‘‘Attendant No Answer Timer’’ maximum value. (If the ‘‘Attendant No Answer Timer’’ is disabled, the alerting interval begins as soon as the call starts ringing at an attendant console.)

  If the alerting interval is reached before the call is answered, then the attendant console is placed into ‘‘position busy mode’’ and the call is forwarded to the attendant group. If the console where the alerting interval is reached is the last active day console, and night service is enabled, then the system is placed into night service.

  For G3V2 and later releases only, the alerting interval may be disabled. In this case, a call continues to ring at the original attendant’s extension until the caller hangs up or another feature disconnects the call (for example, the timeout limit is reached for unanswered DID calls during night service.)
- Line Intercept Tone Timer — the length of time of the line intercept. For example: LITT:10 seconds means the line intercept stops after 10 seconds.

Considerations

Timed Reminder informs the attendant that a call requires additional attention. After the attendant reconnects to the call, the user can either choose to try another extension number, hang up, or continue to wait. This personal attention can help establish rapport with clients and customers.

The Timed Reminder intervals for calls waiting for connection and for calls placed on hold are assigned separately. Each interval can be from 10 seconds to 17 minutes.

If a call has been routed to each attendant and it remains unanswered, the system is placed into night service. If this is happening more frequently than is optimal, the answering intervals may be lengthened by the system administrator.

Interactions

The following features interact with the Timed Reminder feature.

- Attendant Call Waiting

  An attendant-extended call to a busy single-line voice terminal returns to an attendant console if the Timed Reminder Interval expires before the call is answered, or redirects to coverage.

- Attendant No Answer

  When a “dial 0” call is made to the attendant, after the No Answer Timer expires, the secondary alerting tone operates as described in the Attendant No Answer section on the previous page. However, if a call alerts an attendant as a coverage call (unanswered station-to-station call with the “attd” (attendant) as an entry in the called station’s coverage path form) the secondary alerting tone does not operate.

- Call Coverage

  After a voice terminal user transfers a call to an on-premises voice terminal, the call, if unanswered at the expiration of the Timed Reminder Interval, redirects to an attendant console. Redirection to an attendant occurs even if the call has redirected via Call Coverage or Call Forwarding from the transferred-to voice terminal.

  An attendant-extended call redirects to coverage instead of returning to an attendant console, if the coverage criteria are met before the Timed Reminder Interval expires. However, unanswered calls return to a console at the expiration of the Timed Reminder Interval.

  When a “dial 0” call is made to the attendant, after the No Answer Timer expires, the secondary alerting tone operates as described in the Call Coverage section on the previous page. However, if a call alerts an attendant as a coverage call (unanswered station-to-station call with the “attd” (attendant) as an entry in the called station’s coverage path form) the secondary alerting tone does not operate.

- CAS

  If an attendant at the main location transfers a call from a branch location to an extension at the main location, the timed reminder does not apply and the call does not return to the attendant if unanswered.

- Transfer

  A call that is transferred times out and redirects to the attendant after an interval equal to the attendant return call timer.

Administration

Timed Reminder and Attendant Timers are administered on a per-system basis by the System Manager. The screen required is ‘Console Parameters’; the fields required to change the default timers are ‘No Answer Timeout’ and ‘Alerting’ under the heading ‘Incoming Call Reminders’ as well as the ‘Time Reminder on Hold’ and ‘Return Call Timeout’ fields.

Hardware and Software Requirements

No additional hardware or software is required.

Touch-Tone Dialing

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides quick and easy pushbutton dialing. Touch-Tone Dialing is always provided with the system. In addition to the **0** through **9** buttons, the ***** and **#** buttons have special functions, such as forming a part of a feature access code. A distinctive tone is generated when each button is pressed.

If a distant switching system can accept only dial pulse signals, the system converts the touch-tone signals to the required dial pulses for transmission to the distant end.

This feature is referred to as Dual-Tone-Multi-Frequency dialing (DTMF) outside the US.

Considerations

With Touch-Tone Dialing, users are more efficient when placing and handling calls.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Transfer

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows voice terminal users to transfer trunk or internal calls to other voice terminals within the system without attendant assistance.

Single-line voice terminal users momentarily flash the switchhook or press the Recall button, dial the desired extension number, and hang up.

Multiappearance voice terminal users press the Transfer button, dial the desired extension number, and press the Transfer button again.

Transfer is also known as Push Transfer. Please see the Pull Transfer feature for a description of a type of transfer that can be used by voice terminal users to ‘‘pull’’ a held call to their own extension.

Considerations

The Transfer feature provides a convenient way to connect a party with someone better qualified to handle the call. Attendant assistance is not required and the call does not have to be redialed.

Transferred trunk calls can be administered to receive either music or silence.

Multiappearance voice terminals must have an idle call appearance in order to transfer a call.

Interactions

- Transferred calls are eligible to be automatically answered via the Internal Automatic Answer (IAA) feature.
- When a multi-function station (BRI/Digital/Hybrid) dials sufficient digits to route a call, but could route differently if additional digits were dialed, the station will not recognize the Conference or Transfer buttons. The user must delay dialing for three seconds or dial a # to indicate the call can be routed based on the digits already dialed. The Conference or Transfer buttons are then recognized and the operation is completed by the switch.
- Station users *cannot* transfer the attendant.

- Integrated Services Digital Network Basic Rate Interface (ISDN BRI)

  When an ISDN-BRI station assigned with the “Select Last Used Appearance?” field of the “Station” form set to yes completes a transfer while off-hook using the handset, the user will be left hearing a dial tone on the last-used appearance, rather than the silence heard in the same situation by an user of other station types.

- Attendant Conferencing may not operate properly if the CO does not provide answer supervision. If your CO does not provide answer supervision, the “Answer Supervision Timeout” field must be set to a non-zero number and the “Receive Answer Supervision” must be set to “n.”

  If the CO *does* provide answer supervision, the “Answer Supervision Timeout” must be set to “0” and the “Receive Answer Supervision” must be set to “y.”

Administration

The Transfer feature is administered on a per-system basis by the System Manager. The only administration required is the treatment received on transferred trunk calls.

Hardware and Software Requirements

No additional hardware or software is required.

Transfer — Outgoing Trunk to Outgoing Trunk (with Security Measures)

Feature Availability

This feature is available with Generic 3rV1 and all Generic V2 and later releases.

Description

Outgoing trunk to outgoing trunk transfer (OTTOTT) is a feature that permits a controlling party, such as a station user or attendant, to initiate two or more outgoing trunk calls and then transfer the trunks together. The transfer operation removes the controlling party from the connection and conferences the outgoing trunks. Alternatively, the controlling party can establish a conference call with the outgoing trunks and then drop out of the conference, leaving only the outgoing trunks on the conference connection. This is a perilous enhancement of trunk to trunk transfer. OTTOTT allows calls to be established in which the only parties involved are external to the switch and are on outgoing trunks. This type of call can result in locked-up trunks, for example, trunks which cannot be disconnected except by busying out and releasing the affected trunk circuit. To clear the lock-up, a Service Technician must reseat the trunk board, or busy-out and release the affected trunk port.

At least one outgoing trunk must have been administered to support this type of transfer, and must have provided network answer supervision. The answer supervision test increases the probability, but does not guarantee, that a disconnect signal is received from the remote end of the trunk. To mitigate problems associated with its accidental use, this feature is only administrable on trunk groups on the “Trunk Group” form and is not a system option.

DCS networks provide a similar but more restrictive version of this feature, called ‘‘DCS Trunk Turn Around’’, which permits two outgoing trunks to be transferred together when the switch at the remote end of at least one of the trunks agrees to ‘‘turn around’’ the logical direction of the trunk. DCS trunk turn around is permitted, in general terms, when some other party involved in the call (at the remote switch) can provide disconnect supervision.

Without the use of OTTOTT, switches in a DCS network provide transfer between outgoing trunks through DCS Trunk Turnaround. Trunk Turnaround is attempted after a transfer or drop operation when no remaining parties can provide disconnect supervision. A message is sent to all remote switches that are connected to the local switch via an outgoing DCS trunk. If any of the signaled switches can provide disconnect supervision, they reverse their sense of the type of the trunk at their end from incoming to outgoing and respond with a positive acknowledgment. When the local switch receives the positive acknowledgment, it reverses the sense of its end of the trunk from outgoing to incoming. If no response is received by the local node within four seconds, the call is aborted.

Without OTTOTT or DCS, a conference involving two or more outgoing trunks is only permitted when at least one remaining conference party is an attendant, incoming trunk, or station.

Considerations

OTTOTT is not intended for use in DCS networks, since DCS Trunk Turnaround provides comparable capabilities in a much safer way. However, use of OTTOTT with DCS is not prohibited, and may be useful when one or more of the trunks go off the DCS network.

Security Measures

Administrators and System Managers are cautioned that this feature can be used to transfer an outside party to a trunk over which toll calls might be made. Since trunks have to be specifically administered for OTTOTT, the COR and FRL of that trunk group should be examined to determine if it is appropriate. OTTOTT is not a system-wide parameter. It is administered on a trunk group basis. There is however, a systemwide parameter which has to be enabled for this feature to work. If the administrator deems that the feature is not relevant to the business practices for the location the switch serves, the feature need not be enabled. Alternately, if a temporary need presents itself, the feature can be temporarily enabled and then turned off.

Interactions

The following are feature interactions with OTTOTT.

DCS Trunk Turnaround

OTTOTT increases the set of cases in which DCS Trunk Turnaround may be accepted. However, use of OTTOTT in combination with a DCS network is strongly discouraged. The following algorithm specifies the fate of a request for DCS Trunk Turnaround.

The checks are done in the order specified:

- If any of the parties on the call are receiving local dial, busy, intercept, or reorder tone, deny turnaround.
- If any remaining party is an answered station or attendant, then accept turnaround.
- The following applies if any remaining party is on an incoming trunk. For the purposes of this check, an outgoing DCS trunk that has previously been turned around (an odd number of times) via DCS trunk turnaround is

considered an incoming trunk with disconnect supervision. Similarly, an incoming DCS trunk that has previously been turned around (an odd number of times) is considered an outgoing trunk.

- If any remaining party is an outgoing trunk administered for OTTOTT operation that has received answer supervision, accept turnaround.
- If any remaining parties are outgoing DCS trunks, forward the turnaround request for each.
- Otherwise, deny turnaround.

Incoming Disconnect Supervision

Outside of the US, incoming disconnect supervision is a switch capability that restricts transfers or conferences for certain incoming trunks. In the United States, all incoming trunks are assumed to provide disconnect supervision. In some countries this assumption can not be made, so whether an incoming trunk provides disconnect supervision is administered for each trunk group.

Release Link Trunks (RLT)

This type of trunk is used by Centralized Attendant Service (CAS). An outgoing RLT at a remote PBX, the CAS Branch, is used to access an attendant at the CAS Main. The attendant (at the Main) can transfer the incoming caller to a destination (station or trunk) at the Branch. The RLT is typically used only for a short period of time and is usually idled after the transfer is established.

A station at a branch can transfer an outgoing trunk to the CAS attendant (at the Main). This transfer could be viewed as an outgoing-trunk-to-outgoing-trunk transfer (the attendant is accessed via an outgoing RLT), although it is not blocked in G3vsV1/G3sV1 and G3iV1. Since administering outgoing disconnect supervision for RLT trunks provides no additional capability, this administration is not provided for RLT trunks.

Restriction

Restrictions on the transferring party may block a transfer or drop operation even when Outgoing Disconnect Supervision is provided.

Trunk-to-Trunk Transfer

If this feature-related system parameter is set to ‘‘restricted,’’ all trunk-to-trunk transfer/release/drop operations for public trunks (CO, CPE, CAS, DID, DIOD, FX, and WATS) have calls terminated or receive denial. If the parameter is set to ‘‘none,’’ all trunk-to-trunk transfers (except CAS and DCS) have calls terminated or receive denial.

Hence, this option must be set to ‘‘y’’ to enable OTTOTT operation for these types of trunks. The number of public network trunks allowed on a conference call is an administrable system parameter. This number defaults to ‘‘1,’’ so if OTTOTT is being used to connect two or more public network trunks this limit must be increased.

PCOL Interactions

Transfer of personal CO lines (PCOLs) are not subject to the normal restrictions applied to transfer of other trunks. These transfers are allowed since the PCOL appearance remains on one or more stations as a feature button. The user must be aware that the DROP button cannot be used to disconnect the transferred-to party from the call. Hence, if an outgoing PCOL is transferred to an outgoing trunk and if neither of the trunks can supply a disconnect signal, then the two trunks lock up.

Administration

OTTOTT is enabled for all trunks in a particular trunk group by setting the ‘‘Disconnect Supervision - Out?’’ field on the “Trunk Group Administration” form to the value ‘‘y.’’ This field specifies whether trunks in this trunk group provide disconnect supervision when used on an outgoing basis.

OTTOTT is allowed if the system-wide Trunk-to-Trunk Transfer feature is assigned on the ‘‘Feature-Related System Parameters’’ forms and at least one of the outgoing trunks remaining on the call has its ‘‘Disconnect Supervision - Out’’ field set to ‘‘y.’’

This field is set independently of the trunk-to-trunk transfer system parameter, although both options must be set appropriately if outgoing trunk to outgoing trunk transfers are desired. This field appears on the “Trunk Group” form for the following trunk group types:

- Access
- Aplt
- Co (also for G3r)
- Dmi-bos (DID for G3r)
- Fx (also for G3r)
- Isdn-pri (also for G3r)
- Tandem
- Tie
- WATS

The field appears on the form, but cannot be changed when the “Direction” field is set to `incoming`. If the field has a value of y when an incoming only trunk group is added or changed, the field value is changed (by the system) to n when the form is submitted, because the next time the trunk group is displayed the field value is n.

The allowable values are:

n(o) y(es)

The default is n.

The following warning message is generated whenever this field is changed to the value y.

WARNING:
Setting this field to ''y'' may result in trunk lock-ups.

Hardware/Software Requirements

OTTOTT can be activated for trunk groups with trunks assigned to the following trunk boards:

- Central Office Trunk board (TN747B)
- Analog Tie trunk Board (TN760C)
- DS1 board (TN722B & TN767 & TN464C [Universal DS1])

Trunk Flash

Feature Availability

This feature is available with all Generic 3 releases. However, for G3V2 and later releases, this feature encompasses both the G3V1 ‘‘Trunk Flash’’ and ‘‘PCORR’’ features.

Description

Trunk Flash enables multi-function voice terminals to access ‘‘central office customized services’’ that are provided by the far-end/CO located directly behind the DEFINITY system. CO customized services are electronic features (such as conference and transfer) that are accessed by a sequence of flash signals and dialing from the DEFINITY system station on an active trunk call. The Trunk Flash feature can help to reduce the number of trunk lines connected to the DEFINITY switch by:

- Performing trunk-to-trunk call transfers at the far-end/CO, which eliminates the use of a second trunk line for the duration of the call and frees the original trunk line for the duration of the call.
- Performing a conference call with a second outside call party, which eliminates the need for a second trunk line for the duration of the call.

G3V2 supports Trunk Flash activation via Feature Access Code (FAC) and from CAS and non-CAS attendant terminals.

NOTE:
Some analog DTMF telephone sets used in Italy and the United Kingdom are equipped with a ‘‘FLASH’’ button that, when pressed, generates a rotary digit ‘‘1.’’ When an analog station which is administered as a DTMF station (for example, as a 2500 or 71nn-type station) transmits a rotary digit ‘‘1.’’ G3i-Global, G3V2, and later releases treat the digit 1 signal as a recall signal from the station set to the PBX. This is called ‘‘Digit 1 as Flash’’ and is not supported in G3vsV1/G3sV1, G3iV1, or G3rV1.

Considerations

Generic 3 supports the Trunk Flash signal for incoming, outgoing, or two-way call directions on selected two-wire analog (ground-start or loop-start) or digital (DS1) trunks. For G3i-Global, G3V2, and later releases, trunk hunting enhancements have been added which affect trunks that are loop-start and help handle glare problems.

Access to this feature is restricted to trunk Group Types of `co`, `fx`, and `wats` with the “Trunk Flash” field enabled.

In G3V1:

- — A Trunk Flash button can be assigned to multifunction stations and CAS attendants. For stations, this button activates the Trunk Flash feature; for CAS attendants, this button controls certain CAS features via RLT trunks.
- — The Trunk Flash feature is unavailable to analog stations and attendant consoles and can not be used on PCOLs.

In G3V2 and later releases:

- — A Trunk Flash button can be assigned on CAS attendant consoles, non-CAS attendant consoles, and multifunction stations. For CAS attendants, use of this button limited to certain CAS features via RLT trunks. For multifunction and non-CAS attendant consoles, this button is used for the Trunk Flash feature.
- — FAC activation of the trunk flash feature is allowed.

The Flash button is used by the Trunk Flash and CAS features.

Generic 3 features (such as internal conference call, transfer, and call park) may be combined with custom services (that is, CO-based features that are activated/controlled by sending a ‘‘flash’’ signal over the trunk to the CO). However, mixing Generic 3 features with custom services causes complications for the user when tracking a call. DEFINITY systems cannot give the local station user status information on the custom services.

The Trunk Flash feature may only be accessed if the call has only one trunk, the trunk must be outgoing from the PBX’s perspective, and the trunk group of that trunk has ‘‘Trunk Flash’’ enabled. The Trunk Flash feature is disabled when the call involves more than one trunk, even if all the trunks have ‘‘Trunk Flash’’ enabled.

NOTE:
The facility connecting a PBX to the CO is referred to, from the perspective of the PBX, as a trunk and, from the perspective of the CO, as a line.

Any PBX station with a Flash button may access the ‘‘Trunk Flash’’ feature. There may be up to five PBX stations involved in a PBX conference call with the trunk line party. However, to access the Trunk Flash feature, at least one of the PBX stations must have a Flash button.

In a call involving more than one PBX station, one station may press the Flash button, and another station may dial the phone number. The station that dials the phone number is not required to have a Flash button.

There must be no other trunk connection between the PBX that the station is connected to, and the far-end/CO supplying the custom services. If the call connection passes over a ‘‘tie’’ trunk, the station does not have a direct connection to the far-end/CO and, as a result, does not have access to the far-end/CO custom services.

If the far-end/CO does not support custom services, the call may be dropped by the far-end/CO on sending the flash signal or the signal may be ignored and a ‘‘click click’’ sound is heard.

Interactions

The Trunk Flash feature may be combined with other DEFINITY switch features.

Calls made after the flash are not recorded in CDR records.

Administration

The System Manager must perform the following tasks:

- Administer a “Station” form for each voice terminal and attendant console that is authorized for Trunk Flash. Each authorized voice terminal must be assigned a Flash button.
- Administer the “Trunk Flash?” field in the appropriate “Trunk Group” form(s) for CO, FX, and/or WATS group types.

Also, to access Trunk Flash via the Feature Access Codes (FAC), the “Flash Access’’ field on the ‘‘Feature Access’’ form must be administered.

Hardware and Software Requirements

No additional hardware or software is required.

Trunk Group Busy/Warning Indicators to Attendant

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides the attendant with a visual indication that the number of busy trunks in a group has reached an administered level. A visual indication is also provided when all trunks in a group are busy.

The two lamps which provide the visual indications are as follows:

- Warn Lamp

  Located on Trunk Hundreds Select buttons that have three lamps. The Warn lamp lights when a preset number (warning threshold) of trunks are busy in the associated trunk group.

- Busy Lamp

  Located at each of the 12 Fixed Trunk Hundreds Select buttons and on each feature button administered as a Trunk Hundreds Select button. The Busy lamp lights when all trunks in the associated trunk group are busy.

Considerations

The Trunk Group Busy and the Trunk Group Warning Indicators are particularly useful when the Attendant Control of Trunk Group Access feature is provided. The indicators show the attendant that control of access to trunk groups is necessary.

Interactions

If Trunk Hundreds Select buttons are assigned for Loudspeaker Paging Access zones, Trunk Group Busy Indicators provides a visual indication of the busy or idle status of the zones.

Administration

This feature is administered by the System Manager. The following items require administration:

- Trunk Hundreds Select buttons (per attendant console)
- Warning threshold (per trunk group)

Hardware and Software Requirements

No additional hardware or software is required.

Trunk Identification By Attendant

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows an attendant or display-equipped voice terminal user to identify a specific trunk being used on a call. This capability is provided by assigning a Trunk ID button to the attendant console or voice terminal.

The Trunk Identification By Attendant feature can be used when a user is on an established call of one of the following types:

- An incoming trunk call
- An outgoing trunk call
- A transferred or conferenced call involving a trunk
- A trunk-to-trunk call

In addition to its use during an established call, the Trunk ID button can be used while a trunk is being seized, while digits are being outpulsed on a trunk, or during intervals between digit outpulsing.

When a user is connected to a trunk, as described above, and presses the Trunk ID button, the identification of the trunk is displayed on the 40-character alphanumeric display. The trunk identification consists of the trunk access code (two-digit) for that trunk group and the trunk group member number (two-digit).

The trunk identification displayed depends on the type of call in process. If the call is incoming, the incoming trunk identification is displayed. If the call is outgoing, the outgoing trunk identification is displayed. If the call is trunk-to-trunk, the identification displayed is of the last trunk added to the call.

Considerations

Trunk Identification By Attendant is useful whenever it is necessary to identify a particular trunk being used. The feature is particularly useful for identification of a faulty trunk. That trunk can then be removed from service and the problem quickly corrected.

A maximum of one Trunk ID button is allowed per each attendant console and voice terminal with a display.

The Trunk Identification By Attendant feature is denied if there are more than two trunks on the call.

The Trunk Identification By Attendant feature is denied if there are exactly two trunks on the call, and the station pressing the Trunk ID button is not the controlling party.

In the case of a conference resulting from an incoming call followed by an outgoing call, the last trunk added to the conference is the incoming one.

Interactions

The following features interact with the Trunk Identification By Attendant feature.

- Busy Verification

  A trunk being busy-verified can be identified.

- Attendant Display and Voice Terminal Display

  Any action by the user or the system which changes the display removes the trunk identification currently displayed. The lamp associated with the Trunk ID button remains lighted as long as the call on which the button was used remains active. While the lamp is lighted, the user can use the associated button to redisplay the trunk identification.

  If the Trunk ID button is pressed during a call origination (before all digits have been dialed), the trunk identification appears. On a voice terminal display, any subsequently dialed digits are not displayed. On an attendant display, the subsequently dialed digits overwrite other digits on the display.

- Hold

  A trunk held by a user cannot be identified.

Administration

Trunk Identification By Attendant is assigned by the System Manager on a per-voice terminal and per attendant console basis. The only administration required is the assignment of a Trunk ID button.

Hardware and Software Requirements

No additional hardware or software is required.

Trunk-to-Trunk Transfer

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows the attendant or voice terminal user to connect an incoming trunk call to an outgoing trunk.

Considerations

Trunk-to-Trunk Transfer is particularly useful when a caller outside the system calls a user or attendant and requests a transfer to another outside number. For example, a worker, away on business, can call in and have the call transferred elsewhere.

Transferred trunk calls can be administered to receive either music or silence.

Some central office (CO) trunks do not signal the PBX when the CO user disconnects from the call. The system assures that incoming CO trunks without Disconnect Supervision are not transferred to outgoing trunks, or other incoming CO trunks without Disconnect Supervision.

An attendant-assisted call connecting an outgoing trunk or incoming trunk without Disconnect Supervision to an outgoing trunk must be held on the console. The system does not allow the attendant to release such a call. The attendant can, however, use the Forced Release button and disconnect all parties associated with the call.

If a voice terminal user has connected two outgoing trunks or an outgoing call and an incoming call without Disconnect Supervision, the user must remain on the call. Otherwise, the call is dropped. An incoming trunk with Disconnect Supervision can be connected to an outgoing trunk without the user remaining on the call. An incoming trunk can also be connected to another incoming trunk without the user remaining on the call if one of the incoming trunks has Disconnect Supervision.

G3rV1 and G3V2 and later releases support ‘‘Outgoing-Trunk-to-Outgoing-Trunk’’ transfer, which permits a station user or an attendant to initiate two or more outgoing trunk calls. If the Trunk-to-Trunk feature is assigned and one of the trunks is administered as ‘‘y’’ on the “Trunk Group” form ‘‘Disconnect Supervision-Out?” field, the station user or attendant can transfer the trunks together. Transferring in this way removes the controlling party from the connection and conferences the two outgoing trunks together.

The Trunk-to-Trunk Transfer option on the “Feature Related System Parameters” form has no affect on tie trunks. Restricted Trunk-to-Trunk Transfer only affects calls where both trunks are CAS, CO, DIOD, FX, WATS, DID, or CPE.

Interactions

The Attendant Lockout feature does not function on Trunk-to-Trunk Transfer.

Administration

Trunk-to-Trunk Transfer is administered on a per-system basis by the System Manager. Trunk-to-Trunk Transfer is administered on the “Feature-Related System Parameters” form. If the “Trunk-to-Trunk Transfer” field contains `all`, all trunks are transferred. If the field contains `restricted`, public trunks (CAS, CO, DIOD, FX, DID, WATS, and CPE) are not transferred. If the field contains n, all trunks are restricted. Restriction means that any transfer/release/drop operation has calls dropped or receives denial.

To enable ‘‘Outgoing-Trunk to Outgoing-Trunk Transfer,’’ set the ‘‘Disconnect Supervision-Out?’’ field to `y`.

NOTE:
Setting the value of this field to `y` may result in trunk lock-ups.

Hardware and Software Requirements

No additional hardware or software is required.

Uniform Dial Plan (UDP)

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides a common 4- or 5-digit dial plan (specified by Dialplan administration) that can be shared among a group of switches. Interswitch dialing and intraswitch dialing both require 4- or 5-digit dialing. The UDP is used with ETN, Main/Satellite/Tributary, and DCS configurations. Additionally, UDP can be used alone to provide uniform 4- or 5-digit dialing between two or more private switching systems without ETN, Main/Satellite/Tributary, or DCS configurations.

For G3i, in a UDP, the first one, two, three, or four digits of the 4- or 5-digit extension number make up a PBX code which determines the switch to which a call is directed. When a UDP is administered, a list of PBX codes is assigned to each switch. A UDP can have as many as 240 PBX codes.

For G3r, each PBX code is assigned a private network location code (RNX) or node number. The RNX of a PBX in a UDP is the equivalent of an office code of a central office in a public network. It is this RNX that is actually used to determine how a UDP call is routed. Each PBX code is also administered as either local or remote to the switch

UDP routes calls off the local PBX by converting the extension number into a type of private-network number with 7 digits. Such a number is formed by prepending a 3-digit code (of the form XXX) to the (last) four digits of the extension number. Three types of conversion are supported: UDPCode, AARCode, and ENPNode. For UDPCode and AARCode, XXX is a 3-digit private-network location code, and the result is analyzed and routed via AAR. UDPCode conversion prohibits digit conversion via AAR; AAR Code conversion permits it, just as if the user had dialed the AAR number instead of the extension. The third type of conversion, ENPCode, XXX is called an ‘‘ENP code,’’ and is not used for routing; instead, node number routing is used. The ENP code is chosen based on the first one or two digits of the dialed extension. The ENP code can be independent of location since it is not used for routing.

UDP conversion can be specified for individual extensions, or groups of extensions sharing the same leading digits, via the “UDP” form and ‘‘extension codes.’’ An extension code is just the desired leading digits of an extension followed by ‘x’ (wild cards). For example, ‘‘123xx’’ is the extension code for all 100 possible extensions beginning with ‘‘123’’ (‘‘12345’’ is an extension code specifying only one extension). Each extension code can be assigned to one of six possible treatments:

- UDPCode — Conversion to AAR with given location code, further conversion suppressed
- AARCode — Conversion to AAR with given location code, further conversion allowed
- ENPCode — Conversion to private network number (via “ENP” form), route to given node number routing
- TempOOS — Temporarily out of service, give reorder
- Intrcpt — Invalid, give intercept treatment
- blank — This extension code does not apply

Whenever a UDP is used to route a call, the number it outputs is in the form of RNX plus XXXX. This always needs to be taken into account so that the correct digit deletion and/or insertion can be specified within the routing pattern so that the receiving switch gets digits in the format it expects.

To understand the function of a UDP, look at the next figure. In this figure, a five-digit UDP is used in an ETN. Three switches are included in the UDP. Each switch has an assigned RNX and a prefix code (discussed later). Each switch has also been assigned a list of PBX codes with an RNX assigned to each PBX code. Assume that the following PBX codes and associated RNXs have been assigned:

PBX CODE	RNX
41	224
51	223
52	223
60	222
61	222

UDP allows access, via an extension number to extensions on any switch in a UDP network. This is accomplished by converting the extension to a 7-digit private network number, in UDP this is XXX-XXXX and is routed on the location code. In ENP, this is XXX-XXXX and is routed via a location independent destination node number.

If the user at extension 41000 wants to call extension 61234, he or she has two choices of how to do this. The user at extension 41000 can either dial ‘‘61234’’ or, if AAR is provided, the user can dial the AAR access code followed by ‘‘222-1234.’’ If 61234 is dialed, the system recognizes 61 as a PBX code, determines the associated RNX (222), and uses AAR to route the call to 222-1234. If the AAR access code and 222-1234 are dialed, the system finds the routing pattern for RNX 222 and routes the call to the PBX associated with that RNX.

If AAR is active on the network, then subnet trunking can be used to insert the AAR feature access code on the originating end, or digit insertion may be used to insert it on the incoming end. On the destination switch, AAR conversion is administered to convert 222 with 7 digits to an extension by deleting 3 digits and inserting a ‘6’. If AAR is not active, then Subnet Trunking must be used on the source switch to delete the XXX and insert the digit ‘6’ at the beginning of the extension number so that the receiving switch can continue to route correctly.

If the user at extension 51234 on Switch C dials extension 61234, the call must first go through Switch A before proceeding to Switch B. When 61234 is dialed, the system recognizes 61 as a PBX code, determines the associated RNX (222), and uses AAR to route the call. The call will first be routed to Switch A, where Switch A will then recognize the RNX 222 as a remote switch and route the call to Switch B and extension 61234. This same type of call routing occurs when an extension at Switch B calls an extension at Switch C.

If extension 61234 on Switch B calls extension 61235, the system recognizes 61 as a PBX code with an RNX that is local to the switch, and the call is routed directly to extension 61235. See Figure 3-30 below.

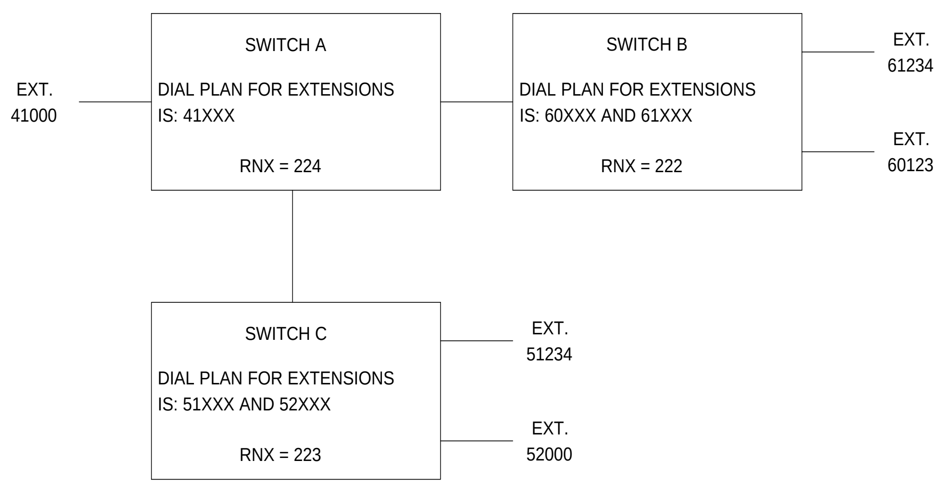


Figure 3-30. UDP Example

For G3vsV1/G3sV1 and G3iV1 only, once a certain PBX code is assigned to a switch, no other switch within the UDP can use that same PBX code.

For G3vsV1/G3sV1 and G3iV1, when a user at a switch that is included in a UDP dials an extension, the system checks to see if the first digit(s) of the extension is an assigned PBX code. If the first digit(s) is not an assigned PBX code, the call is routed via the regular, non-UDP, dial plan. If the first digit(s) is an assigned PBX code, the system translates the PBX code into the administered RNX. (This is in reverse for G3r.) If the PBX code indicates that the called extension is on the same switch, the call is routed to the local extension. If the PBX code indicates that the called extension is at another switch within the UDP, AAR uses the associated RNX to route the call to the correct switch within the private network. (The necessary subset of AAR is provided with the UDP software.) If the PBX code is not assigned a corresponding RNX, the user receives intercept treatment. For G3rV1, G3V2, and later releases, when a user at a switch that is included in a UDP dials an extension, the system first checks to see if the extension is assigned to a local station on that switch. If so, the call is routed to the station, and UDP is not invoked. If the extension is not found locally, the system checks to see if the extension matches an assigned extension code. If the extension matches an assigned extension code, the system performs the specified conversion into a private network number and routes the call as specified. (The necessary subset of AAR is provided with the UDP software.) If more than one extension code matches, the ‘‘best’’ match (most explicit digits) is used. For example, ‘‘1234x’’ will be chosen over ‘‘123xx’’ if ‘‘12345’’ is dialed (but ‘‘123xx’’ will be chosen if ‘‘12355’’ is dialed). If no matching extension code is found, the user receives intercept treatment, or, if enabled, the call will route via ETA.

The UDP allows a user to call other extensions within a private network by dialing a 4- or 5-digit number. However, if AAR is provided, a user can also call other extensions by dialing the AAR access code, the RNX of the switch to be called, and then the desired extension number. For example, if a user on switch A wants to call extension 3797 on switch B, the user can either dial 3797 or dial the AAR access code followed by RNX 3797. When the user dials RNX 3797, AAR will route the call to the correct switch and extension.

The following applies only if AAR is not implemented. If a five-digit UDP is used, the routing pattern of each RNX must be administered so that it inserts a prefix digit at the beginning of the extension. For example, as shown in the earlier figure, if a user on switch A wants to call extension 61234 on switch B, the user could dial 222-3797. Then, the routing pattern assigned to the dialed RNX would insert the prefix 6 at the beginning of the extension and route the call to the desired extension.

Considerations

The UDP feature enables a terminal user at any switch to call any other terminal on any switch in the UDP complex, using only the 4- or 5-digit extension number.

Since extensions beginning with 0 may be routed by some switches to the attendant in a network environment, administrators are discouraged from using this number as the leading digit when assigning extensions.

When calling an extension on another switch, there is a slight delay before call progress tones are applied. This delay is due to the trunk signaling necessary to complete the call to the remote switch.

For G3i only, it is possible that the first one, two, three, or four digits (PBX code) of the 4- or 5-digit extension number could be the same as a local extension number. In this case, the UDP PBX code overrides the extension number at the local switch. Problems can be avoided by assuring that the PBX code does not match an extension number.

The list of PBX codes for a switch can contain PBX codes of varying lengths. For example, a switch may be assigned both 2-digit and 3-digit PBX codes. It is also possible that one PBX code may be included in another. For example, a switch may be assigned both 61 and 612 as PBX codes. In this case, all calls beginning with 61, except those beginning with 612, are routed according to the RNX assigned to PBX code 61. Calls beginning with 612 are routed according to the RNX for PBX code 612. (The system always looks at the first four digits before routing the call.)

For G3r, if AAR is active in the system, tandem tie trunks may be used to send a traveling class mark, or FRL, along with the private network number. UDPCode and AARCode conversions use the FRL assigned to the caller. ENPNode conversion always raise the FRL to the maximum (7). When a user attempts to make a call where the originating user's FRL is insufficient to access the facilities, the user is denied access and is not prompted for an authorization code even if authorization codes are enabled and administered. Instead, denial is automatically administered to the originating user (typically an intercept tone). If AAR is not enabled in the system, tandem tie trunks should never be used to transport UDP numbers (the TCM will not be recognized as such following the extension received at the receiving switch).

Interactions

The following features interact with the UDP feature.

- AAR

  After the system determines the RNX of the switch being called, AAR routes the call to the correct switch. The required subset of AAR is provided with the UDP software. If the AAR feature is provided in addition to the UDP, then the seven-digit AAR number will provide the exact same routing as the UDP.

- DID

  DID calls to five-digit UDP extension numbers require that the DID trunk group insert enough digits to make a five-digit extension number.

- Dial Plan

  All of the extension numbers on a switch are not necessarily part of the UDP. Any that do not belong to the UDP are handled by a regular, non-UDP dial plan associated with the local switch. In G3i only, when administering the dial plan and designating a group of extensions as UDP non-local, you can specify on the "Dialplan" form whether you want to search for local extensions first or last. This allows some flexibility in the changing of extensions from local to non-local. However, after the dial plan is changed to make an extension UDP, nothing can be administered with these extensions on the local switch.

- DCS

  UDP is required when DCS is provided. The necessary UDP software is provided with the DCS software.

- ENP

  For G3rV1, G3V2, and later releases, if an extension code is administered to use ENP node routing, ENP routes the call to the correct switch. If the AAR feature is provided in addition to UDP, then the seven-digit AAR number provides the exact same routing a s the UDP (that is, via ENP).

Administration

The UDP is administered by the System Manager. The following items require administration:

- Whether UDP has 4- or 5-digit extension numbers PBX Codes (expands first one, two, three, or four digits of dialed extension to an RNX) (G3vsV1/G3sV1, and G3iV1)
- AAR Analysis Table (used by AAR to route calls to the correct switch)
- Routing Patterns
- Node Number Routing (used to route ENP calls) (G3r)
- Extension Codes and type(s) of conversion (G3r, G3V2, and later releases)
- AAR Digit Conversion (to define conversions for AAR Code extensions or to define home location codes) (G3rV1, G3V2, and later releases)

NOTE:
If the user changes the “Uniform Dialing Plan” field value from a ‘‘y’’ to a ‘‘n,’’ then a warning message is generated to inform them that this action causes all UDP extension codes to be lost. The same warning message is applied if the “Plan Length” field value is changed from a ‘‘4’’ to a ‘‘5’’ or from a ‘‘5’’ to a ‘‘4.’’

Hardware and Software Requirements

A Processor Interface circuit pack is required for DCS applications. DCS or UDP software is required.

VDN of Origin Announcement (VOA)

Feature Availability

This feature is available with Generic 3 Version 3 and later releases. The Call Vectoring and VDN of Origin Announcements features must be optioned.

Feature History

The VDN of Origin Announcements feature was first available in the System 85 R2V4/Generic 2 releases. In these releases, the feature was known as ‘‘City-of-Origin’’ and ‘‘Queue-of-Origin’’ announcements.

Description

VDN of Origin Announcements provide a short voice message when an agent answers a VDN call. The message informs the agent of the city of origin of the caller or of the service requested by the caller, based on the VDN used to process the call. The message begins as soon as the agent answers the call. While the agent can hear both the message and the caller, the caller is not able to hear the message.

If the agent’s terminal is connected to a DEFINITY G3V3, the caller can hear the agent if the agent speaks during the message. If the agent’s voice terminal is in Auto Answer mode, when the call reaches the voice terminal, the system applies a zip tone to the call. The playback of the announcement begins when the zip tone ends. The agent does not need to wait for the message to complete before speaking to the caller.

If the agent’s voice terminal is connected to a DEFINITY G3V4 (and later releases), the caller cannot hear the agent if the agent speaks during the message since the agent is not connected to the caller until after the message is finished. If the agent’s voice terminal is configured in Auto Answer mode, the agent hears a zip tone before the message. The playback of the message begins when the zip tone ends.

With G3V4 and later releases it is possible to administer the system so that the agent also hears zip tone after the VOA completes. In this way, the agent knows that the announcement is completed and a caller is now connected. The agent must wait until after the second zip tone before speaking to the caller.

For G3V4 and later releases, if the agent is logged in at a multi-line terminal, the call-appearance button for an incoming call continues to flash until after the VDN of Origin message is completed. The agent can shorten the VOA message playback for the incoming call by pressing the flashing call-appearance button while the message is being heard. Also, if the agent is logged in at a terminal with an alphanumeric display, no incoming call information is displayed until after the VOA message is completed.

After listening to the announcement, the agent can appropriately respond to the caller. For example, if a user has two 800 numbers, one for placing orders and one for technical support, they can administer two VDNs to route the calls to the same agents. When a call enters the system and is sent to a VDN, the VDN can have a VDN of Origin announcement assigned to it. The VDN directs the call to a vector, and as a result of vector processing, the call can wait in one or more queues until an agent is available. When an agent answers a call, they hear the appropriate recording for the VDN (for example, ‘‘new order’’ or ‘‘tech help’’). The agent then greets the caller.

If a non-ACD agent is in Automatic Answer mode, when a call reaches the voice terminal, the system applies a call ID tone. The message begins when the call ID tone ends. If the agent is in Manual Answer mode, when a call reaches the voice terminal, the system applies a ringing signal. The message begins when the agent answers the call. Note that in each case, the message begins at the same time the agent is connected to the call.

If an agent needs to hear a message again, and if the agent’s voice terminal is administered with a VOA Repeat feature button, the agent can press the VOA Repeat button. The VOA Repeat button lamp stays lit while the announcement plays. The VOA Repeat button lamp also remains lit if the system needs to queue the request for the announcement. The lamp lights with a broken flutter if the agent presses the VOA Repeat button but the resources to play the announcement are not available within two to three seconds. For G3V3, if the agent presses the VOA Repeat button when the lamp is lit, the lamp flutters to indicate that the subsequent request has been ignored. For G3V4, if the agent presses the VOA Repeat button when the lamp is lit, the playback of the announcement is immediately stopped. If there is no VOA assigned, the VOA Repeat button lamp flutters immediately when the call is answered or when a user presses the button.

Announcements are assigned on a per-VDN basis. However, the VDN of Origin announcement applies to a class of restriction (COR). This means you must administer a COR for all agents who will be expecting or are expected to receive the VDN of Origin announcements. Note that in G3V3 and later releases, the COR feature is upgraded to enable or disable an agent’s ability to receive VDN of Origin announcements.

You can set up VDN of Origin announcements in four different ways. You can use any or all of these arrangements, which are:

- The agent can hear a unique announcement based on the dialed number identification service (DNIS) received from the service end office or carrier switch. Each DNIS is assigned as the VDN of a vector. The announcement associated with the VDN announces the services associated with the DNIS.

  NOTE:
  The specific announcement associated with the current VDN only plays if the "VDN Override" field of the previous VDN is marked ''y'' (yes). If the "VDN Override" field associated with the previous VDN is set to ''n'' (no), the announcement of the previous VDN does not play.

- Vector steps can route (using a route-to step) the call to a VDN. Or, a response to an integrated prompting or **converse-on** step can route the call to a VDN. The announcement associated with the VDN announces the service the caller requested or can announce a condition that caused the call to **route-to** the VDN.
- An incoming call routes to a voice response system, possibly through a vector. Voice prompting directs the caller to enter a touchtone response, and the call is then routed to a specific VDN based on the caller's response. The announcement associated with the VDN indicates the service selected by the caller.
- If the agent requires a call's city of origin, the trunk group is assigned to a particular VDN. The announcement associated with the VDN provides the location of the origin of the trunk group. Subsequent VDNs can be used to handle the call, or multiple VDNs can be assigned to a single vector.

NOTE:
VDN Override applies to VOA in the same way that VDN Override applies to display information. If a VDN with a VOA has VDN Override enabled, the system overrides the original VOA with VOAs in subsequent VDNs to which the call may be routed.

Considerations

Because of the nature of the VDN of Origin announcements (the customer can be waiting while the message plays), the messages should be kept very brief, no more than 1.5 seconds in length.

Because the announcements should be very brief, agents receiving the messages should be using a speakerphone or headset. This way, they do not miss the message while they are picking up the handset. If the agents cannot use a speakerphone or headset, you can administer the voice terminals with a VOA Repeat button that allows them to repeat the message.

The agent who receives the announcement and takes the call must be on the same switch as the VDN of Origin Announcement.

The system maintains a separate logical queue for the VDN of Origin Announcements when you use the TN750 circuit board for integrated announcements. If, because of traffic considerations or inoperative equipment, the VDN of Origin Announcement cannot be given to the agent within 0 to 1 second, the system stops an attempt to provide the announcement.

NOTE:
VOA announcements receive higher priority than other announcements on the TN750. A burst of VDN of Origin Announcements can cause other announcements (such as Forced First Announcements) to be delayed. Therefore, it is recommended that non-VDN of Origin Announcements be recorded as auxiliary or analog.

Auxiliary announcements are connected for a duration of 1 to 2 seconds on a barge-in basis, immediately after the agent answers (or is assigned the call for auto-answer) and the incoming call is extended to the agent. Integrated and non-barge-in auxiliary announcements are connected for the duration of the announcement. This switch does not ensure that the integrated announcement is shorter than the allowed playback time.

The VOA feature supports Auxiliary Trunk (aux-trunk) announcement types. You can use aux-trunk with barge-in, queue, or without queue. For aux-trunk with or without queue, when a VDN call needs the VDN of Origin announcement and the trunk is idle, the call seizes the trunk to start the announcement and the system plays the entire announcement (not just 1 to 2 seconds). However, if the announcement is busy and if aux-trunk has barge-in, the call does not queue but bridges on to the announcement for the 1 to 2 seconds needed. When the announcement reaches its end, it causes the trunk to be released along with the listeners; the next call needing the announcement starts the process over again. For this reason, your aux-trunk announcements should consist of one short announcement repeating over and over for the length of the full announcement time. For example, you might want to record ‘‘New Order’’ over and over as many times as possible, so that when a call bridges to the announcement, the agent hears ‘‘New Order’’ no matter where the agent bridges into the announcement.

Also, if you use aux-trunk or integrated announcement without queue, and a port is busy when a VDN calls comes in, the system cannot play an announcement. If you use aux-trunk or integrated announcement with queue, the system plays the current announcement for an agent and then connects the next agent in the queue.

Interactions

- ASAI Adjunct Routing

  If a vector step calls for Adjunct Routing, the VDN of Origin announcement is played for the agent to which the call is routed.

- Agent Call Handling

  Answering Options

  — Automatic Answer

    An ACD agent set up for Automatic Answer receives zip tone instead of ringing. The VDN of Origin Announcement begins immediately after the zip tone ends. A non-ACD agent set up for Automatic Answer hears an incoming call ID tone when a call comes in. Through the use of a route-to vector step, non-ACD agents can receive a VDN of Origin Announcement. The announcement begins when the incoming call ID tone ends. For G3V4 and later releases, the agent also hears a second zip tone after the announcement indicating connection of caller to agent.

  — Manual Answer

    If an agent is set up for Manual Answer and is not using a headset or speakerphone, that agent can miss the message while picking up the call. Therefore, it is a good idea to provide a headset or speakerphone to agents on Manual Answer.

- Attendant Console

  An attendant's voice terminal, as all other voice terminals, may have only one VOA Repeat button. An attendant can use the VOA Repeat to replay the message for the active call.

- Auto-Available Splits (AAS)

  AAS provides a way for members of an ACD split to be in a continuous Auto-In work mode. Agent positions with this feature are automatically put in the Auto-In mode after a system restart. This feature is intended to be used for splits containing only nonhuman adjuncts (such as AUDIX or Conversant), however, VDN of Origin announcements can be directed to Auto-Available splits.

- Call Forwarding

  VDN of Origin announcements apply to forwarded calls, including those forwarded to a hunt group. The answering station must be on the same switch. If a VDN of Origin announcement is forwarded to another extension, the message is played only if the destination extension is administered with a COR that allows VDN of Origin announcements.

- Call Hold

  An agent cannot use the VOA Repeat button if the agent's calls are all on hold. The VOA Repeat button only applies to active calls.

- Call Pickup

  Call Pickup allows an agent to pick up a ringing call on another extension. If the pick-up phone has COR permissions for VOA, the person picking up the call receives the VOA.

- Conference

  If an agent receives a call and then conferences in additional stations, any station on the connection can replay the message using the VOA Repeat button. Only the person using the button is able to hear the message unless the call is being service observed. In such case, the service observer also hears the message.

- Converse Split

  A converse-on split is a split used in a **converse-on** vector step. When a **converse-on** vector step is executed, VOA is not applied. After returning to the vector, the call can **route-to** a station or VDN where the answering agent will receive VOA (as if the **converse-on** step had not been encountered). Also, the final routed-to agent can replay the announcement if the agent has a VOA Repeat button administered.

- Coverage

  VOA applies to Coverage paths.

- Data Restriction

  The Data Restriction feature prevents tones from being applied to line or trunk circuits during a data call. The VDN of Origin announcement is not played for data restricted calls.

- Direct Agent Calling

  Direct Agent Calling allows a vector to **route-to** a particular ACD agent and have the call treated as an ACD call. The VDN of Origin announcement applies to the Direct Agent calls only if the calls reach the agent through vector processing. Direct Agent calls from a voice terminal on a switch do not encounter a VDN and therefore cannot cause a VDN of Origin Announcement to be played.

- Enhanced Automatic Wake-up

  If you are also using the G3V3 enhancements to Automatic Wake-up with integrated announcements, these two features can contend for use of the integrated announcement ports. VDN of Origin Announcements have priority over Automatic Wake-Up announcements.

- Expert Agent Selection (EAS)

  When you are using EAS, the COR option for the logical agent determines the assignment of VDN of Origin Announcements for each particular extension. EAS uses the COR of the logical agent instead of the COR for the terminal being used by the agent.

- Home Agent

  An initial VDN of Origin Announcement can be assigned to a Home Agent port on the switch. However, because home agents require a dial access code (DAC) to reach features and because the ability to replay a VDN of Origin Announcement does not use a DAC, the VOA Repeat key function is not available to a Home Agent user.

- Hunt Groups

  VDN of Origin announcements apply to calls routed to a hunt group. If this occurs, the COR for extension associated with the answering station determines whether the station can receive a VDN of Origin announcement. If the answering station has a VOA Repeat button, the agent can use it to replay the message.

- Lookahead Interflow

  VDN of Origin Announcements apply only to the switch on which the VDN is defined. If a call interflows to another switch, the VDN of Origin announcement is lost. You can provide some level of transparency by having the interflow to the other switch access a VDN with the same VDN of Origin Announcement. Note that interflow to another switch does pass on the display information.

- Redirection on No Answer (RONA)

  Calls that are not answered before the RONA timer expires are requeued to await another available agent. The timer stops when an agent answers the call. If the requeues are a result of the RONA timer expiring, the VDN of Origin Announcement applies to the call at the time an agent finally answers the call.

- Service Observing

  Service Observing allows an observer to bridge on to ACD agent calls. The system treats the call as a conference connection. Therefore, the service observer can press the VOA Repeat button to replay the VDN of Origin Announcement, and only the service observer hears the announcement. However, if any other party on the call presses the VOA Repeat button, both the user and the service observer hear the announcement.

- Supervisor Assist

  If an agent receives a call and then requests supervisor assistance and conferences in the supervisor, both the agent and the supervisor can use their VOA Repeat button to replay the message. Only the person who presses the button hears the message.

- Transfers

  If an agent receives a VDN call and subsequently transfers the call, the receiving station can use the VOA Repeat button to replay the message.

- VOA Distribution

  If several VOA announcements are used, or if long VOA announcements are used, it is possible that a delay may occur between the zip tone and the announcement. On pre-G3V4 units, it may be possible to install an auxiliary announcement device on the AUX port. An AT&T 15A unit can be used in most applications. Also, if the system is a G3V3 or lower, it may be better to upgrade the system to a G3V4. The G3V4 provides multiple announcement circuit packs to help prevent the announcement delays. Contact your AT&T representative for more information.

Administration

VDN of Origin Announcements feature must be enabled on the "System-Parameters Customer-Options" form. Vectoring must also be enabled.

If you want the attendants and agents to be able to replay VDN of Origin Announcements, you must administer a VOA Repeat feature button for the appropriate terminals. This is done using the "Attendant Console" form and the specific "Voice Terminals Station" forms.

Because agent ability to receive announcements is based on their COR, you must use the "Class of Restriction" form to administer VDN of Origin Announcements for each COR you want to receive the announcements.

Each announcement must be administered on the "Recorded Announcements" form. You can administer aux-trunk types with queue, without queue, and with barge-in. You can also administer integrated types with queue and without queue. Analog and integrated repeating announcement types are not allowed for VOA.

Because VDN of Origin Announcements are associated with VDNs, you must administer announcement numbers on the "Vector Directory Number" form. A VDN of Origin Announcement can be assigned to any number of VDNs, but a single VDN can have, at most, one VDN of Origin Announcement.

Hardware and Software Requirements

This feature requires the following circuit packs: TN7417, TN763C, or TN763D auxiliary trunk for announcement devices, and/or the TN750B or higher integrated announcement boards for internal announcements.

The feature can operate on any attendant terminal and any other voice terminal. If you want to use the VOA Repeat button, the voice terminals must be equipped with feature buttons. Because of the short length of the announcements, agents should use speakerphones or headsets.

Vectoring and VDN of Origin Announcements must be enabled.

Visually Impaired Attendant Service (VIAS)

Feature Availability

This feature is available with G3i-Global, all Generic 3 V2, and later releases.

Description

Provides voice feedback to a visually impaired attendant in either Italian or British English. Each voice phrase is a sequence of one or more single voiced messages.

Six new attendant buttons are defined for the VIAS feature:

- Visually Impaired Service Activation/Deactivation Button: This button activates or deactivates the feature for the console from which it was pressed. When VIAS is activated, an indicator lamp lights next to the button. In addition, all ringers which were disabled (for example, recall, incoming calls, and so on) are enabled.
- Console Status Button: This button allows the visually impaired attendant to determine whether the console is in Position Available or Position Busy state, whether the console is a night console, the status of the attendant queue, and the status of system alarms.
- Display Status Button: This button allows the visually impaired attendant to determine what is shown on the console display. Not all display features are supported by VIAS. Class of restriction information, personal names, and some call purposes ar not supported.
- Last Operation Button: This button voices the last operation performed.
- Last Voiced Message Button: This button allows the visually impaired attendant to retrieve the last voiced message.
- Direct Trunk Group Selection Status Button: This button allows the visually impaired attendant to obtain the status of an attendant monitored trunk group.

The visually impaired attendant may use the Inspect mode locate each button and determine the feature assigned to each without actually executing the feature. To do this, the attendant presses the Inspect button and then presses each button in turn and listens to the voiced information about it. Afterwards, the attendant presses the Normal button to end Inspect mode.

Considerations

Some changes on the attendant console are automatically voiced, for example, alarms reported, night service activated, and call threshold reached.

After system initialization, VIAS is not automatically activated. After a warm restart operation, VIAS remains activated if it was already activated. After recovery and cold restart operations, VIAS is not automatically activated even if it was activated before the recovery or restart attempt. Finally, whenever the attendant console goes to a busyout state and VIAS is activated, VIAS is automatically deactivated.

Interactions

When VIAS is activated, Auto Start is always enabled. The attendant can activate the Don't Split feature as normal if VIAS is activated, however, the

Don't Split feature, if currently activated, is automatically deactivated when the attendant deactivates VIAS.

Administration

VIAS is turned off by default. Before an attendant can use the Activate/Deactivate button, or any of the VIAS buttons, these buttons must be administered with the "Attendant Administration" form.

Hardware and Software Requirements

This feature requires at least one Speech Processor circuit pack to be installed into a system port carrier since VIAS capabilities are performed with speech synthesis messages voiced to the attendant. The United States speech processor circuit pack does not support VIAS.

Voice Message Retrieval

Feature Availability

This feature is available with all Generic 3 releases.

Description

Allows attendants, voice terminal users, and remote access users to retrieve LWC and Call Coverage messages in the form of a voice output. Voice choices are UK and US English and Italian; these choices depend on the voice synthesis circuit pack provided.

Voice Message Retrieval is used only for the retrieval of messages. It can be used to retrieve a user's own messages or to retrieve messages for another user. However, a different user's messages can only be retrieved by a user at a voice terminal or attendant console in the associated coverage path, by an administered systemwide message retriever, or by a Remote Access user when the extension number and associated security code are known.

Messages are protected by restricting unauthorized users from retrieving messages. A Lock function restricts a voice terminal and an Unlock function releases the restriction. The Lock function is activated by dialing a systemwide access code. The Lock function is canceled by dialing a systemwide access code and then an Unlock security code unique to the voice terminal. These functions apply only to the voice terminal where the operation is performed. The systemwide access codes and security code used for the Lock and Unlock functions are the same as those used for LWC message retrieval by display. A status lamp can be assigned to show the locked or unlocked status of a terminal.

Voice Message Retrieval is activated by dial access code. Separate access codes are used for Message Retrieval and Coverage Message Retrieval (someone else's messages). Voice Message Retrieval is activated as follows:

- To retrieve one's own messages:

  Dial the access code for Voice Message Retrieval of LWC messages. Then, dial a # sign to indicate that the extension whose messages are to be retrieved is the dialing extension, or enter a specific extension and corresponding password (same as the security code used for the Lock and Unlock functions).

- To retrieve someone else's messages:

  Dial the access code for Voice Message Retrieval of Call Coverage messages, followed by the extension of the user (within the same coverage path) whose messages are to be retrieved.

After Voice Message Retrieval has been activated, message retrieval may not be allowed for the following reasons:

- Speech Synthesizer circuit pack fails or all voice channels are busy — Reorder tone is provided.
- An attempt is made to retrieve a message for a user not in the same coverage path or an invalid password has been entered — ‘‘Message Retrieval Denied’’ is heard and the user is disconnected.
- Message retrieval for the terminal whose messages are to be retrieved is locked — ‘‘Message Retrieval is locked’’ is heard and the user is disconnected.
- No message is heard within 10 seconds — “Press 4 for help’’ is heard and if no messages are heard within another 10 seconds, the user is disconnected.

When Voice Message Retrieval has successfully been activated, voice messages are heard as follows:

- If no messages are left, one of the following is heard:
  - ‘‘No messages for [abc]’’ — where [abc] are the initials associated with the user whose messages are to be retrieved
  - ‘‘No messages for extension [xxxxx]’’ — where [xxxxx] is the extension number when no name is associated with the user whose messages are to be retrieved
- If one message, or more, is left, one of the following is heard:
  - ‘‘[n] messages(s) for [abc]’’ — where [n] is the number of messages left for the user whose messages are being retrieved
  - ‘‘[n] messages(s) for [xxxxx]’’
  - ‘‘Messages for [abc]’’ — when an AP is provided
  - ‘‘Messages for [xxxxx]’’ — when an AP is provided

When a user’s initials are given in a voice message, as previously described, the initials are computed in the “first name(s) followed by last name’’ order. If a single name (Brown, for example) is administered, the entire name is spelled out.

When a user has activated Voice Message Retrieval and has heard the number of messages that has been left, one of the following functions, as described below, can be performed:

- NEXT — dial #
- REPEAT — dial **5** or ***5**
- DELETE — dial **3** or ***3**
- CALL — dial **8** or ***8**
- HELP — dial **4** or ***4**

If **NEXT** is selected, the next message, if there is one, is played. The following messages may be heard:

- ‘‘No more messages for [abc]’’ or ‘‘No more messages for extension [xxxxx]’’ — when there are no more messages.
- ‘‘[abc] called [n] times, last message at [time] [date] extension [xxxxx]’’ — when the user has received more than one message from the same caller; [abc] are the caller’s initials; [n] is the number of times called; [time] is expressed an hour followed by minute (for example, ‘‘Nine Thirty-Five PM’’); [date] is expressed as month followed by day (for example, ‘‘July third’’) or “today,” if applicable; [xxxxx] is the calling party’s extension number.
- ‘‘extension [xxxxx] called [n] times, last message at [time] [date]’’ — when the user has received more than one message from the same caller and no name is associated with the extension.
- ‘‘[abc] called at [time] [date] extension [xxxxx]’’ or ‘‘extension [xxxxx] called at [time] [date]’’ — when only one message has been left by a particular extension number.

If the REPEAT function is selected, the synthesized voice repeats the previously retrieved message with the calling party’s name spelled out (instead of initials). The name is spelled out as it is administered in the system (with pauses between first and last names and also between first names if there is more than one). If no name is associated with the extension, the current message is repeated. If a message has not been retrieved, “Press pound for the next message’’ is heard.

If the DELETE function is selected, the previously retrieved message is deleted and the user hears “Message is deleted.’’ If no message was previously retrieved, ‘‘Press pound for next message’’ is heard. After a message is deleted, the user can still place a call to calling party of the deleted message, via the CALL function, as long as no other function has been entered between DELETE and CALL.

If the CALL function is selected, the extension of the calling party from the previously retrieved message is called. If no message was previously retrieved, “Press pound for next message’’ is heard. Otherwise, the call is initiated and the user leaves the message retrieval mode.

If the HELP function is selected, the following speech synthesized message is heard: “Press pound for the next message, press 3 to delete the message, press 4 for help, press 5 to repeat the message, press 8 to place the call.’’

The system expects the user to enter a function after each voice message. If a function is not entered before a specified time or if an invalid digit (digit other than #, *, 3, 4, 5, or 8) is dialed, the voice message “Press 4 for help’’ is heard. If no other input is entered within 10 seconds after this message, the user is automatically disconnected.

Voice Message Retrieval can be deactivated to get out of the voice message retrieval mode by doing any of the following:

- Hang up
- Press the Drop or Disconnect button
- Activate CALL function

Considerations

With Voice Message Retrieval, a display-equipped voice terminal is not required to retrieve messages. Authorized users on any touch-tone terminal can retrieve messages. This results in significantly reduced traffic to the Message Centers and systemwide message retrievers.

The number of simultaneous Voice Message Retrieval users possible is dependent on the number of speech synthesizer circuit packs used in the system.

Voice Message Retrieval cannot be accessed by rotary dialing voice terminals.

Certain voice terminals and attendants can be designated for systemwide message retrieval. These systemwide retrievers are the same as those used for display message retrieval and have the same privileges.

When a terminal is in the Voice Message Retrieval mode, it cannot be used to make calls or access other features.

Interactions

The following features interact with the Voice Message Retrieval feature.

- AUDIX Interface

  Retrieval of LWC messages via Voice Message Retrieval is separate and distinct from AUDIX voice message retrieval. LWC messages left for a principal on AUDIX may not be accessed via Voice Message Retrieval; however, the invoker of Voice Message Synthesized Retrieval is told if there are any new messages for the principal on AUDIX:

  — Voice Message Retrieval *voices* that there are message messages (dialing 8-callout calls AUDIX).

  — The display retrieval displays `Message Center AUDIX Call`.

  If your system has a voice synthesis circuit pack, and if LWC Activation is enabled, then messages are retrieved from two locations:

  — LWC messages sent by pressing the LWC button on a voice terminal are retrieved via Voice Message Synthesized Retrieval.

  — All other messages are retrieved via AUDIX.

If your system does not have a TN725B Voice Messaging Retrieval board, and if LWC Activation is enabled, then LWC messages sent by pressing the LWC button on a voice terminal are not retrievable by a nondisplay set.

- Bridged Call Appearance

  Activation of Voice Message Retrieval on a Bridged Call Appearance functions the same as if it was activated by the primary extension associated with the Bridged Call Appearance.

- Call Forwarding

  A forwarded-to user cannot retrieve messages for a forwarding user unless the forwarded-to user is in the forwarding user's coverage path.

- Call Pickup

  A user cannot retrieve messages for a member of his or her Call Pickup group, unless the retrieving user is in the other user's coverage path.

- LWC

  Voice Message Retrieval enhances LWC by allowing any authorized touch-tone terminal user to retrieve messages.

Administration

Voice Message Retrieval is administered by the System Manager. The following items require administration:

- Voice Message Retrieval Access Code for LWC message retrieval

  (per system)

- Voice Message Retrieval Access Code for Call Coverage message retrieval (per system)
- Lock and Unlock Access Codes (per system)
- Unlock Security Code (per voice terminal)
- Identities of authorized systemwide message retrievers

Hardware and Software Requirements

Requires a TN725 (American English), TN457 (British English), or TN433 (Italian) Voice Synthesizer. Each circuit pack has four ports to provide Voice Message Retrieval. No additional software is required.

Voice Response Integration (VRI)

Feature Availability

Voice Response Integration (VRI) is available with G3V2 and later releases.

Description

VRI is designed to enhance the integration of the DEFINITY Communications System. VRI is designed to integrate the DEFINITY system Call Vectoring with the capabilities of voice response units (VRUs), particularly AT&T's CONVERSANT Voice Information System.

VRI can do the following:

- Execute a VRU script while retaining control of the call in DEFINITY system vector processing.
- Execute a VRU script while the call remains in the split queue and retains its position in the queue.
- Pool CONVERSANT ports for multiple applications. (Previously, this was possible only when ASAI was present.)
- Use a VRU as a flexible external announcement device.
- Pass data between a DEFINITY switch and a VRU.
- Tandem VRU data through a DEFINITY switch to an ASAI host.

The capabilities in the previous list are provided by the **converse-on** command, which is an enhancement to the Basic Call Vectoring customer option. The **converse-on** Call Vectoring step is specifically designed to integrate a VRU with the DEFINITY system Automatic Call Distribution (ACD). VRI allows VRU capabilities to be used while keeping control of the call in the DEFINITY system ACD. The inclusion of VRUs with vector processing provides the following advantages:

- Access to local and host databases
- Validation of caller information
- Text to speech capabilities
- Speech recognition
- Increased recorded announcement capacity
- Audiotex applications
- Interactive Voice Response (IVR) applications
- Transaction processing applications

One of the advantages of VRI is that it allows users to make more productive use of queuing time. For example, while the call is waiting in queue, the caller can listen to product information by using an audiotex application or by completing an interactive voice response transaction. In some cases, it may even be possible to resolve the customer's questions while the call is in queue. This can help reduce the queuing time for all other callers during peak intervals.

During the execution of a VRU script, if the caller previously queued to an ACD split, the caller retains his/her position in queue. If an agent on the DEFINITY switch becomes available to service the call, the line to the VRU is immediately dropped, and the calling party is connected to the available agent.

If the **converse-on** command is successful, it delivers the call to a pre-determined split (skill), which is referred to as the **converse-on split** (skill). A **converse-on split** is administered and behaves exactly as any other split in the system. The term **non-converse-on split** is used to refer to traditional ACD splits accessed by means of **queue-to main** and **check-backup** vector steps.

The members of the **converse-on split** are the DEFINITY ports connected to the VRU. If all VRU ports are currently busy, the call queues to the **converse-on split** at the administered priority.

Once the call is answered by the VRU, the **converse-on** command may or may not pass data to the VRU (depending upon the parameters of the command). Up to two items of data may be passed. This data may be used to select the VRU script to be executed or it may be information passed to the script.

The following values to be passed to the VRU may be administered within the **converse-on split (skill)** command:

NOTE:
Up to two items of information can be passed to the VRU.

- **Administered digit string**. This string can contain up to six characters consisting of one or more digits (0 through 9) or asterisks (*). The pound sign (#) may not be included in a digit string because it is reserved as the end-of-string character. However, a single "#" may be administered.
- **"ani"**: If the call is an internal call or an incoming DCS call, this data type causes the extension of the calling party to be outpulsed. If the call is an incoming ISDN PRI call with ANI (BN) provided to the DEFINITY system, the calling party number/billing number (CPN/BN) of the calling party is outpulsed to the voice information system. If there is no ANI (BN) to send, the end-of-string pound sign (#) is the only character outpulsed. Any other type of incoming call results in "#" being outpulsed.
- **"vdn"**: This data type causes the VDN extension to be outpulsed. In cases where multiple VDNs are accessed, normal VDN override rules determine which VDN extension is outpulsed.

- **‘‘digits’’:** This data type can be used only if Call Prompting is optioned, and it causes the most recent set of digits collected in vector processing to be outpulsed. If no digits are available, the end-of-string pound sign (#) is the only character outpulsed.
- **‘‘qpos’’:** This data type causes the value of the queue position of a call in a nonconverse split to be outpulsed. This value is a variable length data item from which between one and three digits can be outpulsed. If the call is not queued, the end-of-string pound sign (#) is the only character that is outpulsed.

  NOTE:
  The use of this keyword is not recommended with multiple split queuing because any queue position value that is sent may not be meaningful. However, if the call is queued to multiple nonconverse splits, the value of the caller’s queue position in the first nonconverse split is sent.

  This data may be used by the voice information system to inform callers of their position in queue or to decide whether to execute a long or short version of a voice response script.
- **‘‘#’’**: This is the only character outpulsed. Outpulsing this character causes the corresponding *prompt and collect* command in the voice response script to be skipped.
- **‘‘none’’**: This data type causes no characters to be outpulsed. Also, no end-of-string pound character (#) is outpulsed, and no time delays are invoked.
- “**wait**”: This data type can be used only if the “Vectoring (G3V4 Advanced Routing)” customer option is enabled. It causes the expected wait time of the call to be outpulsed. If the call is not queued or if it is queued only to splits that are unstaffed or splits where all agents are in AUX work mode, the end-of-string character “#” is the only character outpulsed.

Regardless of whether or not data is passed, the caller is then connected to the VRU, which in turn executes the VRU script. At this point, any audible feedback provided by the vector is disconnected, and no further vector steps are executed until the VRU script is completed.

If the call was queued to a **non-converse-on split** before the **converse-on** command was executed, the call retains its position in the **non-converse-on split** queue. If an agent from the **non-converse-on split** becomes available to service the call while the VRU script is still being executed, the DEFINITY system drops the line to the VRU and connects the caller to the available agent. The VRU in turn detects the disconnect and terminates the VRU script.

Once the VRU script is completed, the VRU may or may not return data to the DEFINITY system. Whether or not data is returned, once the VRU drops the line to the DEFINITY system, vector processing is then re-activated on the switch starting with the vector step which follows the **converse-on** step.

Digits returned by the VRU are treated as dialahead digits. Call prompting must be optioned to collect and use these digits. The rules for collecting and processing VRU-returned digits are identical to those for Call Prompting.

Digits returned from the VRU can be used in the following way:

- Displayed on the answering agent's display set (automatically for 2-line displays (such as a 7407 or Call Master) using the CALLR-INFO button for other display sets).
- Treated as an extension in a **route-to digits** vector step. For example the following:
  - `converse-on split. . . . (VRU returns 4 digits)`
  - `collect 4 digits after announcement none`
  - `route-to digits coverage y`
- Used for vector conditional branching in an **if digits equals** vector step. For example the following:
  - `converse-on split . . . (VRU returns 1 digit)`
  - `collect 1 digit after announcement none`
  - `goto vector 101 if digits = 1`
  - `goto vector 102 if digits = 2`
  - `goto vector 103 if unconditionally`
- Tandemed to an ASAI host.
  - VRI also provides the integration of VRUs with ASAI hosts. Since collected digits are also passed to ASAI hosts in ''Call Offered to Domain Event' reports and in `route request` messages, caller digits or database information returned from the VRU may also be tandemed through the DEFINITY system to ASAI hosts.

    For example:

    1. `converse-on split ... (VRU returns 9 digits)`
    2. `collect 9 digits after announcement none`
    3. `adjunct route link Y`

    In this vector, the digits returned from the VRU are forwarded to the ASAI host in the adjunct routing ''route request'' message.

Refer to *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide,* 555-230-520, for additional information.

Security Measures

When a converse step is used to access a VRU application that returns data for a collect digits step, the opportunity for toll fraud exists when the VRU application fails to return any data. To avoid this type of toll fraud be certain that one of the following is true:

- If the collected digits are used to route calls internally, be certain that the Class of Restriction (COR) for the Vector Directory Number (VDN) does not allow calls to route externally.
- If it is necessary to use the collected digits to route calls externally, use a password to verify that the collected digits have been passed by the VRU application. For example, in the following vector the VRU application returns a three-digit password followed by the eight-digit external number. The vector routes calls without the correct password to a different vector and routes calls with the correct password to the collected digits.

```
converse-on split 10 pri m passing none and none
collect 3 digits after announcement none
goto vector 23 if digits <> 234
collect 8 digits after announcement none
route-to digits with coverage n
```

Considerations

VRI allows the CONVERSANT to be used as a flexible external announcement device. Although only one caller can be connected to each port within this scenario, up to 48 callers can be simultaneously connected to CONVERSANT. VRI can be used to pass the announcement number to CONVERSANT, thus allowing any announcement to be played on any port.

Interactions

The following features interact with VRI.

NOTE:
If not noted below, the **converse-on split** has the same interaction as other vector controlled splits.

- Adjunct Switch Applications Interface (ASAI)

  Data returned from the VRU can be tandemed through the DEFINITY system to the ASAI host with VRI. This provides better integration of VRUs and ASAI hosts.

  If a **converse-on** step places a call to an ASAI-monitored domain, ASAI event messages are sent over the ASAI link.

When an ASAI-monitored call is placed by a **converse-on** vector step, the ALERTing message sent to the ASAI host includes a Cause IE, Coding Standard 3 value 23 (CS3/23). As a result, the ASAI host is informed that the call has not been de-queued from any **non-converse-on splits**.

If a **converse-on** step is executed while an adjunct routing request is outstanding, the route request is canceled.

Any attempt by ASAI to transfer or conference the call is denied.

- Answer Supervision

  Answer supervision is returned only once during the life of a call. If a call is answered as a result of a **converse-on** step, answer supervision is sent if it hasn't previously been sent. If digits are passed to the VRU, answer supervision is not sent until after the digits are outpulsed.

- Audio Information Exchange (AUDIX)

  If a **converse-on** step calls AUDIX, the call is treated as a direct call to AUDIX. The caller hears the ''welcome to AUDIX'' message and may retrieve his or her messages in the usual manner.

  If a call is forwarded to a VDN and then delivered to an AUDIX hunt group by a **converse-on** step, the call to AUDIX is treated as a redirected call, and the caller may leave a message for the principal.

- Automatic Answering with Zip Tone

  When administering conversant ports as agents of a **converse-on split**, agents should not be administered as automatic answer. If auto-answer is administered, the DEFINITY system provided zip tone may interfere with the interaction between CONVERSANT and the calling party.

- Auto-Available Splits

  A **converse-on** vector step may place a call to an auto-available split. Except in cases where the **converse-on split** is ASAI-controlled, auto-available **converse-on splits** are recommended for VRI.

- Basic Call Management System (BCMS)

  BCMS tracks calls placed by a **converse-on** step to a BCMS-measured hunt group. The VDN tracks this call as waiting in the vector. A call is considered ''answered'' when it is answered by a **non-converse-on split** but never when it is answered by a **converse-on split**. The **converse-on split** tracks this as a separate ''answered'' call when the VRU answers. Trunk totals may no longer match split totals, however, VDN totals and trunk totals match.

- Call Detail Recording (CDR)

  For incoming calls to a VDN, the duration of the call is recorded from the time answer supervision is returned. Answer supervision is returned for a successful **converse-on** step. No ineffective call attempt records are generated for **converse-on** steps that fail. Also, no outgoing calls can be placed by a **converse-on** step.

- Call Management System (CMS)

  CMS tracks calls placed by a **converse-on** step to a CMS-measured hunt group, split or skill. **Converse-on** vector steps may be administered from CMS R3V2 and later releases.

  The VDN tracks this call as waiting in the vector. A call is considered ‘‘answered’’ when it is answered by a **non-converse-on split** but never when it is answered by a **converse-on split**. The **converse-on split** tracks this as a separate ‘‘answered’’ call when the VRU answers. Trunk totals may no longer match split totals, however, VDN totals and trunk totals match.

- Call Park

  Calls placed by a **converse-on** step may not be parked.

- Call Pickup

  Calls placed by a **converse-on** step ringing at an agent station may be picked up if that agent is part of a pickup group. Subsequent transfers are denied.

- Call Prompting

  The Call Prompting customer option must also be enabled to gain full VRI functionality. Without Call Prompting, any data returned by CONVERSANT cannot be collected and processed by the switch.

  If the **converse-on** step places a call to a split of live agents, any digits collected previously may be displayed by agents using the callr-info button.

- Call Vectoring — Basic

  The **converse-on** vector step is an enhancement to the Basic Call Vectoring customer option. This option must be enabled in order to invoke the VRI feature.

- Class of Restriction (COR)

  As is the case for the **queue-to main split** and **check-backup split** vector steps, no COR checking is carried out when a **converse-on** vector step places a call to a split.

- Conference

  Any attempt to conference a call placed by a **converse-on** step is denied.

- Direct Department Calling (DDC)

  A **converse-on split** may be administered as a direct department calling split.

- Distributed Communications System (DCS)

  If an incoming DCS call is placed to a vector with a **converse-on split *x* pri m passing ani...** step, the DCS extension of the calling party is outpulsed.

- Expert Agent Selection (EAS)

  The **converse-on** vector step may place a call to a skill hunt group.

- Feature Access Codes

  Dialing the converse data feature access code at any time other than during a converse-on session produces an intercept tone.

- Four Priority Levels

  A call placed by a **converse-on** step may be queued at one of four priority levels: low, medium, high or top.

- Hold

  Although not recommended, a **converse-on** step may place a call to a split of live agents. An agent answering a converse call may put the call on hold. Music on hold, if administered, is suppressed.

  If an agent from a non-converse-on split services the call while the caller is placed on hold, the agent who placed the caller on hold is dropped from the call, and the caller is connected to the agent from the non-converse-on split.

- Hold — Automatic

  Automatic hold applies to converse-on calls.

- Hunt Groups

  The **converse-on** vector step may deliver a call to a vector-controlled hunt group, ACD split, Agent Skill, Message Center or AUDIX hunt group.

- Integrated Services Digital Network (ISDN)

  The **converse-on** step may be administered to outpulse to CONVERSANT the CPN/BN (calling party number/billing number) of the calling party via use of the ‘‘caller’’ keyword.

- Intra-switch CDR

  If a converse-on call is answered successfully, and if either the calling party or the VDN relevant to the call is administered for intra-switch recording, the timing for the call is started. If either the calling party or VDN is so administered, the CDR record shows ‘‘calling party to VDN’’ as the originating and answering parties.

- Line Side T1 Connectivity

  T1 connectivity between DEFINITY and CONVERSANT is supported for the VRI feature. The DS1 board must be a TN767E (or later) or TN464F (or later). On the switch side, all the converse agents should be administered as DS1FD type stations. The feature operation of the converse step using

Line Side T1 is identical to that over a Tip/Ring line. In particular, the delay timing and outpulsing speed is the same as for analog lines. Currently, T1 connectivity to CONVERSANT is only supported in the United States and Canada.

- Live Agents

  Although not recommended, the switch does not prevent a **converse-on** step from delivering a call to a group of live agents. To the agent, the call looks like any other ACD call. However, certain features, such as call transfer, conference, and supervisor assist are denied.

  The answering agent can display any digits collected prior to executing the **converse-on** step by using the ‘‘callr-info’’ button. Collected digits are displayed automatically on the second line if the answering agent has a two-line display set such as a 7407 or a Call Master station.

  It is possible for the agent to return data to vector processing by using the transfer feature and by dialing the converse-on data return code followed by the digits themselves.

- Look-Ahead Interflow

  If a call on the receiving switch is answered by a VRU, or a call placed by a **converse-on** vector step is answered by a VRU, or is queued to the **converse-on split** while a Look-Ahead Interflow call attempt is outstanding, the Look-Ahead Interflow call attempt is accepted.

  A **converse-on** step that fails is neutral (neither accepted or rejected).

- Message Center

  The **converse-on** vector step may deliver calls to message hunt groups. Such calls are treated as direct calls to the message.

  If a call is forwarded to a VDN and then delivered to a message split by a **converse-on** step, the call is treated as a redirected call.

  A call can be queued to three different skills and then to a converse-on skill group as a result of a **converse-on** step.

  A call can be queued to three different splits and then to a converse-on split as a result of a **converse-on** step.

- Music-on-Hold

  During the data return phase of a **converse-on** step, the caller is temporarily placed on hold. Music on hold, if administered, is suppressed.

- Non-Vector Controlled Splits

  A **converse-on** step may not place a call to a non-vector-controlled split.

- Priority Queuing

  The queue priority of a call placed by a **converse-on** step is administrable on the vector step.

- Queue Status

  All queue status display, queue status indication and queue warning wall lamp feature capabilities also apply to **converse-on splits**.

- Queuing

  Converse-on calls queue when they are delivered to busy hunt groups. Call Vectoring audible feedback is not disconnected while a converse-on call is in queue.

  If a **converse-on** step is executed while a call is queued to a **non-converse-on split**, the call remains in queue for the **non-converse-on split**.

  The queue priority of the call is administrable on the vector step.

- Recorded Announcement

  VRI may be used to increase the system's recorded announcement capacity by offloading some recorded announcements to the VRU. Callers can be redirected by the **converse-on** step to a group of VRU ports via passing the number of the announcement to be played. Only one call at a time is connected to a VRU port.

- Redirection on No Answer (RONA)

  If a **converse-on** step places a call to a hunt group with a ''no answer timeout'' administered, and if the call rings at an agent terminal/port for longer than the administered timeout, the call is redirected, and the agent/port is put into the AUX work state (or logged out if the agent is a member of an auto-available split).

  Thereafter, under RONA, the call is requeued to the split unless there is no room in the queue or unless this is an auto-available split whose agents are all logged out. If the call cannot be requeued, the **converse-on** step fails, a vector event is logged, and vector processing is restarted at the next vector step.

- Service Observing

  Calls placed by a **converse-on** step may be service observed. To prevent the observer from hearing tones that are being outpulsed to the VRU, the observer is not connected to the call until the data passing phase is complete. If data is returned by the VRU, the observer is put in service observing pending mode, and the calling party is temporarily put on hold while the VRU digits are outpulsed. Upon completion of the converse-on session, and once the VRU hangs up the line, the observer remains in service observing pending mode.

It is not recommended that a service observing warning tone be administered since the warning tone may interfere with the interaction between the CONVERSANT and the calling party.

- System Measurements

  System measurements track converse-on calls to hunt groups.

- Touch-Tone Dialing

  Any touch-tone dialing by the calling party during the digit passing phases of a converse-on session does not result in corruption of data or in the collection of this data in the form of dial-ahead digits by the DEFINITY Communications System.

  Only after the DEFINITY switch to CONVERSANT digit passing phase is complete can the calling party enter touch-tone digits in response to a CONVERSANT prompt. Only after the CONVERSANT to the DEFINITY switch data return phase is complete and an additional **collect <#> digits** vector step is executed can the calling party enter a touch-tone response to a DEFINITY system prompt.

- Transfer

  A call distributed by a **converse-on** step may not be transferred. The only form of transfer allowed is the data passing operation during the data return phase at the end of a CONVERSANT script.

  If an illegal attempt to transfer a converse-on call is made, a vector event is logged, the line to the CONVERSANT is dropped, and vector processing is re-activated at the next vector step.

  If an illegal transfer is attempted by a live agent with a multifunction set, the transfer is denied and the agent may reconnect to the call.

- Transfer out of AUDIX

  If a **converse-on** step delivers a call to an AUDIX hunt group, and if the calling party then attempts to transfer out of AUDIX, the transfer is denied. Vector processing continues with the next step.

- Uniform Call Distribution (UCD)

  A **converse-on split** may be administered as a Uniform Call Distribution split.

- VDN Display Override

  If a call that accesses multiple VDNs encounters a **converse-on** step passing ‘‘vdn,’’ normal display override rules determine which VDN number is outpulsed to the VRU.

- Vector-controlled splits

  A **converse-on** step may place a call to a split only if that split is administered as a vector-controlled split.

Administration

Basic Call Vectoring must be enabled in order to use the **converse-on** vector step. To enable complete functionality of VRI, Call Prompting must also be optioned.

If Call Prompting is not optioned, the Converse Data Return code cannot be administered, and data cannot be returned from CONVERSANT. Also, since the DEFINITY system cannot collect any digits from the caller, the ‘‘digits’’ keyword cannot be administered for the <data_1> or <data_2> fields. All other VRI capabilities are available.

Hardware and Software Requirements

Basic Call Vectoring must be enabled in order to use the **converse-on** vector step. To enable complete functionality of VRI, Call Prompting must also be optioned.

No new or changed hardware is required for this feature.

The TN744 Call Classifier circuit packs required by Call Prompting are not required for the returning of digits in-band from CONVERSANT to the DEFINITY system. Instead, general purpose TTR (TN748) circuit packs are used.

Voice Terminal Alerting Options

Feature Availability

This option is available for the CALLMASTER Voice Terminal with G3V1.1, G3V2, and later releases. It is available for Digital Voice Terminals and Hybrid Voice Terminals only with G3V2 and later releases.

Description

Provides multi-appearance voice terminal users with a range of audible ringing treatments. These treatments primarily affect the audible ringing applied to calls directed to off-hook stations. They also affect the ringing treatment applied to calls directed to idle and active CALLMASTER Voice Terminals.

The Voice Terminal Alerting Option for Digital Voice Terminals and Hybrid Voice Terminals has four possible values. They are:

- continuous

  All calls directed to this station will ring continuously until they are answered, redirected or abandoned.

- single

  If this station is off-hook and idle or off-hook and active and a call is delivered, the call will receive a single ring cycle and then ring silently until the call is answered, redirected or abandoned. ‘‘single’’ is the default value.

- if-busy-single

  If this station is off-hook and idle and a call is delivered to it, then the call will ring continuously. If this station is off-hook and active and a call is delivered to the it, then the call will receive a single ring cycle and then ring silently.

- silent

  If this station is off-hook and idle or off-hook and active, incoming calls will receive no audible ringing.

The Voice Terminal Alerting Option for CALLMASTER Voice Terminal has four possible values. They are:

- continuous

  All calls directed to this CALLMASTER Voice Terminal will ring continuously until they are either answered, redirected or abandoned.

- single

All calls delivered to this CALLMASTER Voice Terminal will receive a single ring cycle and then ring silently until answered, redirected or abandoned. single is the default value.

- if-busy-single

  If this CALLMASTER Voice Terminal is idle and a call is delivered to it, the call will ring continuously. If this terminal is active on a call and another call is delivered it, then the call will receive a single ring cycle and then ring silently.

- silent-if-busy

  If this CALLMASTER Voice Terminal is idle and a call is delivered to it, the call will ring continuously. If this terminal is active on a call, incoming calls will receive no audible ringing.

Considerations

Voice Terminal Alerting Options do not affect the ringing treatment applied to switch-hook equipped terminals which are on-hook. These terminals will always ring continuously when a call is directed to them. They will ring audibly until the call is answered, redirected or abandoned.

Certain ringing treatments are specified by these options for "idle" stations. When a station has no calls or all calls are ringing or on hold, the station is considered to be "idle".

This option is not available with Analog Voice Terminals, MET Sets (Multi-button Electronic Telephones) and ISDN-BRI Voice Terminals.

Interactions

- Bridged Call Alerting

  If set to yes then bridged call appearances will ring audibly when a call is directed to that station. The audible alerting applied to the station will be determined by the Voice Terminal Alerting Option administered on the station which is ringing.

- Distinctive Ringing

  A station can still be administered for Distinctive Ringing. The Voice Terminal Alerting Option will have no affect on its operation.

- Priority Calling

  Priority calls receive a distinctive audible alerting. The Voice Terminal Alerting Option has not affect on this feature.

Administration

The Alerting Options - Multi-Appearance Voice Terminal feature is administered on a per-voice terminal basis by the System Manager.

The following item requires administration for Digital Voice Terminals and Hybrid Voice Terminals:

— ‘‘Active Station Ringing’’ option (per voice terminal)

The following item requires administration for CALLMASTER Voice Terminals:

— ‘‘Idle/Active Ringing’’ option (per voice terminal)

Hardware and Software Requirements

No additional hardware is required.

No additional software is required to use Voice Terminal Alerting Options with CALLMASTER Voice Terminals. G3V2 or later release software is required to use Voice Terminal Alerting Options with Digital Voice Terminals and Hybrid Voice Terminals.

Voice Terminal Display

Feature Availability

This feature is available with all Generic 3 releases.

Description

Provides multiappearance voice terminal users with updated call and message information. This information is displayed on a display-equipped terminal. The information displayed depends upon the display mode selected by the user.

Terminal users may select any of the following as the display message language: English (default), French, Italian, Spanish, or other. Please see the "Administrable Language Displays" feature for more information.

NOTE:
The rest of this description applies when English is selected.

Several modes can be assigned to buttons and then selected by pressing the assigned button. All buttons are located on the display module or voice terminal. All buttons are administrable.

- Normal Mode

  Displays call-related information for the active call appearance. This display includes information identifying the call appearance, calling or called party, and calling or called number. The display must be in the Normal mode to answer incoming calls and to display information associated with the Automatic Incoming Call Display feature.

- Inspect Mode

  Displays call-related information for an incoming call when the user is active on a different call appearance. This button is pressed when the user is active on one call appearance and receives a call on another appearance.

- Stored Number Mode

  Displays the last number the user dialed (Last Number Dialed feature), the number stored in an Abbreviated Dialing button administered to the voice terminal, a number stored in an Abbreviated Dialing list, or a number assigned to a button administered through the Facility Busy Indication feature.

- Date and Time Mode

  Displays the current date and time of day.

- Integrated Directory Mode

  Turns off the touch-tone signals and allows the touch-tone buttons to be used to key in the name of a system user. After a name is keyed in, the display shows that name and associated extension number. (Refer to the Integrated Directory feature for complete details.)

- Message Retrieval Mode

  Retrieves messages for voice terminal users. If no messages are stored, display shows NO MESSAGES.

- Coverage Message Retrieval Mode

  Retrieves messages for voice terminal users who do not have a display module assigned to their voice terminal. Retrieval permission must be administered for a user to retrieve another user's messages. Messages can be retrieved at any time. The retriever does not need to lift the handset to retrieve messages. Also, messages can be retrieved even if the retriever is active on a call.

Elapsed Time is a display feature that can be invoked while the display is in the Normal display mode. It displays elapsed time in hours, minutes, and seconds. The timing starts or stops when the button is pressed. This button can be pressed at any time.

The Message Retrieval, Coverage Message Retrieval, or Integrated Directory buttons have three other associated buttons:

- Next Message

  Retrieves the next message or displays END OF FILE, PUSH Next TO REPEAT when in the retrieval mode. Displays the next name in the alphabetical listing when in the Integrated Directory mode. This button must be assigned when a Retrieval button is assigned.

- Delete

  Deletes the currently displayed message. This button must be assigned when a Retrieval button is assigned.

- Return Call

  Automatically returns the call requested by the currently displayed message or the currently displayed name and extension number.

The system provides the following call-related information:

- Call Appearance Identification

  The call appearance buttons are designated on the display by a lowercase letter; for example, a, b, and c. The display shows a= for a call incoming on the first call appearance button, b= for a call incoming on the second call appearance button, and so on.

- Calling Party Identification

  When the call is from a system user, the display shows the caller's name or a unique identification administered for the voice terminal being used, along with the calling party's extension number. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, and the trunk access code, assigned to the trunk group used for the call. If a user is active on a call, and receives a subsequent call, the display automatically shows the identification of the subsequent caller for a few seconds then automatically restores the display associated with the active call.

- Called Party Identification

  On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name and extension number. If no name is accessed, the dialed digits continue to be displayed.

  On outgoing calls, the display shows the digits as they are dialed, followed by the name and trunk access code assigned to the trunk group being used. The System Manager can suppress the name of any trunk group. If such a trunk group is accessed, the name portion of the display is blank.

- Call Purpose

  This identifies the reason for an incoming call or a redirected call. (A normal incoming call is not identified by a call purpose.) The following call purpose identifiers can be displayed:

  f — Call Forwarding — Indicates that another user has forwarded calls to this voice terminal.

  s — Send All Calls — Indicates that the called user is temporarily sending all calls to coverage, and that the call has been redirected to this voice terminal.

  c — Cover All — Indicates that the called user has Cover All criteria assigned.

  d — Coverage on Don't Answer — Indicates that the call was redirected because the called voice terminal was not answered. This identifier also indicates that the called voice terminal user has a temporary bridged appearance of the call.

  b — Busy — Indicates that the called voice terminal user is active on a call, and the called voice terminal user has a temporary bridged appearance of the call.

  B — Busy — Indicates that the called voice terminal user is active on a call, and the called voice terminal user does not have a temporary bridged appearance of the call.

Callback — Indicates that the call is an Automatic Callback call from the system.

Icom — Indicates that the incoming call is an Intercom call.

Park — Indicates that the user parked a call.

Pickup — Indicates that the user answered a Call Pickup group member's call.

Priority — Indicates that the incoming call has priority status.

Some typical displays are as follows:

- Internal call

```
a=3602
```

then

```
a=        TOM BROWN                3062
```

or

```
a=        EXT 3602                 3062
```

- Outgoing trunk call

```
b=87843541
```

Where 8 is the trunk access code and 784-3541 is the number dialed:

then

```
b=    OUTSIDE CALL                     8
```

or

```
b=              WATS                101
```

- Incoming trunk call

```
a=    OUTSIDE CALL                   102
```

Where 102 is the trunk access code of the incoming trunk group.

- Conference call originated by the attendant

```
b=    CONFERENCE 4
```

Where 4 is the number of conferees. The number does not include the conference call originator.

- Internal call redirected to coverage

```
b=      EXT 3174   to   EXT 3077     d
```

or

```
b=      BOB SMITH  to  JOYCE THOMAS  d
```

Where ***d*** indicates that **Go To Cover** was activated by the calling voice terminal user.

- Incoming trunk call redirected to coverage

```
b=    OUTSIDE CALL  to  DON SMITH    s
```

Where ***s*** indicates that **Send All Calls** was activated by the called voice terminal user.

- Message Retrieval

```
IN PROGRESS
```

then

```
MESSAGES FOR BETTY R. SIMS
```

then

```
JOE JONES 10/16 11:40a 2 CALL 3124
```

This message means that Joe Jones called Betty Sims the morning of October 16. The second message was stored at 11:40 a.m. Joe wants Betty to call his extension number, 3124.

- Integrated Directory mode

```
CARTER, ANN        3408             3
```

This display shows the name and extension number as administered in the system. The 3 indicates that three buttons were pressed to reach this particular display.

Considerations

The Voice Terminal Display feature provides an instant display of information associated with certain system features, functions, and services. Information that allows personalized call answering is available on many calls. Retrieval of stored information, such as messages received and directory information, is easy as well as convenient.

The system supports as many display modules as are needed, provided the voice terminal has a sufficient number of buttons.

Certain voice terminals and the attendant group can be designated for systemwide message retrieval. Users of these voice terminals or consoles can retrieve LWC and Call Coverage messages for other voice terminal users including DDC groups, UCD groups, PCOL groups, and TEGs. Selected users cannot retrieve messages for other selected users. Systemwide retrieving voice terminals or consoles are assigned when the system is implemented.

If the following conditions are met, messages for a voice terminal user can be retrieved at selected terminals or any attendant console:

- The retriever must be in the user's Call Coverage path.
- Permission to retrieve messages must be assigned for the user's voice terminal.

If permission is granted, any voice terminal with a display module or the attendant group in the user's Call Coverage path can retrieve messages for that user.

When all messages have been displayed and deleted for an extension number, the Message lamp on the voice terminal and any associated Remote Message Waiting Indicator, if assigned, go dark.

The display module used with voice terminals is similar to the attendant console display. However, the display module has an On-Off button, and can be turned off when not in use. The display module can be used only with specific voice terminals.

If you are using a 7506D or 7507D to make calls that require additional digits, a comma may appear in the dial sequence after you receive second dial tone or after the call has been set up. The comma is used to separate the called number from subsequent information.

Interactions

The following features interact with the Voice Terminal Display feature.

- Bridged Call Appearance

  A call from the primary extension number or a bridged call appearance of the primary extension number is displayed as a call from the primary extension number.

- Last Number Dialed

  If the Last Number Dialed feature access code is dialed after the stored number button has been pressed, the last number dialed is no longer displayed. However, if the Last Number Dialed button is pressed after the stored number button has been pressed, the last number dialed is displayed.

- Single-Digit Dialing and Mixed Station Numbering

  If prefixed extensions are used in the system's dial plan, the prefix is not displayed when the extension is displayed. The Return Call button can be used to dial prefixed extensions, because the system dials the prefix, even though it is not displayed.

Administration

Voice Terminal Display is administered on a per-voice terminal basis by the System Manager. The following items require administration.

- Whether or not a display module is provided (per display capable voice terminal)
- Whether or not to restrict other users from reading or canceling the voice terminal's message (per display module)
- The following buttons (per display module):
  - Normal
  - Inspect
  - Stored Number
  - Date and Time
  - Elapsed Time
  - Integrated Directory
  - Message Retrieval
  - Coverage Message Retrieval
  - Next Message (must be assigned with either Retrieval button)

— Delete (must be assigned with either Retrieval button)

— Return Call (optional with either Retrieval button or the Integrated Directory button)

Hardware and Software Requirements

DCP services require a display-equipped voice terminal and one port on a TN754 or TN2181 Digital Line circuit pack (TN413, TN754B, and TN2181 support A-law). No additional software is required.

ISDN-BRI services require the following hardware:

- TN778 Packet Control circuit pack, which provides the interface to the LAN (packet) bus on G3i for establishing the signaling connectivity
- TN556 BRI-S/T (4 wire) port circuit pack or TN2198 ISDN BRI-U (2 wire) line circuit pack
- AT&T ISDN 7506 and 7507 voice terminals

Voice Terminal Flash Timing

See the "Recall Signaling" feature.

VuStats

Feature Availability

VuStats is available with G3V3 and later releases.

What Is VuStats?

VuStats presents Basic Call Management System (BCMS) statistics on voice terminal displays. By pressing a button, agents, split supervisors, call center managers, and other users can access statistics for agents, splits or skills, VDNs, and trunk groups.

With G3V3 these statistics are limited to information collected either during the current BCMS interval, or in some cases since the agent logged in. G3V4 and later releases can also display historical data accumulated over an administered number of intervals. In either case, the information is limited to 40 characters at any one time. It can display on demand or update periodically.

With VuStats, any digital display voice terminal user can view BCMS statistics normally available only on BCMS reports or management terminals. These statistics can help agents monitor their own performance, or can be used to manage splits or small call centers.

NOTE:
Although VuStats can run with either BCMS or CMS enabled, neither is required. BCMS does not need to be optioned for VuStats to be available.

The following picture illustrates a CallMaster terminal with a VuStats display.

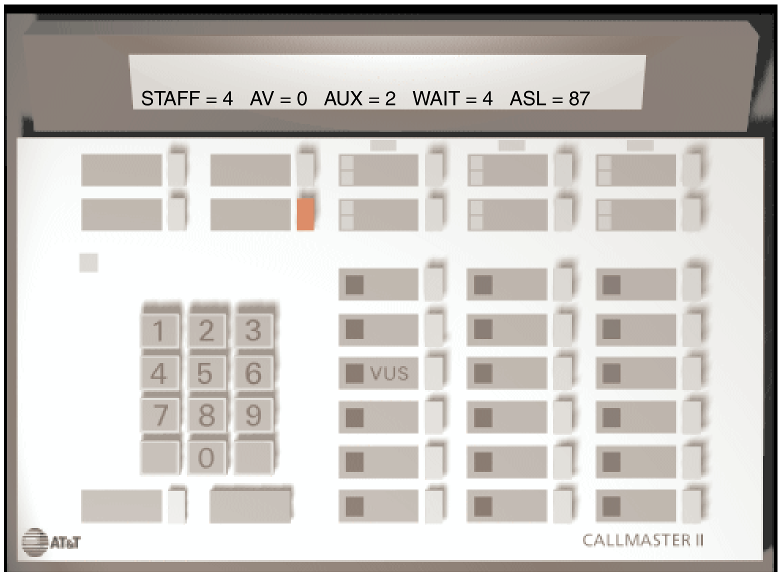


Figure 3-31. CallMaster Terminal with VuStats Display

VuStats Applications

Self-Managed Teams

VuStats can provide call center agents with statistics about their performance. For example:

- Agents can review their statistics against a predefined objective for a given split. This could include an agent comparing their average talk time per call to a split objective for average talk time, or comparing their calls handled against a split objective for calls handled. The agent can also review a split objective for percent of calls within service level to see if they should improve their performance to improve the overall split percentage.
- Agents can also review their statistics against split averages for all agents. In this case, an agent could compare their average talk time per call to the average talk time per call for the entire split.

Statistics also are useful to call center agents in managing their time. For example:

Agents might examine statistics about the number of agents logged into a split, the number of agents available to receive Automatic Call Distribution (ACD) calls, the number of agents who have entered AUX work mode, the number of abandoned calls, or the time the oldest call in queue has been waiting. This information can be used to determine if it would be best to log out of one split to devote time to a busier split, or if it was an appropriate time to take a lunch break.

Information Distribution

VuStats can increase the distribution of performance statistics throughout the call center. It replaces the need for additional CMS terminals for supervisors while also providing essential statistics to agents without the need for additional hardware. It lessens requirements for BCMS paper reports.

Call Center Management

Even small call centers without BCMS or CMS enabled can receive VuStats statistics. These statistics can be used to evaluate agent performance, balance workloads, examine split staffing requirements, and generally optimize call center performance.

Detailed Description

VuStats takes information stored on the switch in BCMS database tables and shows that information on a display terminal. In general, the system administrator uses the "VuStats Display Format" form to create a format defining the information to appear on a display. Up to 25 different formats can be created. One or more buttons on the voice terminal are then assigned a format. When one of these buttons is pressed, the information associated with the format is displayed on the voice terminal.

There is a limit to the number of VuStat buttons that can be administered with the "ID " field filled in. For detailed instructions on administering VuStats, or for completing the VuStats Display Format form, see the *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

As stated, the format defines what information appears on the display. Specifically it determines: what text and database items are shown on the display; how the information looks on the display; and when the display updates. The following sections detail the information that can be administered to appear on the VuStats display.

What Information Appears on the Display

Object Type

To define the information that will appear on the display, the system administrator must first determine who will use the information or what object the information will describe. This is the object type. The object type can be an agent (or agent extension), split/skill, trunk group or vector directory number (VDN). Each object type is associated with specific information in the BCMS database tables. This object type is entered on the “VuStats Format Display” form and controls the types of information that are available for display. The following list describes each of these object types.

- Agent — The agent object type provides agents with their own statistics, or statistics about the splits/skills they log into.
- Agent-extension — The agent-extension object type provides supervisors or other users with statistics about agents or the splits/skills the agents log into. With an object type of agent-extension, the VuStats display can be administered to automatically show statistics for a specific agent. Or, the supervisor can enter the ID of any agent they want to review.
- Split — The split object type is used to display statistics about a specific split/skill. Much of the information available with a split object type is also available with the agent or agent-extension object type. The selection of object type will depend on the application.
- Trunk-group — The trunk-group object type is used to display statistics about a specific trunk group.
- VDN — The VDN object type is used to display statistics about a specific VDN.

Data Type

Once the object type is selected, the system administrator must define what data should be displayed for that object. For example, with an object type of agent, VuStats could display information of interest to an agent, including: the total number of calls the agent has answered since logging in, the average time the agent has spent on each ACD call, the number of agents available to receive calls for a split, or the percent in service level for a given split.

On the other hand, for an object type of split, VuStats could display information describing the split and performance for the split. For example: the average speed of answer; the number of calls waiting; or information about agent work states.

The data displayed for each object is defined with a data type, which is entered on the “VuStats Format Display” form. A maximum of 10 data types can be entered for each display. For a complete description of each data type see "Tables of Data Types" in the “VuStats” section in the *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

Interval

With the exception of two agent shift data items, in G3V3 the information displayed on the terminal only represents data collected during the current interval. In other words, if VuStats displays the average After Call Work (ACW) time, it represents the average amount of time the agents on the split have spent in ACW during the current interval. Likewise, if VuStats displays the number of

ACD calls, it is the number of calls received during the current interval. The interval is either 1/2 or 1 hour depending upon the value administered for the BCMS measurement interval on the “Feature-Related System Parameters” form.

With G3V4 and later releases, VuStats also can be administered to show historical data. It can show statistics that have accumulated for the day or for an administered number of intervals. For example, if VuStats displays the number of ACD calls for the past four completed intervals, it will display the number of ACD calls received in the past two hours (1/2 hour intervals) or four hours (one hour intervals) plus those completed during the current interval. The use of historical data can impact processor occupancy depending upon the number of active users, their update rates, and the number of historical data types.

With an object type of agent or agent-extension, shift data is available for the number of ACD calls answered and the average ACD talk time for that agent. Shift data accumulates during the time an agent is logged in. If the agent logs out of all splits, shift data clears. For this reason, if shift data is required, agents should enter AUX work mode when taking breaks rather than logging out.

Thresholds

With G3V4 many data types can be administered with a threshold comparator and value. When the condition defined by the threshold is true, and the data type is shown on the display, the VuStats button lamp flashes. For example, suppose a format is created in which the oldest call waiting data type is administered with a threshold of >= (greater than or equal to) five minutes. Whenever that VuStats format is displayed, if the oldest call in queue has been waiting for five minutes or longer, the VuStats lamp flashes on the voice terminal. Each time the display updates the threshold is checked for each data type being displayed.

Text

Text associated with data can be entered on the “VuStats Display Format” form. This text will appear on the display to identify the data. For example, in the illustration at the beginning of this section, "AUX=" was entered as text on the “VuStats Display Format” form to identify the data type "split-agents-in-aux" (the number of agents currently in AUX work mode for a specified split). Text is optional. When used, it appears exactly as entered on the form.

Remembering that each display is limited to 40 characters, it is important to limit the amount of text, and to use abbreviations when possible. For example "S=" could indicate that the number following the equals sign is the split number.

Display Linking

Display formats can be linked so that users can step through a series of displays thereby increasing the amount of information they can view. For example, a display providing information for the agent’s first split could be linked to a second display providing the same information for the agent’s second split. Or, a split supervisor might want one display providing information about the work states of all agents on a split linked to a second display that provided information about calls waiting, number of calls abandoned, or the oldest call waiting for the split.

To allow agents to view linked displays, a next button must be administered on the voice terminals. Displays should have the same object type if they are linked.

For detailed information to administer what will appear on the display, see "What Information Appears" in the VuStats section in the *DEFINITY Communications System Generic 3 Version 4 Implementation, 555-230-655,* or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

How the Information Looks

VuStats statistics appear on the second line of two-line DCP terminal displays or on the first line of one-line DCP terminals and all BRI terminals. For voice terminals with 2 x 24 displays, the display automatically wraps to the second line of the display. When VuStats is activated, it overwrites and cancels any display feature on the second line of a two-line display and on the first line of a one-line display.

The format of the information on the display is defined exclusively on the “VuStats Display Format” form. Using this form, the system administrator defines:

- The text associated with each data type and the amount of space left for each piece of data. For example, "AUX=$$$" indicates the text AUX= will appear with 3 spaces left for the data associated with this text.
- The order in which the data types will appear on the display.
- The format for time related data types. For example, whether the data will display as hours rounded to the nearest hour, minutes rounded to the nearest minute, or minutes and seconds.
- The display that should follow the current one when the next button is pressed.

Name database items, for example the name of a split or VDN, will be truncated on the right to fit the data field size specified for the data type. Numerical database items will display asterisks if the number is too large to fit the data field size specified for the data type.

For detailed information to administer how the information will look on the display terminal, see "How the Information Looks" in the VuStats section in Chapter 3 of the *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

When the Information Updates, or Vanishes

When activated, any display feature that uses the second line of a two-line display, or the first line of a one-line display will overwrite and cancel the VuStats feature.

The VuStats display can be cleared by pressing the normal button. The VuStats button lamp will go dark, and the display will return to normal mode.

The system administrator defines how long the VuStats display will remain on the terminal, and how frequently the information on the display will update.

VuStats can be administered to display information continuously until the agent presses the normal button or another operation overwrites the VuStats display. Or, the display can be administered to only appear for an interval of 5, 10, 15, or 30 seconds and then disappear. When this VuStats display interval ends, the display automatically returns to normal mode.

Continuous displays can be administered to update the displayed VuStats statistics every 10, 20, 30, 60 or 120 seconds. Or, statistics can update every time an agent changes work mode, every time a BCMS Measurement Interval is completed on the hour or half-hour, or not update at all.

For detailed information to administer when the information updates, and for how long it appears, see 'When the Display Updates" in the VuStats section in the *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

End User Operation

Information about end user operation of the VuStats feature is also included in *DEFINITY Communications System Generic 3 Automatic Call Distribution (ACD) Agent Instructions, 5*55-230-722, and *DEFINITY Communications System Generic 3 Automatic Call Distribution (ACD) Supervisor Instructions,* 555-230-724. The following sections give some pointers on how agents and supervisors will interact with the VuStats feature.

Agent Operation (Agent Object Type)

VuStats display formats created to allow agents to view their own performance statistics are created with the agent object type. Therefore, to view statistics the agent simply presses the VuStats button on the voice terminal. The display shows the statistics for that agent as defined in the display format assigned to the button. The agent must be logged in at the time the VuStats button is pressed if EAS or BCMS/VuStats loginIDs are used. When VuStats is active the VuStats button lamp stays lit. Or, if a threshold is met, it will flash. The statistics will update based on the administered update interval.

On a two-line DCP display, VuStats statistics appear on the second line. Any information currently on the second line of the display is overwritten by VuStats. On a one-line display or BRI terminal, VuStats statistics overwrite the entire display line. To return to incoming call information, the agent presses the normal button.

If the display format links to another display, the agent can press the next button to see additional information.

If the VuStats button is pressed at a terminal where an agent is not logged in when EAS or BCMS/VuStats loginIDs are in use, the button lamp flashes at a broken flutter indicating access to the feature is denied. An error message on the display indicates that the agent must log in.

If the button lamp flashes at a broken flutter for two seconds and no error message is displayed, it indicates that the maximum number of users are already displaying VuStats.

If the display shows asterisks instead of data, it indicates that the value of the data was too large to fit in the number of digits allocated in the display format.

Supervisor Operation (Agent-extension, split, VDN, or Trunk Group Object Type)

When a supervisor or other user accesses VuStats statistics for an agent-extension, split, VDN, or trunk group, one of two conditions can exist.

1. The VuStats button can be administered with an ID indicating the specific agent, split, VDN, or trunk group the information describes.
2. The user enters, on the dial pad, the ID or extension for any agent, split, VDN, or trunk group they want to monitor.

Administered ID

If the ID is administered for the button, the supervisor simply presses the VuStats button on the voice terminal. The display shows the statistics for that agent, split, VDN, or trunk group as defined in the display format and ID assigned to the button.

When VuStats is active, the VuStats button lamp stays lit. Or, if a threshold is met, it will flash. The statistics update based on the administered update interval.

On a two-line DCP display, VuStats statistics appear on the second line. Any information currently on the second line of the display is overwritten by VuStats. On a one-line display or BRI terminal, VuStats statistics overwrite the entire display line. The normal button cancels VuStats.

If the display format links to another display, the supervisor can press the next button to see additional information.

If the VuStats button is administered for an invalid agent, split, VDN, or trunk group, the button lamp flashes at a broken flutter indicating access to the feature is denied. An error message on the display indicates invalid administration. This should be reported to the system administrator.

If the button lamp flashes at a broken flutter for two seconds and no error message is displayed, it indicates that the maximum number of users are already displaying VuStats.

If the display shows asterisks instead of data, it indicates that the value of the data was too large to fit in the number of digits allocated in the display format.

ID Not Administered

If the ID is not administered, the user first presses the VuStats button. The button lamp lights and the user receives dial tone. The user enters the agent extension or login ID, split number, VDN extension, or trunk group number followed by the pound sign. If a valid ID or extension is entered, confirmation tone is returned and the VuStats statistics appear on the display.

When VuStats is active the VuStats button lamp stays lit. Or, if a threshold is met, it will flash. The statistics update based on the administered update interval.

On a two-line DCP display, VuStats statistics appear on the second line. Any information currently on the second line of the display is overwritten by VuStats. On a one-line display or BRI terminal, VuStats statistics overwrite the entire display line. The normal button cancels VuStats.

If the display format links to another display, the supervisor can press the next button to see additional information.

If the user enters an invalid agent extension or login ID, split number, VDN extension, or trunk group number, the button lamp flashes at a broken flutter indicating access to the feature is denied. An error message on the display indicates an invalid entry.

If the button lamp flashes at a broken flutter for two seconds and no error message is displayed, it indicates that the maximum number of users are already displaying VuStats.

If the display shows asterisks instead of data, it indicates that the value of the data was too large to fit in the number of digits allocated in the display format.

Error Messages

The following error messages may display on the voice terminal when VuStats is being used.

Table 3-86. VuStats Display Error Messages

Error Message	Condition
FORMAT [xx] NOT DEFINED where xx is the format number assigned to the button.	The VuStats button is pressed but the display format assigned to the button is not defined. (The “Format Description” field on the “Display Format” form is blank.)
FORMAT [xx] DOES NOT ALLOW OR REQUIRE ID where xx is the format number assigned to the button.	The VuStats button is pressed, agent is the object type of the format, and the ID associated with the button is defined (administered on the “Station” or “Attendant” form).
AGENT NOT LOGGED IN	The VuStats button is pressed, agent is the object type of the format and no agent is logged in on the station set.
[object type] [xxxx] NOT ADMINISTERED where object type is AGENT, SPLIT/SKILL, TRUNK GROUP, or VDN and xxxx is the ID assigned to the button or entered by the user.	The VuStats button is pressed, and the ID or extension assigned to the button or entered by the user has not been administered (does not exist in the system).
[object type] [xxxx] NOT MEASURED where object type is AGENT, SPLIT/SKILL, TRUNK GROUP, or VDN and xxxx is the ID assigned to the button or entered by the user.	The VuStats button is pressed, and the ID or extension assigned to the button or entered by the user is not measured (is not administered as measured on the appropriate “Hunt Group”, “VDN”, or “Trunk Group” form).

Call Center Example

Because of its flexibility VuStats can be used in a variety of ways, meeting the requirements of multiple users with different applications. This call center example is just one representative sample of a VuStats application.

The following illustration shows the structure of a sample call center.

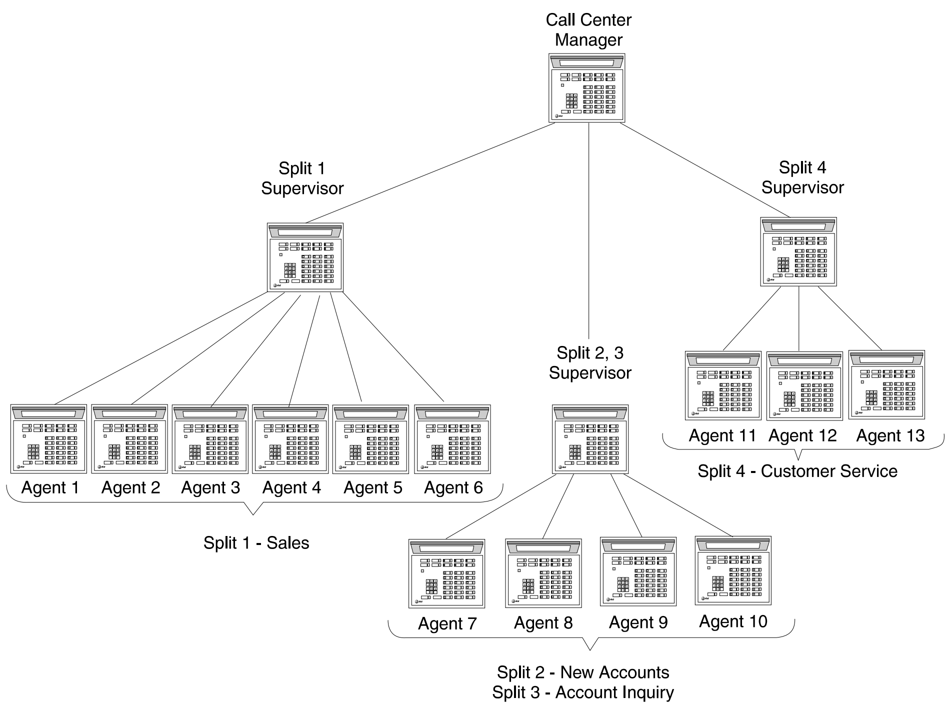


Figure 3-32. Sample Call Center

In the call center pictured above, the following conditions are true:

- If the split queue is full, calls to split 1 will **route-to** split 2 or to split 3 and visa versa. (All agents in these splits are trained to handle all three types of calls.)
- Agents in split 4 do not receive calls from other splits.
- Two VDNs **route-to** split 1. One of these VDNs, named "elite," is for high volume sales customers.
- Agent 12 is used as a backup for split 1. Under certain conditions, the agent goes into AUX work mode on split 4 and logs into split 1.

The following sections describe the VuStats display requirements for each group, agent 12, and the three split supervisors.

Split 1

Split 1 has the objective of processing 40 sales calls per hour. To evaluate their success towards reaching this goal, agents might want to know:

- The objective.
- The rate at which they have been answering calls.
- The rate at which all members of the split have been answering calls.
- How their average time spent talking on each ACD call compares to the average for other agents on the split.

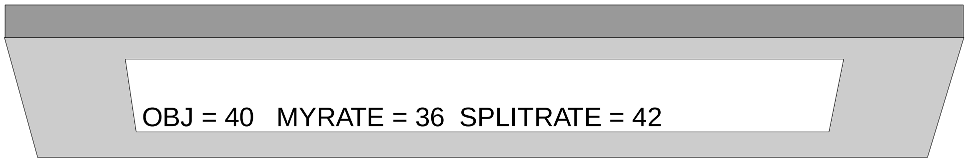
Agents also would like to be alerted when the call rate for the split drops to 20 or fewer calls. The display will be administered to cause the VuStats button to flash if the rate drops this low.

Agents in this company also decide independently when to go into AUX work mode to take breaks or complete work not directly related to calls, such as copying or filing. To choose a good time to enter AUX work mode, the agent might want to know:

- The number of agents staffed.
- The number of agents available.
- The number of agents in AUX work mode.
- The number of calls waiting.
- The percent of calls within the acceptable service level.

Each agent in split 1 will require only 1 VuStats button. Three formats will be required to present the information requested.

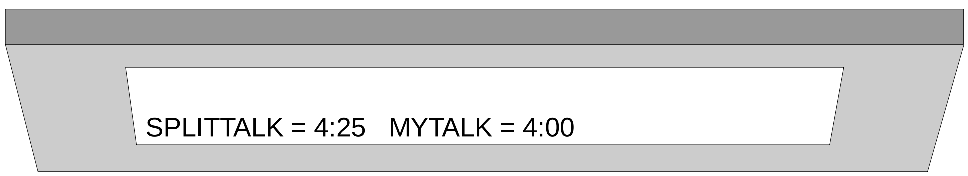
Format 1



(The split objective is 40 calls per hour. The agent is handling calls at a rate of 36 per hour. The split is handling calls at a rate of 42 per hour.)

NEXT

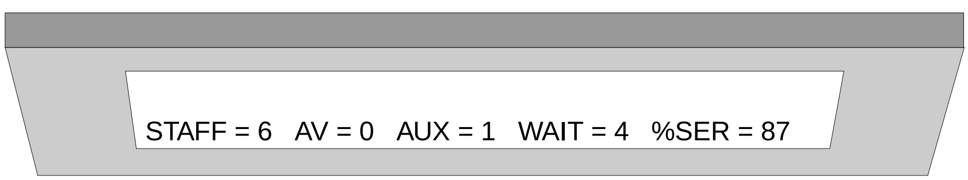
Format 2



(The average talk time per call for the split is 4:25, the average talk time per call for this agent is 4:00.)

NEXT

Format 3



(For the split: agents staffed equals 6; agents available equals 0; agents in AUX work mode equals 1; calls waiting in queue equals 4; and percent of calls answered within the acceptable service level equals 87%.)

Figure 3-33.   Split 1 Displays

Splits 2 and 3

Agents in splits 2 and 3 have not defined objectives for their splits. However, agents still want to review basic performance statistics including:

- The total ACD calls the agent has answered during the day.
- The average speed of answer during the day for each split the agent is logged into.

The agents also want to be able to view statistics to determine when to enter AUX mode. In this example a separate display is set up for each split.

Each agent will require only 1 VuStats button. Three formats will be required to present the information requested.

Format 4

ACD = 30   ASA 2 = 1:02   ASA 3 = 0:53

(The number of ACD calls answered by this agent during the day is 30; the average speed of answer for split 2 is 1:02 minutes, and the average speed of answer for split 3 is 53 seconds.)

NEXT

Format 5

S = 2 STAFF = 4   AV = 0   AUX = 0   WAIT = 6 %SER = 90

(In split 2 agents staffed equals 4; agents available equals 0; agents in AUX work mode equals 0; calls waiting in queue equals 6; and percent of calls answered within the acceptable service level equals 90%.)

NEXT

Format 6

S = 3 STAFF = 4   AV = 0   AUX = 0   WAIT = 8   %SER = 87

(In split 3 agents staffed equals 4; agents available equals 0; agents in AUX work mode equals 0; calls waiting in queue equals 8; and percent of calls answered within the acceptable service level equals 87%.)

Figure 3-34. Splits 2 and 3 Displays

Split 4

Split 4 has the objective of maintaining 95 percent of calls answered within the acceptable service level. To meet this objective agents need to know:

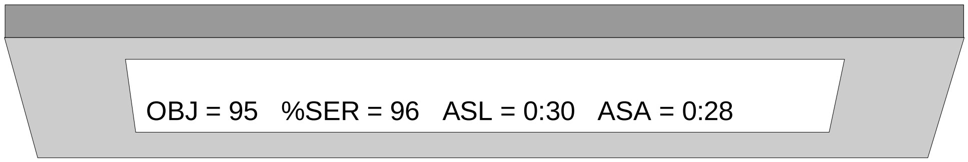
- The objective.
- The current percent of calls within the acceptable service level.
- The number of seconds within which calls must be answered to be acceptable.
- The current average speed of answer for the split.

Agents also would like to be alerted when the percent within service level drops below 90 percent. The display will be administered to cause the VuStats button to flash if the percent drops this low.

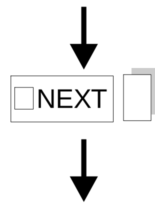
The agents also want to be able to view statistics to determine when to enter AUX mode.

Each agent in split 4 will require 1 VuStats button. Two formats will be required to present the information requested.

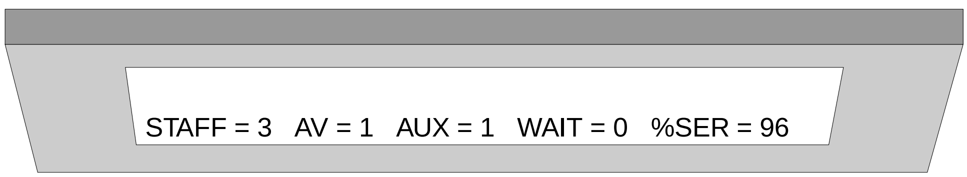
Format 7



(The split objective is 95% of calls answered with the acceptable service level, the percent of calls answered within the acceptable service levels equals 96%, the number of seconds within which a call must be answered to be within the acceptable service level is 30, and the average speed of answer for the interval is 28 seconds.)



Format 8



(For the split: agents staffed equals 3; agents available equals 1; agents in AUX work mode equals 1; calls waiting in queue equals 0; and percent of calls answered within the acceptable service level equals 96%.)

Figure 3-35. Split 4 Displays

Agent 12

Agent 12 acts as a backup for split 1 and so has special VuStats requirements. Agent 12 will enter AUX mode on split 4 and log into split 1 when: no agents are available on split 1; more than nine calls are waiting in the split 1 queue; and at least one other agent is available in split 4. Therefore agent 12 must know:

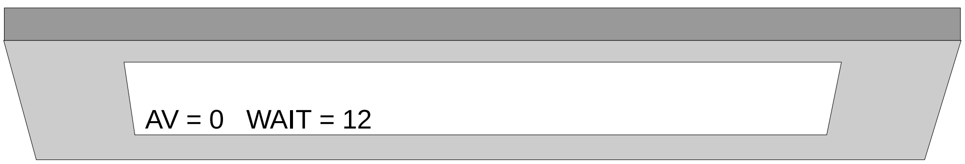
- The number of agents available on split 1.
- The number of calls waiting on split 1.
- The number of agents available on split 4.

The agent can determine the number of agents available on split 4 by viewing format 3 while logged into split 4.

Because the agent is not logged into split 1 when the information is requested, an object type of split is used to access split 1 information. A second VuStats button is used for the split display. The button can be administered with the ID for split 1 to eliminate the need to enter the split number when accessing this display and, if necessary, to prevent the agent from viewing statistics for other splits.

One additional format will be required to present split 1 statistics to agent 25.

Format 9



(For split 1: the number of available agents is 0; the number of calls waiting in queue is 12.)

Figure 3-36. Agent 25 Split 1 Display

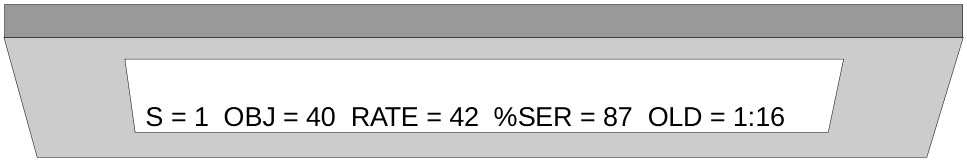
Split Supervisors

All three supervisors are interested in receiving statistics for the splits they manage including:

- The objective for the split.
- The rate at which calls are being answered for the split.
- The percent of calls answered within the acceptable service level.
- The oldest call waiting for the split.

Each supervisor will require 1 VuStats button. One format will be required to present the information requested.

Format 10



(This example shows the display for the split 1 supervisor. The split number is 1; the objective is 40 calls per hour; calls are being answered at a rate of 42 per hour; the percent in service level is 87; the oldest call in the split has been waiting 1 minute and 16 seconds.)

Figure 3-37. Split Supervisor Display

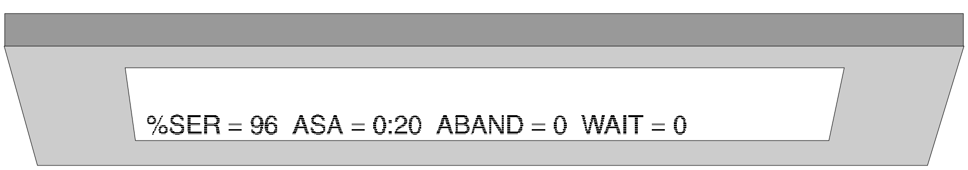
Supervisor 1

Supervisor 1 is also concerned with monitoring the "elite" VDN that routes to split 1. Since this VDN is only accessed by high volume clients, the supervisor wants to be certain that service is always available. Therefore the supervisor wants to know:

- The percent of calls answered within the acceptable service level.
- The average speed of answer for a call coming in on the VDN.
- If any calls have abandoned.
- If any calls are waiting.
- How long the oldest call in the queue has been waiting.

A second VuStats button and one additional format are required for the VDN display.

Format 11



(The percent of calls answered within the acceptable service level is 96; the average speed of answer for a call on the split is 20 seconds; no calls have abandoned the split during the current interval; no calls are waiting in the queue.)

Figure 3-38. Supervisor 1 VDN Display

Sample Administration

To implement this call center example, a system administrator would need to create a total of eleven VuStats display formats: eight with an object type of agent; two with an object type of split; and one with an object type of VDN. See "Call Center Example" in the VuStats section of the *DEFINITY Communications System Generic 3 Version 4 Implementation,* 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for instructions about how this example could be administered.

Considerations

Some of the VuStats data is accumulated for an agent's login session. If agents log in in the morning and then log out for lunch, they end their sessions and the system clears their accumulated data. When agents log in after lunch, they begin new sessions. To accumulate a full day's statistics, agents and/or supervisors must keep a running total of all the agents' login sessions. To circumvent the need to keep track of login sessions use historical data or have agents go into AUX mode when they are on break.

The DEFINITY systems have varying limits on how many measured agents or login IDs can be monitored and how many terminals can display VuStats data simultaneously. See Appendix A, "System Parameters" for capacities.

Interactions

- **BCMS**

  If BCMS is not active, you cannot receive BCMS reports. The VuStats feature displays data collected by BCMS, but you do not need to have BCMS enabled in order to use VuStats. VuStats operates normally without BCMS turned on.

- **Call Prompting**

  A Call Prompting digits display automatically cancels the VuStats feature. The Call Prompting display overwrites the VuStats display and the VuStats button lamp extinguishes. When the agent reactivates VuStats, the VuStats display overwrites the Call Prompting display.

- **Call Work Codes**

  When the call work code (CWC) is activated, it automatically cancels the VuStats feature. The CWC display overwrites the VuStats display and the VuStats button lamp extinguishes.

  If VuStats is activated while a CWC is being entered— that is, the pound (#) sign is not yet dialed— the CWC display is overwritten. In this case, the CWC is not sent and must be reentered.

- **Change Skills**

  An agent changing skills automatically cancels the VuStats feature. The display of the new skills overwrites the VuStats display and the VuStats button lamp extinguishes. When the agent reactivates VuStats, the VuStats display overwrites the display of the new skills.

- **Integrated Directory**

  If a user activates the Integrated Directory feature, the VuStats feature is automatically cancelled. The Integrated Directory display overwrites the VuStats display and the VuStats button extinguishes. When VuStats is reactivated, the VuStats display overwrites the Integrated Directory display.

- **Move Agents from CMS**

  If an agent moves from one split or skill to another, the change does not affect the ID administered to the VuStats (vu-display) button.

  If an agent is moved from one split or skill to another, the switch does not reassociate VuStats buttons (associated with the agents previous split or skill) to the new split or skill. Therefore, in a call-environment in which agents are often moved between splits or skills, VuStats buttons should not have an assigned association with a specific split or skill. Instead, a VuStats button associated with the agent format (without an ID) should be assigned to each agent's voice terminal, and a split reference can be used to view the agent's first, second, third, or fourth split/skill.

- **Queue-Status Indications**

  The queue-status button display automatically cancels the VuStats feature. The queue-status display overwrites the VuStats display and the VuStats lamp extinguishes. When VuStats is reactivated, the VuStats display overwrites the queue-status display.

- **Service Observing**

  On terminals with a one-line display, the Service Observing button display automatically cancels the VuStats feature. The Service Observing display overwrites the VuStats display and the VuStats lamp extinguishes. When VuStats is reactivated, the VuStats display overwrites the Service Observing display

Administration

In order to use VuStats, Automatic Call Distribution (ACD) and VuStats must be optioned on the “System-Parameters Customer-Options” form. Hunt Groups, Trunk Groups, and VDNs must also be administered as measured ‘internal’’ or ‘‘both.’’

Use the “VuStats Display Format” form to administer the display formats. Up to 25 different 40-character display formats can be created.

A feature button for ‘‘vu-display’’ must be administered on those attendant consoles and voice terminals that will display VuStats data. A ‘‘next’’ button must be administered for voice terminals that will display linked formats.

For detailed instructions for administering VuStats, or for completing the “VuStats Display Format” form, see the *DEFINITY Communications System Generic 3 Version 4 Implementation, 555*-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

Hardware Requirements

VuStats is only available for attendant consoles and voice terminals with digital display capability. VuStats is most effective on voice terminals with two-line displays (of 40 characters per line), such as the CallMaster voice terminals. This type of display allows users to see call appearances on the top line and VuStats on the bottom line. With other displays, VuStats overwrites call appearance information when a user presses the VuStats button.

Wideband Switching

Feature Availability

The Wideband Switching feature is optional with G3V2 and later releases.

Description

The Wideband feature provides you with the ability to dedicate 2 or more ISDN B-channels or DS0 endpoints for applications which require large amounts of bandwidth. The ISDN-PRI divides a T1 (E1 for international switches) trunk into 24 (31 for E1) information channels and one signaling channel for standard narrowband communication. Certain applications, such as video conferencing, require greater bandwidth, and it becomes necessary to aggregate several of those narrowband channels into one ‘‘wideband’’ channel to accommodate the extra bandwidth requirement. The DEFINITY switch may serve as a gateway to many types of high bandwidth traffic.

Applications involving the Wideband Switching feature are listed below:

- Video conferencing
- WAN Disaster Recovery
- Scheduled batch processing (for example, nightly file transfers)
- LAN Interconnections and Imaging
- Other applications involving high-speed data or video transmission or a high degree of bandwidth

NOTE:
For additional information on the DEFINITY system Wideband Switching feature, see the *DEFINITY Communications System Generic 3 Wideband Technical Reference*, 555-230-230, *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653.

Technical Description

The Wideband Switching feature provides end-to-end connectivity between endpoints at data rates from 128 to 1536 Kbps over a T1 facility, or up to 1984 Kbps over an E1 (European Standard) facility, according to the ITU-T and ANSI standards for 384 (H0, a 6 channel aggregate), 1536 (H11, a 24 channel aggregate or dedicated T1 trunk), and 1920 (H12, a 31 channel aggregate, or dedicated E1 trunk) Kbps rates and ITU-T and ANSI standards for NX64 Kbps rates (where *N* may equal 2 to 24 on a T1 trunk and 2 to 31 on an E1 trunk).

NOTE:
Typically, a T1 facility provides for 23 B-channels and 1 data channel while an E1 facility provides 30 B-channels and 1 D-channel. If however, Non-Facility Associated Signaling (NFAS) is used, a group of T1/E1 facilities can be configured to share a single data channel allowing all but one of the T1/E1 facilities to be configured with an extra B-channel.

DEFINITY switches can be configured for Wideband Switching for various uses, including the following:

- Sending data over a domestic network T1 facility, typically providing 1 D-channel and 23 B-channels.
- Sending data over an international network E1 facility, typically providing 1 D-channel and 30 B-channels.
- An ISDN-PRI line-side (switch to terminal) application

Each B-channel can provide 64 Kbps transmission rates. There are two service types for encoding the aggregation of channels, both of which are supported by the DEFINITY switch.

NX64 allows the aggregation of any channel groupings while H0, H11, and H12 have fixed channel assignments.

Table 3-87 summarizes the ISDN channel types with the associated data rates that support Wideband Switching.

Table 3-87. ISDN Channel Types and Wideband Associated Data Rates

Channel Type	Data Rate
H0 (6 channels)	384 Kbps
H11 (24 channels)	1536 Kbps
H12 (31 channels)	1920 Kbps
NXDS0 (2 to 31 channels)	128 to 1984 Kbps

Any endpoint with an ISDN-PRI interface can be administered for Wideband Switching applications. For both network and line-side use, a Universal DS1 (UDS1) circuit pack is the interface for carrying wideband calls. On the line side, an ISDN-PRI terminal adaptor such as the Paradyne Acculink Bandwidth Controller is used to support switched and permanent connections.

Channel Allocation

Wideband channel allocation is performed using one of three allocation algorithms: fixed, flexible, or floating. This subject is discussed in greater detail in the Implementation Guide, but in brief:

- Fixed allocation provides contiguous channel aggregation and the starting channel is constrained to a predetermined starting point. (Used only for H0, H11, and H12 calls.)
- Flexible allocation allows a Wideband call to occupy non-contiguous positions within a single T1 or E1 facility.
- Floating allocation enforces contiguous channel aggregation but the position of the first channel is not constrained like it is in fixed allocation.

Typical Uses

Many Wideband Switching applications are video applications such as video conferencing or data applications.

A typical video application is illustrated in Figure 3-39. The video application uses an ISDN-PRI interface to DS0 1 through 6 of the line-side facility.

NOTE:
A line-side facility is the use of ISDN-PRI between the switch and the terminal, in this case a video conference facility. Most lines between switch and audio terminals use the ISDN-BRI, and do not have the available bandwidth for this kind of application. Special line-side uses of ISDN-PRI have to be arranged as special implementation needs.

In actual use, the channels aggregated for Wideband use may not be contiguous and vary constantly.

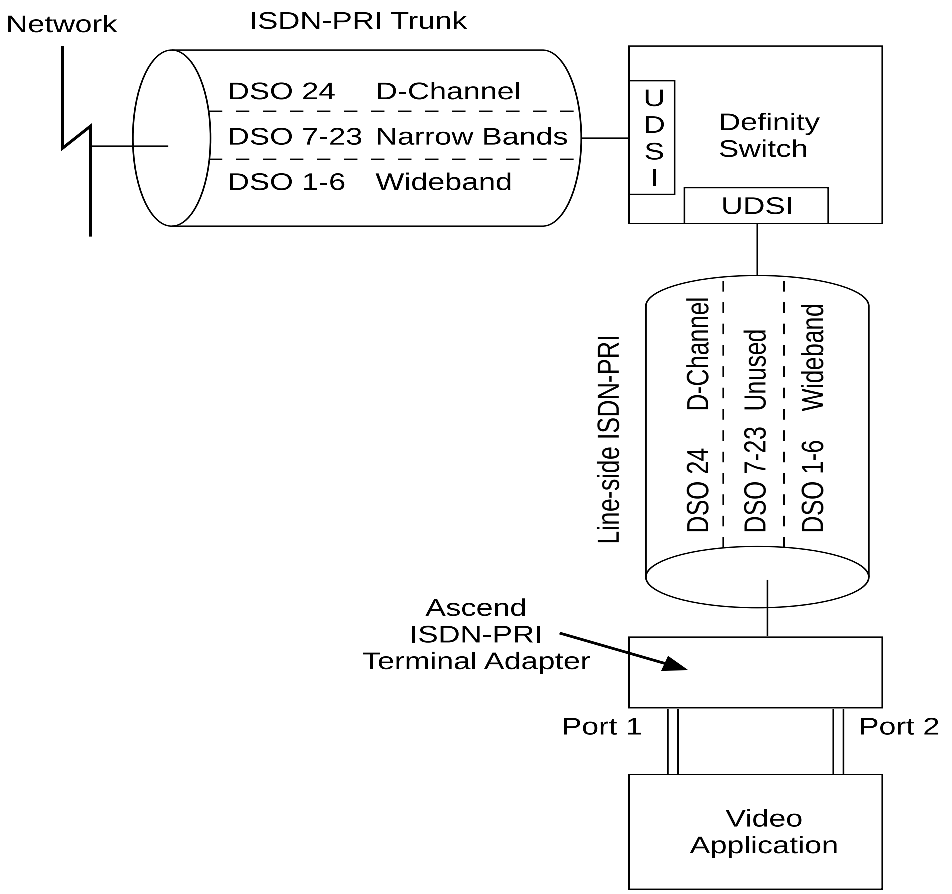


Figure 3-39. Wideband Switching Video Application

Considerations

Glare and blocking are the two most pertinent trunking considerations when using Wideband Switching.

Glare

Glare arises when both sides of an ISDN-PRI interface simultaneously select the same B-channel for call initiation. For example, the user side of the ISDN-PRI interface selects a B-channel for an outgoing call. Before the network receives the SETUP message, the network might select the same B-channel for a call. Even with channel negotiation schemes, glare might cause the call from the user side to be dropped. Since Wideband Switching uses multiple B-channels, the possibility of glare is greater than it is with other calls.

To reduce the possibility of glare, the DEFINITY switch should be administered so that each side of the ISDN-PRI interface select channels from opposite ends of the interface.

In other words, one end should be administered to select B-channels from the high side (DS0 channel 23) and the other administered to select B-channels from the low side (DS0 channel 1) so as not to interfere with each other. In addition, there are algorithms in the switch's operating code which attempt to gain control of channels closest to the starting DS0 so that Wideband channels are contiguous.

For example, if the user side is provisioned to start at the high side (DS0 23) and DS0 22 is idle but DS0 23 is active, DS0 22 should be reselected for the next call. This is known as linear trunk hunting. The DEFINITY switch uses the linear trunk hunt method and only the direction of hunt is administrable. See the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for additional details.

Blocking

Blocking occurs when the number of B-channels required to make a call are not available. Narrow band (64 Kbps) calls require only one channel so blocking is less likely than with Wideband calls which require multiple B-channels. Blocking also occurs for Wideband calls when bandwidth is not available in the appropriate format (that is, fixed, floating, or flexible, see earlier description).

To avoid this situation, the DEFINITY switch selects trunks for both Wideband and narrowband calls to maximize availability of idle fixed channels for H0, H11, and H12 calls and idle floating channels for NXDS0 calls. The strategy for preserving idle channels to minimize blocking depends on the channel type. The chances for blocking are reduced if the flexible algorithm is used.

Interactions

The following features interact with Wideband Switching:

- Administered Connections

  Provides call initiation for Wideband Access Endpoints (WAEs). All administered connections that originate from WAEs use the entire bandwidth administered for the WAE. The destination of an Administered Connection can be a PRI Endpoint.

- Automatic Circuit Assurance (ACA)

  Treats Wideband calls as single trunk calls so that only a single ACA referral call is made if an ACA referral call is required. The referral call is on the lowest B-channel associated with the Wideband call.

- Call Detail Recording (CDR)

  Wideband calls trigger CDR records containing the Wideband Bearer Capability Class (BCC) and the call bandwidth.

  Both access endpoints and PRI endpoints involved in a call are treated as line-side entities in that they do not trigger a standard CDR record. Intra-switch CDR may be used to trigger intra-switch records involving PRI endpoints. However, the intra-switch CDR capability is automatically disabled as soon as a trunk is involved with the call. Therefore, trunk-to-endpoint and endpoint-to-trunk calls must be handled by trunk group administration using standard CDR records.

- Call Management System (CMS)

  Wideband calls may be carried over trunks that are measured by the CMS but Wideband endpoints are not measured by CMS.

- Call Vectoring

  Wideband endpoints may use a vector directory number when dialing. In other words, PRI endpoints may dial a Vector Directory Number (VDN) when Vector Routes lead to other Wideband endpoints. For example PRI endpoint 1001 dials VDN 500. VDN 500 points to Vector 1. Vector 1 can point to other PRI endpoints such as **route-to** 1002, or **route-to** 1003, or **busy**.

  This is only used by certain specific applications. Typically when an incoming Wideband call to a DEFINITY switch wants to hunt for an available Wideband endpoint, the call could **route-to** a VDN which would then send the call to the first available Wideband endpoint according to the vector routing program.

- Class of Restriction (COR)

  The COR identifies calling and called party privileges for the Wideband endpoint(s). Administer the COR in such a way that Account Codes are not required. Also, Forced Entry of Account Codes should be turned off for Wideband endpoints.

- Class of Service (COS)

  The COS determines the class of features that a Wideband endpoint can activate. Administer the COS in such a way as to block the activation of Call Forwarding.

- Facility and Non-Facility Associated Signaling

  Non-Facility Associated Signaling with or without the D-channel backup (DCBU) requires administration via Signaling Groups for trunk-side Wideband interfaces (not supported on the line-side).

- Facility Busy Indication

  It is possible to administer a Busy Indicator button for a Wideband endpoint extension but the button does not accurately track the endpoint status.

- Facility Test Calls

  Facility Test Calls can be used to perform loop-back testing of the Wideband call facility.

- Generalized Route Selection

  Generalized Route Selection supports a Wideband Bearer Capability Class (BCC) to identify Wideband calls. The Generalized Route Selection feature searches a routing pattern for a preference that has the Wideband BCC. Routing preferences that support Wideband BCC also support other BCCs to allow different call types to share the same trunk group.

- Integrated Services Digital Network — Primary Rate Interface

  Standard ISDN-PRI interfaces can be administered for Wideband switching.

- Leave Word Calling

  Dialing a Wideband endpoint extension as part of the Leave Word Calling feature applies an intercept to the activator.

- Subnet Trunking

  Subnet Trunking is accomplished via Generalized Route Selection and Routing Pattern administration.

Administration

NOTE:
Refer to the *DEFINITY Communications System Generic 3 Version 4 Implementation*, 555-230-655, or *DEFINITY Communications System Generic 3 V2/V3 Implementation*, 555-230-653, for procedures describing how to administer DEFINITY system features to include Wideband Switching; most importantly see the section on administering ISDN-PRI.

The following forms have to be administered for Wideband service:

- Access Endpoint
- PRI Endpoint
- Trunk Group (ISDN-PRI)
- Routing Patterns

Access Endpoint

Access endpoint administration is required to support Wideband access endpoints (WAEs). WAEs are administered by defining the starting DS0 and the number of DS0s that comprise the WAE.

PRI Endpoint

PRI endpoints are administered to define the number of DS0s to be associated with an extension. The PRI endpoint's number of DS0s (that is, the width) is limited by the type of ISDN-PRI interface but is always between 1 and 31. If the PRI endpoint is administered for a width narrower than the maximum number of

B-channels supported by the line-side facility, then multiple PRI endpoints can be administered within that same line-side facility. The parameters H0, H11, H12, and NXDS0 must be administered to identify the algorithm that the PRI endpoint uses to provide Wideband service. If the NXDS0 algorithm is administered, then administration to identify it as floating or flexible (on the "PRI-Endpoint" form contiguous ***y/n***)is also required. All combinations of these are allowed except floating NXDS0 with H0. Other parameters can be administered for each PRI endpoint; they include: COR, COS, and whether Auto Restoration should apply to calls originating from the PRI endpoint.

Trunk Groups

Trunk group administration is required to specify the Wideband service(s) that are to be supported by each trunk group. The parameters H0, H11, H12, and NXDS0 must be administered to identify the algorithm that the trunk group uses to provide Wideband service. If the NXDS0 algorithm is administered, then administration to identify it as floating or flexible (on the "Trunk Group" form contiguous ***y/n***)is also required. All combinations of these are allowed except floating NXDS0 with H0. In addition, the trunk search direction must be administered for the direction that a trunk group searches B-channels within its facilities (descend — ascend or ascend — descend).

Routing Patterns

Routing parameter administration must be performed to specify the Wideband Bearer Capability Class (BCC). This administration allows Generalized Route Selection (GRS) to allow Wideband traffic to choose specific trunk preferences, namely those trunks supporting Wideband network services. In no way does this administration imply that Wideband and narrowband traffic must be carried separately. In fact, any combination of narrowband BCC can be optioned with the Wideband BCC.

Customer Option Form

Wideband switching must also be administered using the ''Customer Option'' form. The other Wideband parameter administration is only performed if Wideband switching is administered for the switch.

Hardware and Software Requirements

Wideband software is optional. The TN464C (Universal DS1) circuit pack or later must be used on both the line-side and network side of the switch for any facility that carries Wideband traffic. The only other additional hardware/software required for Wideband switching is that the PRI endpoints must adhere to the ISDN-PRI Wideband interface requirements and to PRI line-side requirements.

World Class Basic Rate Interface (BRI)

Feature Availability

Available with G3V4 and later releases

Description

World Class BRI (WCC BRI) provides an international BRI platform that offers multiple protocol options to meet specific country and application requirements. It provides access to Video Conferencing, Desktop Video Conferencing, Data Transmission, and other non-voice based applications that use BRI as a communication interface. Voice access is not supported for G3V4 though voice features are not blocked for WCC BRI terminals.

WCC BRI devices must be administered as the new type "wcbri." WCC BRI users select a country protocol for each terminal that will use the feature. This selection determines both the codeset modifications required to meet the national standards as well as the terminal initialization procedures if required.

WCC BRI supports the following country protocols:

- Bellcore National ISDN-1 protocol in the United States (TR268)
- National protocols in Australia (AUSTEL TS013, Telecom Australia TPH 1962), Japan (NTT BRI) and Singapore (FETEX 150 TIF 218)
- ETSI NET 3 protocol (ETS 300 102) for use in most of Europe.

WCC BRI supports multipoint (up to two devices per port) only for the Bellcore National ISDN-1 Country Protocol option.

Considerations

All devices administered on the same port must be of the same Country Protocol. All devices connected to the same DEFINITY system must use the same companding mode as the system TDM bus.

Interactions

- ASAI

  ASAI will continue to be accessed via current ASAI administration. ASAI should not be accessed via "wcbri" type administration.

Administration

The system administrator must specify in the “Endpt Init?” field on the “Data Module” or “Station” forms whether the administered WCC BRI device supports Bellcore NI-1 endpoint initialization procedures. This field should be set to “y” for all Bellcore NI-1 devices, and the field should be set to “n” for all other WCC BRI devices (Australia, Japan, Singapore and ETSI). If the “Endpt Init” field is set to “y” then the administrator must also specify an Endpoint ID. In conjunction with the SPID, the Endpt ID allows for multipoint configuration conformance to Bellcore terminal initialization procedures.

A Country Protocol also must be specified for each terminal.

Hardware and Software Requirements

WCC BRI requires a TN556B or later suffix BRI port circuit pack. A TN2198 two-wire BRI port circuit pack can be used in place of the TN556. In this case an NT1 is also required.

WCC BRI supports terminals conforming to country protocols specified under “Description”.

World Class Tone Detection

Feature Availability

This feature is available with G3i-Global and Generic V2 and later releases.

Description

World Class Tone Detection allows the DEFINITY switch to identify and handle different types of call progress tones, depending on the system administration. The tone detector and identification is used to display on Data Terminal Dialing and for deciding when to send digits on trunk calls through Abbreviated Dialing, ARS, AAR, and Data Terminal Dialing.

- ‘‘Tone detect mode 1’’ designates countries that use the same tone plan as Italy.
- ‘‘Tone detect mode 2’’ designates countries that use the same tone plan as Australia.
- ‘‘Tone detect mode 3’’ designates countries that use the same tone plan as the United Kingdom.
- ‘‘Tone detect mode 4’’ designates countries that use dial tones between 345 Hz and 625 Hz.
- ‘‘Tone detect mode 5’’ designates countries that use dial tones between 345 Hz and 1190 Hz.
- “Tone detect mode 6” designates countries that use the same dial tone plan as the United States.
- The ‘‘level of tone detection precise’’ is used in countries whose tone plan matches the tone board’s tone detect mode.
- The ‘‘level of tone detection medium’’ is used in countries that, except for the continuous dial tone and discontinuous other tones, have tones with characteristics that do not match those expected by the tone detector board’s detect mode.
- The ‘‘level of tone detection broadband’’ is used in countries that have a discontinuous dial tone.

Considerations

When the administered Level of Tone Detection is ‘‘medium’’ or ‘‘broadband,’’ multiple-line Data Terminal Dialing is disabled.

Interactions

The following features interact with the World Class Tone Detection feature.

- Data Call Setup Multiple-Line Data Terminal Dialing is supported ONLY if the administered Level of Tone Detection is ‘‘precise’’.
- The feedback displayed by Data Terminal Dialing in response to received tones is affected by settings of World Class Tone Detection.
- Abbreviated Dial, AAR, ARS, and Data Terminal Dialing can specify a wait for dial tone in outpulsing digits over a trunk. The World Class Tone Detection feature should be configured to match the characteristics of the dial tone that will be detected.

Administration

Administration of World Class Tone Detection consists of two distinct parts: tone detection level and tone detection algorithm. System Aministrators address how tone detection occurs and when it occurs in the outpulsing sequence. The following specific items are administered:

- “Level of Tone Detection” and “Off Premises Tone Detection Timeout” fields are administered on the “Feature-Related System Parameters” form.
- “Tone Detection Mode,” “Dial Tone Validation Timer” (for Tone Detection mode 4 and 5 only), “Outpulse Without Tone y/n,”and “Interdigit Pause” fields are administered on the “System-Parameters Country-Options” form.
- The “AAR/ARS Route Pattern” forms allows insertion of special characters “+” and “!” that indicate when to wait for a dial tone.
- The “Abbreviated Dialing” form allows entry of ~w and ~W that also indicate when to wait for a dial tone.

Hardware and Software Requirements

Tone Detection Modes 1, 2, and 3 are meaningful only if the system tone detectors are TN420Bs or greater.

Modes 4 and 5, the Dial Tone Validation Timer, and the Interdigit Pause are meaningful only if the system tone detectors are TN744C-Tone Detector/Call Classifier, TN420C-Tone Detector, or TN2182-Tone Clock/Detector/Generator.

Tone detection for countries using the same tone plan as the United States is also available using an alternate tone detector.

TN420B only recognizes Tone Detection Modes 1,2,and 3. TN420C, TN744C, and TN2182 recognize modes 1,2,3,4,5,and Dial Tone Validation and Interdigit Pause. TN2182 and TN744C recognize mode 6. TN748C provides precise tone detection for the United States tone plan.

World Class Tone Generation

Feature Availability

This feature is available with G3i-Global, all Generic V2, and later releases.

Description

Allows administrators to specify the base call progress tone set to be generated by the system and to then customize the set by selecting different values for frequency and cadence for up to 6 individually administrable tones.

Considerations

If a particular tone (for example, conference tone) is not defined for the administered base tone generation set (and not administered via the individual tone administration), silence will be used.

Interactions

Brief call waiting tones are optimal because, while the tone is sounding, speech cannot be heard on the call. (For G3i-Global, G3V2, and later releases, the system administrator can administer system so that repeated tones are heard.)

Administration

The base tone generation set and individual tones, along with companding “Mode”, “Analog Ringing Cadence”, and “Digital Loss Plan” fields are administrable on Pages 1 through 7 of the “System-Parameters Country-Options” form by the system technician, the local installer and a remote installer.

Hardware and Software Requirements

Any DEFINITY system Tone Generation board can be used. Some boards are capable of producing all North American and international tones.

The TN768 generates tones only for North America. The TN780 can generate almost all the tones required for the world. The TN2182 can generate more tones than the TN780 and adds support for Hong Kong and France.

System Parameters

A

Overview

This appendix provides information on the overall characteristics and capacities of the system.

The items presented in this chapter are grouped here for easy reference. However, most items are discussed under each applicable feature.

Administration Not Required

Administration is not required to activate the following features.

1. Attendant Auto-Manual Splitting
2. Attendant Call Waiting
3. Attendant Recall
4. Attendant Release Loop Operation
5. Automatic Incoming Call Display
6. Conference — Attendant
7. Conference — Terminal
8. Hold
9. Line Lockout
10. Move Agents from CMS
11. Recall Signaling
12. Recent Change History
13. Senderized Operation

14. Straightforward Outward Completion
15. Temporary Bridged Appearance
16. Through Dialing
17. Touch-Tone Dialing (for Terminals)
18. Transfer

Administration Required

Administration is required to activate the following features.

1. AAR/ARS Partitioning
2. Abandoned Call Search
3. Abbreviated Dialing
4. Add/Remove Skills
5. Administered Connections
6. Administration Without Hardware
7. Advice of Charge
8. Alphanumeric Dialing
9. Alternate Facility Restriction Levels
10. Agent Call Handling
11. Attendant Control of Trunk Group Access
12. Attendant Direct Extension Selection With Busy Lamp Field
13. Attendant Direct Trunk Group Selection
14. Attendant Display (Buttons only)
15. Attendant Room Status
16. Attendant Serial Calling
17. Attendant Intrusion (Call Offer)
18. Attendant Override
19. Attendant Priority Queue
20. Audible Message Waiting
21. Audio Information Exchange (AUDIX) Interface
22. Authorization Codes
23. Automatic Alternate Routing
24. Automatic Callback
25. Automatic Call Distribution

26. ACD - Auto-Available Split
27. Automatic Circuit Assurance
28. Automatic Route Selection
29. Auto Start/Don’t Split
30. Automatic Wakeup
31. Basic Call Management System
32. Bridged Call Appearance — Multi-Appearance Voice Terminal
33. Bridged Call Appearance — Single-Line Voice Terminal
34. Busy Verification of Terminals and Trunks
35. Call-by-Call Service Selection
36. Call Coverage
37. Call Detail Recording
38. Call Forward Busy/Don’t Answer
39. Call Forwarding All Calls
40. Call Park
41. Call Pickup
42. Call Prompting
43. Call Vectoring
44. Call Waiting Termination
45. CDR Account Code Dialing
46. Centralized Attendant Service
47. Class of Restriction
48. Code Calling Access
49. Customer-Provided Equipment (CPE) Alarm
50. Data Call Setup
51. Data Hotline
52. Data-Only Off-Premises Extensions
53. Data Privacy
54. Data Restriction
55. Default Dialing
56. Dial Access to Attendant
57. Digital Multiplexed Interface
58. Direct Department Calling

59. Direct Inward Dialing
60. Direct Inward Dialing and Outward Dialing — International
61. Distinctive Ringing
62. Do Not Disturb
63. DS1 Tie Trunk Service
64. EIA Interface
65. Emergency Access to the Attendant
66. Enhanced Abbreviated Dialing
67. Expert Agent Selection
68. Extension Number Portability
69. Extended Trunk Access
70. Facility and Non-Facility Associated Signaling
71. Facility Busy Indication
72. Facility Test Calls
73. Forced Entry of Account Codes
74. Generalized Route Selection
75. Hold - Automatic
76. Hot Line Service
77. Hunting
78. Inbound Call Management
79. Individual Attendant Access
80. Information Systems Network (ISN) Interface
81. Internal Automatic Answer (IAA)
82. Integrated Directory
83. Integrated Services Digital Network — Basic Rate Interface
84. Integrated Services Digital Network — Primary Rate Interface
85. Intercept Treatment
86. Intercom — Dial
87. Inter-PBX Attendant Calls
88. Intraflow and Interflow
89. Last Number Dialed
90. Leave Word Calling
91. Look Ahead Interflow

92. Loudspeaker Paging Access
93. Loudspeaker Paging Access — Deluxe
94. Malicious Call Trace
95. Manual Message Waiting
96. Manual Originating Line Service
97. Manual Signaling
98. Merlin/System 25 Voice Terminal Support—731xH Series
99. Modem Pooling
100. Multi-Appearance Preselection and Preference
101. Multiple Call Handling
102. Multiple Listed Directory Numbers
103. Music-on-Hold Access
104. Names Registration
105. Night Service
106. Off-Premises Station
107. PC Application Software Translation Exchange (PASTE)
108. PC/PBX Connection
109. Permanent Switched Calls
110. Personal Central Office Line
111. Personalized Ringing
112. Power Failure Transfer
113. Priority Calling
114. Privacy — Attendant Lockout
115. Privacy — Manual Exclusion
116. Property Management System Interface
117. QSIG Global Networking
118. Queue Status Indications
119. Recorded Announcement
120. Recorded Telephone Dictation Access
121. Remote Access
122. Report Scheduler and System Printer
123. Restrictions
124. Restriction - Fully Restricted Service

125. Ringback Queuing
126. Ringer Cutoff
127. Ringing—Abbreviated and Delayed
128. Rotary Dialing
129. Security Violation Notification
130. Service Observing
131. Single-Digit Dialing and Mixed Station Numbering
132. Switch-Based Bulletin Board
133. Ten-Digit to Seven-Digit Conversion (G1.1)
134. Tenant Partitioning
135. Terminal Translation Initialization
136. Terminating Extension Group
137. Timed Reminder
138. Time of Day Routing
139. Touch-Tone Dialing (for Trunks)
140. Transfer -Outgoing Trunk to Outgoing Trunk
141. Trunk Flash
142. Trunk Group Busy/Warning Indicators to Attendant
143. Trunk-to-Trunk Transfer
144. Uniform Call Distribution (see Direct Department Calling)
145. Uniform Dial Plan
146. VDN of Origin Announcement (VOA)
147. Voice Message Retrieval
148. Voice Terminal Display
149. VuStats
150. World Class Core BRI

Feature Access

Dial Access Only

The following features or feature options can be activated and/or deactivated by dialing the assigned Feature Access Code or Trunk Access Code.

- Abbreviated Dialing:
  - List 1
  - List 2
  - List 3
  - Program
- Agent Call Handling
  - Agent Log-In
  - Agent Log-Out
  - Manual-In
  - Auto-In
  - After Call Work
  - Auxiliary Work
  - Assist
- AP Demand Print
- Automatic Route Selection
- Automatic Callback (activate and deactivate) (applies to single-line voice terminals only)
- Automatic Wakeup
  - Wakeup Call
  - Verify Wakeup Announcement
- Call By Call Service Selection
- CDR Account Code Dialing
- Call Forwarding All Calls (activate and deactivate)
- Call Park and Call Park Answer Back
- Call Pickup
- Code Calling Access

- Controlled Restriction:
  - Single Voice Terminal (activate and deactivate)
  - Group of Voice Terminals (activate and deactivate)
- Data Origination (associated with Data Call Setup and Pooled Modem)
- Data Privacy (associated with Data Call Setup and Pooled Modem)
- Do Not Disturb
- Emergency Access to the Attendant
- Facility Test Calls
- Generalized Route Selection
- Hunt Group Make Busy (activate and deactivate) (associated with Direct Department Calling and Uniform Call Distribution)
- Integrated Services Digital Network — Primary Rate Interface
- Last Number Dialed
- Leave Word Calling:
  - Cancel a Message
  - Display Module Lock
  - Display Module Unlock
  - Store a Message
- Loudspeaker Paging Access
- Loudspeaker Paging Access — Deluxe
- Malicious Call Trace
- PC Application Software Translation Exchange (PASTE)
- Priority Calling
- Private Network Access
- Property Management System Interface
- Public Network Access
- Recorded Telephone Dictation Access
- Send All Calls (associated with Call Coverage)
- Trunk Answer From Any Station (associated with Night Service)
- Voice Message Retrieval
  - Message Retrieval Mode
  - Coverage Message Retrieval Mode
  - Delete Message
  - Repeat Message

— Next Message

— Help

— Call

Button Access Only

The following features or feature options must be assigned to a button. Feature Access Codes cannot be provided.

- Agent Change Alerting (alrt-agchg)
- Alternate Facilities Restriction Levels (attendant)
- Attendant Incoming Serial Calling (attendant)
- Attendant Intrusion (Call Offer) (attendant)
- Attendant Override (attendant)
- Automatic Callback (applies to multi-appearance voice terminals only)
- Automatic Circuit Assurance
- Bridged Call Appearance — Multi-Appearance Voice Terminal
- Bridged Call Appearance — Single-Line Voice Terminal
- Busy Verification of Terminals and Trunks
- Call Coverage:
  - — Consult
  - — Coverage Callback
  - — Coverage Message Retrieval
  - — Go To Cover
- Data Extension (associated with Data Call Setup)
- Display — Attendant or Voice Terminal:
  - — Date and Time
  - — Timer (Elapsed Time)
  - — Inspect
  - — Integrated Directory
  - — Normal
  - — Stored Number (associated with Abbreviated Dialing)
  - — Don’t Split (attendant console only)
- Facility Busy Indication

- Intercom:
  - Automatic
  - Dial
- Leave Word Calling:
  - Delete Message
  - Message Retrieval
  - Next Message (also used with Integrated Directory)
  - Return Call (also used with Integrated Directory)
- Manual Message Waiting
- Manual Signaling
- Personal Central Office Line
- Privacy — Manual Exclusion
- Property Management System Interface
  - Message Waiting Notification (Activate)
  - Message Waiting Notification (Deactivate)
  - Checkout
- Queue Status Indications
  - NQC (number of queued calls)
  - OQT (oldest queued time)
  - AQC (attendant queued calls)
  - AQT (attendant queued time)
- Ringer Cutoff
- Special Characters (associated with Abbreviated Dialing)

  Pause, Wait, Mark, and Suppress can each be assigned to a button or a Function Entry button can be assigned. Pressing Function Entry and then dialing 1, 2, 3, or 4 depicts Pause, Wait, Mark, or Suppress, respectively.
- Terminating Extension Group
- Time of Day Routing
  - Immediate Manual Override
  - Clocked Manual Override
- Trunk Identification by Attendant
- VuStats

Dial and Button Access

The following features or feature options can be activated and/or deactivated by dialing the assigned Feature Access Code; they can also be assigned to a button for button access.

- Abbreviated Dialing:
  - List 1
  - List 2
  - List 3
  - Program
- Agent Call Handling
  - Manual-In
  - Auto-In
  - Auxiliary Work
  - After Call Work
  - Assist
  - Release
- Automatic Wakeup:
  - Auto Wakeup Entry
  - Failed Messages Wakeup
- Call Forward Busy/Don’t Answer
- Call Forwarding All Calls
- Call Park and Call Park Answer Back
- Call Pickup
- Do Not Disturb
- Emergency Access to the Attendant
- Hunt Group Make Busy (activate and deactivate) (associated with Direct Department Calling and Uniform Call Distribution)
- Last Number Dialed
- Leave Word Calling:
  - Malicious Call Trace
  - Cancel a Message
  - Display Module Lock
  - Store a Message

- Priority Calling

  The Priority Calling access code and extension number to be called, or the Priority Calling access code only, can be assigned to an Abbreviated Dialing button.

- Send All Calls (associated with Call Coverage)
- Service Observing

Feature Status Button Indicators

The following buttons are not operational, but can be assigned to indicate the status of a feature or feature option. The light associated with the button lights when the assigned feature or option is active or is in use.

- Group Call (Lights to indicate that an incoming call is associated with a Call Coverage Answer group, a Direct Department Calling group, or a Uniform Call Distribution group.)
- Lock (Associated with the Voice Terminal Display; lights when activated and means that Leave Word Calling message retrieval is denied from that terminal. Other display modes still work, including Coverage Message Retrieval.)

Overview

This appendix provides information on the overall characteristics and capacities of the system.

System Hardware and Software Capacity Limits

The maximum parameters for the DEFINITY® Communications System Generic 1 and Generic 3 hardware and software items are listed on the following pages. Unless otherwise noted, these parameters apply to both the Single-Carrier Cabinet and Multi-Carrier Cabinet systems. Also:

- For G3sV1/G3sV2 and G3vsV1/G3vsV2, when a capacity limit is the same for Advantage Business Package (ABP) and Premier Business Package (PBP), the common limit is listed. When a capacity limit for ABP and PBP differ, the ABP limit is given first followed by the PBP limit.

- Terminal and digital station capacities are reduced by such administered items as: attendant consoles, number of EAS login IDs, and number of ACD agents
- Most G3iV1 and G3i-Global capacity limits are the same. In the few cases where a G3i-Global capacity limit differs from the corresponding G3i capacity limit, the G3i limit is given followed by the G3i-Global limit in parenthesis.

NOTE:
Not all maximum capacities listed in the following tables can be reached simultaneously with all versions or all configurations of the system.

Table A-1. Maximum System Parameters for G3V1

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Abbreviated Dialing (AD)				
AD Lists per System	200	200	1,600/2,400	5,000
AD List Entry Size	24	24	24	24
AD Entries per System[1]	2,000	2,000	10,000/12,000	50,000
Enhanced List (System List)	NA/1	NA/1	1	1
Maximum Entries	NA/1,000	NA/1,000	1,000	1,000
Group Lists	100	100	100	100
Maximum Entries	90	90	90	90
Group Lists per Extension	3	3	3	3
System List	1	1	1	1
Maximum Entries	90	90	90	90
Personal Lists	200	200	1,600/2,400	5,000
Maximum Entries	10	10	10	10
Personal Lists per Extension	3	3	3	3
Applications Adjuncts				
CallVisor ASAI Adjuncts	NA/NA	NA/4	8	8
Asynchronous Links (RS-232)	5	5	5	10
CDR Output Devices	2	2	2	2
Journal:System Printer	2:1	2:1	2:1	2:1
Property Management Systems	1	1	1	1
BX.25 Physical Links[2]	4	4	8	16
Application Processors (such as 3B2-MCS)	1	1	1	7
AUDIX Adjuncts	1	1	1	8
CMS Adjuncts	1	1	1	1
ICM Adjuncts ISDN Gateway	NA/1	NA/1	1	1
BX.25 Processor Channels	64	64	64	128
Hop Channels	64	64	64	128

1. 100,000 for G3rV3
2. For Single-Carrier Cabinets (SCC), only four BX.25 physical links are supported in G3sV2 and G3iV2.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
ARS/AAR[1, 2]				
AAR/ARS Patterns (Shared)	NA/40	NA/40	254	640
ARS/AAR Table Entries (NPA, NXX, RXX, HNPA, FNPA)	2,000	2,000	2,000	2,000
Choices per RHNPA Table	12	12	12	12
Digit Conversion Entries	300	300	300	400
AAR/ARS Digit Conversion				
Digits Deleted for ARS/AAR	23	23	23	23
Digits Inserted for ARS/AAR	18	18	18	18
AAR/ARS Sub-Net Trunking				
Digits Deleted for ARS/AAR	23	23	23	23
Digits Inserted for ARS/AAR	36	36	36	36
Entries in HNPA & RHNPA Tables	1,000	1,000	1,000	1,000
FRLs	8	8	8	8
Inserted Digit Strings[3]	450	450	1,200	3,000
Patterns for Measurement				
Shared Patterns for Measurement	20	20	20	25
RHNPA Tables	32	32	32	32
Routing Plans	8	8	8	8
Toll Tables	32	32	32	32
Entries per Toll Table	800	800	800	800
Trunk Groups in an ARS/AAR Pattern	6	6	6	16
UDP (Entries)	NA/240	NA/240	240	50,000
TOD Charts	8	8	8	8

1. AAR is **not an optional feature** in G3sV2 ABP.
2. ARS is available in G3s if the Automatic Route Selection Option is selected.
3. This is the number of 12 character inserted-digit-strings available for AAR/ARS preferences.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Attendant Service				
Attendant Consoles (day:night)[1]	4:1	6:1	6:1/15:1	27:1
Attendant Console 100s Groups/Attendant	20	20	20	20
Attendant Control Restriction Groups	64	64	64	96
Centralized Attendant Service				
Release Link Trunks at Branch	NA/99	NA/99	99	99
Release Link Trunk Group at Branch	NA/1	NA/1	1	1
Release Link Trunks at Main	NA/100	NA/100	400	4,000
Release Link Trunk Group at Main[2]	NA/32	NA/32	99	666
Other Access Queues				
Maximum Number of Queues	1	1	1 (12)	12
Maximum Number of Queue Slots[3]	30	30	50/80	80
Size range of Reserved Queue	NA	NA	NA (2-75)	2-75
Reserved Queue Default Size	NA	NA	NA (5)	5
Queue Length	30	30	30/80	300
Switched Loops/Console	6	6	6	6
Authorization				
Authorization Codes	1,500	1,500	5,000	90,000
Classes of Restriction	64	64	64/96	96
Classes of Service	16	16	16	16
Length of Authorization Code	4-7	4-7	4-7	4-7
Length of Barrier Code	4-7	4-7	4-7	4-7
Length of Forced Entry Account Codes	NA/1-15	NA/1-15	1-15	1-15
Restricted Call List	1	1	1	1

1. For G3vs, there can be four day consoles if there are no night consoles. Three of the four must be powered by auxiliary power.
2. This number is the same as the number of trunk groups in the system.
3. The ‘‘Maximum number of queue slots’’ is referred to as ‘‘emergency access queue length’’ in G1.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Authorization (Continued)				
Remote Access Barrier Codes	10	10	10	10
CDR Forced Entry Account Code List	NA/1	NA/1	1	1
Toll Call List	1	1	1	1
Unrestricted/Allowed Call Lists	10	10	10	10
Total Call List Entries	1,000	1,000	1,000	1,000
Automatic Callback Calls	20	20	160/240	1,000
Automatic Wakeup				
Simultaneous Display Requests	10	10	10	10
Wakeup Requests per System	200	200	1,600/2,400	10,000
Wakeup Request per Extension	1	1	1	1
Wakeup Requests per 15-minute Interval	150	150	300	950
Basic CMS				
Daily Summary Reports	7	7	7	7
Measured Agent Logins, 75 for G3vs and G3s, 200 for G3i, and 600 for G3r	75	75	200	200
Measured Splits	12/24	12/24	99	99
Measured Trunk Groups	16/32	16/32	99	32
Measured VDNs	NA/24	NA/24	99	512
Reporting Periods (30 or 60 min)	25	25	25	25
Number of Terminal User IDs	5	5	5	8

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Cabinets				
Expansion Port Network (EPN)				
Multi-Carrier Cabinet (MCC)[1,2]	NA	NA	2	21
Single-Carrier Cabinet (SCC)[1,2]	NA	NA	8	80
Small[1] (Upgrades only)[3]	NA	NA	2	20
Inter-Port Network Connectivity				
Port Networks[1]	1	1	3	22
Maximum Number of Port Networks per Cabinet	1	1	1	2
Switch Node Carriers[1] (Simplex)	NA	NA	NA	2
Switch Node Carriers[1] (Duplex)	NA	NA	NA	4
DS1 Converter Complex[1] (Simplex)	NA	NA	NA	20
DS1 Converter Complex[1] (Duplex)	NA	NA	NA	40
Processor Port Network (PPN)				
Multi-Carrier Cabinet (MCC)[4]	NA	NA	1	1
Single-Carrier Cabinet (SCC) or Ehanced Single-Carrier Cabinet (ESCC)	NA	4	4	NA
Compact Single-Carrier Cabinet (CSCC)	1	NA	NA	NA
Remote Modules				
Remote Port Network	NA	NA	2/1	21

1. 43 for G3rV3 for MCC
   164 for G3rV3 for SCC
   41 for G3rV3 for Small
   44 for G3rV3 for Port Networks
   3 for G3rV3 for Switch Node Carriers (Simplex)
   6 for G3rV3 for Switch Node Carriers (Duplex)
   41 for G3rV3 for DS1 Converter Complex (Simplex)
2. The EPNs in G3r can be DS1-remote EPNs.
3. Small systems refer to the two-carrier cabinet systems that are no longer sold to new customers.
4. MCC includes Medium Cabinet.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Call Appearances				
Bridged Images per Appearance	7	7	7	15
Call Appearances per Station[1]	54	54	54	54
Maximum Appearances per Extension	10	10	10	10
Minimum Appearances per Extension	2	2	2	2
Total Bridged Appearances	200	200	1600/2,400	10,000
Maximum Simultaneous Off-Hook per Call[2]	5	5	5	5
Call Coverage				
Coverage Answer Groups (CAG)	30	30	200	500
Coverage Paths	150	150	600	5,000
With Hospitality Parameter Reduction	NA/5	NA/5	5	5
Coverage Paths Including in Call Coverage Report	100	100	100	100
Coverage Path per Station[3]	4	4	4	4
Coverage Points in a Path	3	3	3	3
Maximum Users/Coverage Path[4]	500	500	2900 (3500)	21,875
Members per CAG	8	8	8	8
Call Detail Recording				
CDRU Trackable Extensions	200	200	1,600/2,400	10,000
Intra-switch Call Trackable Extensions[5]	100	100	100	500
Number of CDRUs per System[6]	2	2	2	2

1. The number of call appearances is the sum of primary and bridged appearances; at most ten can be primary. A maximum of 54 administrable buttons are supported for the 7434D terminal — 34 buttons in the basic terminal and an additional 20 buttons in the coverage module.
2. Does not apply to conferencing.
3. Only available with ABP when AT&T Voice Power adjunct (AUDIX, AUDIX Voice Power, AUDIX Voice Power Lodging, and DEFINITY AUDIX) are used.
4. The maximum number of users per coverage path equals the number of dial plan extensions (including hunt groups, TEGs, etc.).
5. 1,000 for G3iV3/5,000 for G3rV3
   10,604 for G3V3
6. The CDRU adjunct capacity is 40,000 calls per hour, and exceeds the system call capacity for all systems except for G3r.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Call Forwarding (Follow-me)				
Call Forwarded Digits (off-net)	16	16	16	16
Call Forwarded Numbers	200	200	1,600/2,400	10,000
Call Park				
Attendant Group Common Shared Extension Numbers	10	10	10	40
Number of Parked Calls[5]	180	180	723	5,302
Call Pickup Groups				
Call Pickup Members per Group	50	50	50	50
Call Pickup Members per System	200	200	1600/2400	10,000
Number of Groups	100	100	800	5,000
With Hospitality Parameter Reduction	NA/5	NA/5	5	5
Call Vectoring/Call Prompting				
Expert Agent Selection				
Skill Groups	NA	NA	NA	NA
VDN Skill Preferences	NA	NA	NA	NA
Multiple Skills per Call	NA	NA	NA	NA
Multiple Skills per Agent	NA	NA	NA	NA
Agent Login IDs	NA	NA	NA	NA
Multiple Splits per Call	NA/3	NA/3	3	3
Priority Levels	NA/4	NA/4	4	4
Recorded Announcement	NA/128	NA/128	128	256
Steps per Vector	NA/15	NA/15	15	15
Vector Directory Numbers[1]	NA/100	NA/100	500	3,000
Measured VDNs	NA/100	NA/100	500	2,000
Vectors per System	NA/48	NA/48	256	512

1. The total of VDN, Station, and Login ID extensions cannot exceed 25,000.
   2,000 for G3rV3
   7,084 for G3rV3 for "Simultaneous three-way Conference Calls"
   3,520 for G3rV3 for “Simultaneous six-way Conference Calls”

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
CallVisor ASAI				
Active Station Controlling Association	NA	NA/250	2,000/NA	6,000
Call Controllers per Call	NA	NA/1	1/NA	1
Call Monitors per Call	NA	NA/14	14/NA	14
Extension Controllers per Station Domain	NA	NA/2	2/NA	2
Maximum Simultaneous Call Classifications	NA	NA/40	40/NA	100
Number of ASAI Links	NA	NA/4	8/NA	8
Notification Requests[1]	NA	NA/50	170/NA	460
Simultaneous Active Adjunct Controlled Calls	NA	NA/75	300/NA	3,000
Switch to Adjunct Associations	NA	NA/127	127/NA	127
Conference Parties	6	6	6	6
Simultaneous 3-way Conference Calls[1,2]	161	161	483	3,542
Simultaneous 6-way Conference Calls[1,3]	80	80	240	1,760
Data Parameters				
Administered Connections	NA/24	NA/24	128	128
Permanent Switched Call	NA	NA	NA	NA
Alphanumeric Dialing				
Maximum Entries	50	50	200	1,250
Characters per Entry	22	22	22	22
Digital Data Endpoints	75	75	800	5,000

1. The total of VDN, Station, and Login ID extensions cannot exceed 25,000.
   2,000 for G3rV3
   7,084 for G3rV3 for "Simultaneous three-way Conf. Calls"
   3,520 for G3rV3 for "Simultaneous six-way Conf. Calls"
2. Simultaneous 3-way Conference Call=(483 / 3)* number PNs. Simultaneous 3-way Conference Call is limited by the number of Simultaneous
3. Simultaneous 6-way Conference Call=(483 / 6)* number PNs.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Dial Plan				
DID LDNs	8	8	8	20
Extensions[1]	500	500	2900/ 3500	21,875

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Extension Number Portability[2]	NA/240	NA/240	240	50,000
Feature Dial Access Codes				
Number of Access Codes	70	70	70	70
Number of Digits	1-3	1-3	1-3	1-4
Integrated Directory Entries	204	207	1607/ 2407	10,000
Maximum Extension Size	5	5	5	5
Minimum Extension Size	1	1	1	1
Miscellaneous Extensions[3]	150	150	900	3317
Names				
Number of Names[4]	448/464	448/464	3,406/4,215	22,569
Number of Characters in a Name	15	15	15	15
Non-DID LDNs	50	50	50	666
Prefix Extensions	Yes	Yes	Yes	Yes
Trunk Dial Access Codes				
Number of Access Codes	105	105	197	884
Number of Digits	1-3	1-3	1-3/1-4	1-4
Do Not Disturb (DND)				
DND Requests per System	200	200	1600/ 2400	10,000
Simultaneous Display Requests	10	10	10	10
Facility Busy Indicators				
Buttons per Tracked Resource	100	100	100	100
Number of Indicators (Station and Trunk Groups)	450	450	2400/3600	2400

1. Extensions include stations, data endpoints, hunt groups, announcements, TEGs, VDNs, common shared extensions, and code calling IDs.
2. The numbers shown in ‘‘Extension Number Portability’’ are Uniform Dialing Plan (UDP) entries.
3. Used for PCOL groups, common shared extensions, access endpoints, administered TSCs, code calling IDs, VDNs, LDNs, hunt groups, announcements, and TEGs.
4. The Number of Names = number of stations + attendant consoles + trunk groups + digital data endpoints + miscellaneous extensions.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Hunt Groups or Splits				
Announcements per Group	2	2	2	2
Announcements per System	128	128	128	256
Groups and/or Splits	12/24	12/24	99	255
With Hospitality Parameter Reduction	NA/5	NA/5	5	5
Group Members per Group/Split	150	150	200	999
Group Members per System	150	150	500	3,000
Measured ACD Agents (Switch Limits)				
Agents Logged in per System	75	75	400	1023
Logged-In Splits per Agent	3	3	3	3
ACD Supervisor Assist Per System[1]	12/24	12/24	99	255
Queue Slots per Group	200	200	200	999
Queue Slots per System	200	200	1,000	10,000
Intercom Translation Table (ICOM)				
Automatic/Manual and Dial ICOM Groups per System	10	10	32	256
Auto/Manual	16	16	32	256
Dial	16	16	32	256
Members per ICOM group				
Auto	32	32	32	32
Dial	32	32	32	32
Members per System	320	320	1,024	8,192
Last Number Dialed				
Entries per System[2]	275	275	2,400/ 3,200	15,000
Number of Digits	16	16	16	24

1. One supervisor assist per split.
2. Last Number Dialed Entries = Stations + Digital Data Endpoints.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Leave Word Calling (Switch-Based)[1]				
Messages Stored	450	450	2,000	2,000
Messages per User	10	10	10	16

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Remote Message Waiting Indicators				
Per Extension	80	80	80	80
Per System	80	80	80	500
Simultaneous Message Retrievers	60	60	60	400
System-Wide Message Retrievers	10	10	10	10
Malicious Call Trace				
Maximum Simultaneous Traces	NA	NA	NA	16
MLDN				
Via DID	8	8	8	20
Via CO	50	50	50	50
Modem Pool Groups				
Mode 2/Analog				
Group Members per System	64	64	160	2,016
Number of Groups	2	2	5	63
Members per Group	32	32	32	32
Networking				
CAS Nodes	NA/99	NA/99	99	99
DCS Nodes				
BX.25	NA/20	NA/20	20	63
ISDN PRI	NA/63	NA/63	63	63
Hybrid	NA/63	NA/63	63	63
UDP Nodes	NA/240	NA/240	240	999
Paging				
Code Calling IDs	125	125	125	125
Loudspeaker Zones	9	9	9	9

1. Leave Word Calling is available with G3s ABP only if the Voice Mail Application Support Option is purchased.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Partitions				
Attendant Partition	1	1	1	1
Extension Partition Group	8	8	8	8
Extension Partition	8	8	8	8
Personal CO Lines (PCOL)				
PCOL Appearances	4	4	4	16
PCOL Lines (Trunk Groups)	15	15	40/200	100
PCOL Trunks Per Trunk Group	1	1	1	1
Port Circuit Pack Slots[1]				
Per Expansion Port Network (EPN)				
Multi-Carrier Cabinet (MCC) Simplex	NA	NA	99	99
Multi-Carrier Cabinet (MCC) Duplex	NA	NA	98	98
Single-Carrier Cabinet (SCC) Simplex	NA	NA	71	71
Single-Carrier Cabinet (SCC) Duplex	NA	NA	70	70
Small Cabinet Simplex (Upgrade only)	NA	NA	39	39
Small Cabinet Duplex (Upgrade only)	NA	NA	38	38
Per Processor Port Network (PPN)				
Multi-Carrier Cabinet (MCC) Simplex	NA	NA	89	80
Multi-Carrier Cabinet (MCC) Duplex	NA	NA	78	60
Single-Carrier Cabinet (SCC) Simplex	NA	NA	64	NA
Single-Carrier Cabinet (SCC) Duplex	NA	NA	56	NA
Enhanced Single-Carrier Cabinet (ESCC) Simplex	NA	70	NA	NA
Enhanced Single-Carrier Cabinet (ESCC) Duplex	NA	NA	NA	NA
Compact Single-Carrier Cabinet (CSCC) Simplex	10	NA	NA	NA

1. Only port slots are included in this count. For example, there are 99 port slots per MCC EPN cabinet. One slot in the cabinet is already dedicated for the Tone/Clock board. Other service circuits may be required which would further reduce the number of port slots available. In G3 carriers, the 21st slot of MCC port carriers may be equipped with service boards that do not require tip and ring connections.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Recorded Announcements				
Analog Queue Slots per Announcement	50	50	150	300
Analog Queue Slots per System	50	50	150	300
Calls Connected per Announcement				
Integrated Announcement or Auxiliary Trunk	5	5	5	255
Analog Ports	5	5	5	128
Channels per Integrated Announcement Circuit Pack	16	16	16	16
Integrated Announcement Circuit Pack	1	1	1	1
Integrated Announcement Recording Time (Minutes:Seconds)				
16 KB recording	8:32	8:32	8:32	8:32
32 KB recording	4:16	4:16	4:16	4:16
Integrated Queue Slots per System	50	50	50	300
Recorded Announcements	128	128	128	256
System Administration				
Administrable History File Entries	50	50	250	1,250
Simultaneous Administration Command	1	1	1	5
Simultaneous Maintenance Command	1	1	1	3
Simultaneous SM Sessions	3	3	5	8
Printer Queue Size	50	50	50	50
Speech Synthesis Circuit Packs	6	6	6	6
Channels per Speech Circuit Pack	4	4	4	4

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Terminating Extension Groups (TEG)				
TEGs	32	32	32	32
Users That May Share a TEG	4	4	4	4
Time Slots				
Simultaneous Circuit Switched Calls[1]	180	180	723	5,302
Total Slots[2]	512	512	1536	11,264
Time Slots for Voice and Data[3]	483	483	1449	10,604
Time Slots per Port Network	512	512	512	512
Tone Classifiers				
Call Classifier Boards	NA/10	NA/10	10	50
Call Progress/Touch Tone Receivers	NA/80	NA/80	80	400
Tone Detector Boards	10	20	20	50
General Purpose Tone Detectors	20	40	40	100
Touch-Tone Receivers	40	80	80	200
Prompting TTR Queue Size	NA	NA	80	80
TTR Queue Size	4	4	4	4
Trunks				
DS1 Circuit Packs	8	8	30	166
Queue Slots for Trunks	32/64	32/64	198	1,332
PRI Interfaces via PI[4]	NA/4	NA/4	8	NA
PRI Interfaces via PKTINT	NA/4	NA/4	NA/8	166

1. 241 Simultaneous Circuit-Switched Calls per port network, except for G3s, where the maximum is 180.
2. 512 time slots per port network.
3. 483 time slots for Voice and Data per port network. Even though an EPN is supported in G3sV2, giving a total of two port networks, G3sV2 is designed to support only 180 Simultaneous Circuit-Switched Calls.
4. Only one PI board is supported in G3vs/G3s (both MCC and SCC), and therefore a total of four physical links, used for BX.25 or PRI, are available.
   In G3i, two PI boards can be supported in the MCC, and therefore a total of eight physical links (used for BX25 or PRI) are available. Since the SCC can only support one PI board, a total of four physical links (used for BX25 or PRI) are available in the SCC 286 and Medium configurations.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Trunks (Continued)				
PRI Temporary Signaling Connections				
TSCs in System	NA/164	NA/164	656	4,256
Call Associated TSCs	NA/100	NA/100	400	4,000
Non Call Associated TSCs	NA/64	NA/64	256	256
Administered TSCs	NA/32	NA/32	128	128
Ringback Queue Slots	120	120	120	1,332
Total PRI Interfaces (30)	NA/4	NA/4	8	166
Trunk Groups Hourly Measurements	NA	NA	NA	75
Trunk Groups in the System	16/32	16/32	99	666
Trunk Members in a Trunk Group	50/99	50/99	99	255
Trunks in System (Including Remote Access)[1]	50/100	50/100	400	4,000
With Hospitality Parameter Reduction	NA/NA	NA/NA	50	50
Measured Trunks in System	50/100	50/100	400	2000

1. G3vs has the same software capacities for stations and trunks as G3s. However, these software capacities are limited by the cabinet hardware. A typical switch may have 20 to 50 stations with 10 to 20 trunks. Station capacities can be reached only by administration without hardware (AWOH). This includes extensions administered without hardware.

Table A-1. Maximum System Parameters for G3V1 — *continued*

ITEM	G3vsV1 ABP/PBP	G3sV1 ABP/PBP	G3iV1/ G3i-Global	G3rV1
Voice Terminals				
Associated Data Modules (such as DTDMs)	75	75	800	5,000
BRI Stations[1]	NA	50	1,000	5,000
Digital Stations	200	200	1,600	10,000
Stations	200	200	1,600	10,000
Station Button Capacity (K Units)[2]	102.6	102.6	547.2	4120

1. All BRI stations can be display stations.
2. In G3, ''Station Button Capacity (units) 'replaces' Maximum Button Modules.''

The following examples show how these units can be used. The assumption is that only three call appearances are assigned to the sets (except analog sets which have no call appearance).

— Analog sets (for example, 7104A): G3r, 76 units; all other releases, 62 units.

— Digital sets with 10 buttons (for example, 7403D): G3r, 124 units; all other releases, 102 units.

— Digital sets with 34 buttons, without display (for example, 7405D): G3r, 412 units; all other releases, 342 units.

— Digital sets with 34 buttons, with display (for example, 7405D): G3r, 568 units; all other releases, 472 units.

— 7406D Digital sets with display: G3r, 412 units; all other releases, 342 units.

BRI sets with 17 buttons, with display (for example, 7506D): G3r, 304 units; all other releases, 250 units.The station button capacity can support all stations equipped as 7406D digital sets with display. For example, a total of 342* 1200 + 410.4K units for the G3iV1.1-286.

Table A-2. Maximum System Parameters for G3V1.1 or V2

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Abbreviated Dialing (AD)					
AD Lists Per System	1,600	200	200	2,400	5,000
AD List Entry Size	24	24	24	24	24
AD Entries Per System[1]	10,000	2,000	2,000	12,000	50,000
Enhanced List (System List)	1	NA/1	NA/1	1	1
Maximum Entries	1,000	NA/1,000	NA/1,000	1,000	1,000
Group Lists	100	100	100	100	1,000
Maximum Entries	90	90	90	90	90
Group Lists per Extension	3	3	3	3	3
System List	1	1	1	1	1
Maximum Entries	90	90	90	90	90
Personal Lists	1,600	200	200	2,400	5,000
Maximum Entries	10	10	10	10	10
Personal Lists per Extension	3	3	3	3	3
Applications Adjuncts					
CallVisor ASAI Adjuncts	8	NA/NA	NA/4	8	8
Asynchronous Links (RS-232)	5	5	5	5	10
CDR Output Devices	2	2	2	2	2
Journal:System Printer	2:1	2:1	2:1	2:1	2:1
Property Management Systems	1	1	1	1	1
BX.25 Physical Links[2]	8	4	4	8	16
Application Processors (such as 3B2-MCS)	1	1	1	1	7
AUDIX Adjuncts	1	1	1	1	8
CMS Adjuncts	1	1	1	1	1
CM Adjuncts					
ISDN Gateway	1	NA/1	NA/1	1	1
BX.25 Processor Channels	64	64	64	64	128
Hop Channels	64	64	64	64	128

1. 100,000 for G3rV3
2. In the case of Single-Carrier Cabinets (SCC), only four BX.25 physical links are supported in G3sV2 and G3iV2.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
ARS/AAR[1, 2]					
AAR/ARS Patterns (Shared)	254	20/40	20/40	254	640
ARS/AAR Table Entries (NPA, NXX, RXX, HNPA, FNPA)	2,000	2,000	2,000	2,000	2,000
Choices per RHNPA Table	12	12	12	12	12
Digit Conversion Entries	400	400	400	400	400
AAR/ARS Digit Conversion					
Digits Deleted for ARS/AAR	23	23	23	23	23
Digits Inserted for ARS/AAR	18	18	18	18	18
AAR/ARS Sub-Net Trunking					
Digits Deleted for ARS/AAR	23	23	23	23	23
Digits Inserted for ARS/AAR	36	36	36	36	36
Entries in HNPA & RHNPA Tables	1,000	1,000	1,000	1,000	1,000
FRLs	8	8	8	8	8
Inserted Digit Strings[3]	1,200	450	450	1,200	3,000
Patterns for Measurement					
Shared Patterns for Measurement	20	20	20	20	25
RHNPA Tables	32	32	32	32	32
Routing Plans	8	8	8	8	8
Toll Tables	32	32	32	32	32
Entries per Toll Table	800	800	800	800	800
Trunk Groups in an ARS/AAR Pattern	6	6	6	6	16
UDP (Entries)	240	NA/240	NA/240	10,000	50,000
TOD Charts	8	8	8	8	8

1. AAR is **not an optional feature** in G3sV2 ABP.
2. ARS is available in G3s if the Automatic Route Selection Option is selected.
3. This is the number of 12 character inserted-digit-strings available for AAR/ARS preferences.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Attendant Service					
Attendant Consoles (day:night)[1]	15:1	4:1	6:1	15:1	27:1
Attendant Console 100s Groups/Attendant	20	20	20	20	20
Attendant Control Restriction Groups	96	96	96	96	96
Centralized Attendant Service					
Release Link Trunks at Branch	99	NA/99	NA/99	99	255
Release Link Trunk Group at Branch	1	NA/1	NA/1	1	1
Release Link Trunks at Main	400	NA/100	NA/100	400	4,000
Release Link Trunk Groups at Main[2]	99	NA/32	NA/32	99	666
Other Access Queues					
Maximum Number of Queues	12	12	12	12	
Maximum Queue Slots[3]	80	30	30	80	80
Size Range of Reserved Queue	2-75	2-25	2-25	2-75	2-75
Reserved Queue Default Size	5	5	5	5	5
Queue Length	80	30	30	80	300
Switched Loops/Console	6	6	6	6	6
Authorization					
Authorization Codes	5,000	1,500	1,500	5,000	90,000
Classes of Restriction	96	96	96	96	96
Classes of Service	16	16	16	16	16
Length of Authorization Code	4-7	4-7	4-7	4-7	4-7
Length of Barrier Code	4-7	4-7	4-7	4-7	4-7
Length of Forced Entry Account Codes	1-15	NA/1-15	NA/1-15	1-15	1-15
Restricted Call List	1	1	1	1	1

1. For G3vs, there can be four day consoles if there are no night consoles. Three of the four must be powered by auxiliary power.
2. This is the same as the number of trunk groups in the system.
3. Referred to as ‘‘emergency access queue length’’ in G1.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Authorization (con't)					
Remote Access Barrier Codes	10	10	10	10	10
CDR Forced Entry Account Code List	1	NA/1	NA/1	1	1
Toll Call List	1	1	1	1	1
Unrestricted/Allowed Call Lists	10	10	10	10	10
Total Call List Entries	1,000	1,000	1,000	1,000	1,000
Automatic Callback Calls	160	20	20	240	1,500
Automatic Wakeup					
Simultaneous Display Requests	10	10	10	10	30
Wakeup Requests per System	1,200	200	200	2,400	15,000
Wakeup Request per Extension	1	1	1	1	1
Wakeup Requests per 15-minute Interval	300	150	150	450	950
Basic CMS					
Daily Summary Reports	7	7	7	7	7
Measured Agent Logins, 75 for G3vs and G3s, 200 for G3i, and 600 for G3r	200	75	75	200	200
Measured Splits	99	12/24	12/24	99	99
Measured Trunk Groups	32	16/32	16/32	32	32
Measured VDNs	99	NA/24	NA/24	99	512
Reporting Periods (30 or 60 min)	25	25	25	25	25
Number of Terminal User IDs	5	5	5	5	8

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Cabinets					
Expansion Port Network (EPN)					
MCC[1,2]	2	NA	NA	2	21
SCC[1,2]	8	NA	NA	8	80
Small[1] (Upgrades only)[3]	2	NA	NA	2	20
Inter-Port Network Connectivity					
Port Networks[1]	3	1	1	3	22
Maximum Number of Port Networks/Cabinet	1	1	1	1	2
Switch Node Carriers[1] (Simplex)	NA	NA	NA	NA	2
Switch Node Carriers[1] (Duplex)	NA	NA	NA	NA	4
DS1 Converter Complex[1] (Simplex)	NA	NA	NA	NA	20
DS1 Converter Complex[1] (Duplex)	NA	NA	NA	NA	40
Processor Port Network (PPN)					
MCC[4]	1	NA	NA	1	1
SCC/ESCC	4	NA	4	4	NA
CSCC	NA	1	NA	NA	NA
Remote Modules					
Remote Port Network	2	NA	NA	2	21

1. 43 for G3rV3 for “MCC”
   164 for G3rV3 for “SCC”
   41 for G3rV3 for “Small”
2. The EPNs in G3r can be DS1-remote EPNs.
3. Small systems refer to the two-carrier cabinet systems that are no longer sold to new customers.
4. MCC includes Medium Cabinet.

Table A-2. Maximum System Parameters for Hardware and Software Items for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Call Appearances					
Bridged Images per Appearance	7	7	7	7	15
Call Appearances per Station[1]	54	54	54	54	54
Maximum Appearances per Extension	10	10	10	10	10
Minimum Appearances per Extension	2	2	2	2	2
Total Bridged Appearances	1,600	200	200	2,400	25,000
Maximum Simultaneous Off-Hook per Call[2]	5	5	5	5	5
Call Coverage					
Coverage Answer Groups (CAG)	200	30	30	200	750
Coverage Paths	600	150	150	600	7,500
With Hospitality Parameter Reduction	5	NA/5	NA/5	5	5
Coverage Paths Included in Call Coverage Report	100	100	100	100	100
Coverage Path per Station[3]	4	4	4	4	4
Coverage Points in a Path	3	3	3	3	3
Maximum Users per Coverage Path[4]	2900	500	500	3,500	36,065
Members per CAG	8	8	8	8	8
Call Detail Recording					
CDRU Trackable Extensions	1,600	200	200	2,400	25,000
Intra-switch Call Trackable Extensions[5]	100	100	100	100	500
Number of CDRUs per System[6]	1	1	1	1	1

1. The number of call appearances is the sum of primary and bridged appearances; at most ten can be primary. A maximum of 54 administrable buttons are supported for the 7434D terminal — 34 buttons in the basic terminal and an additional 20 buttons in the coverage module.
2. Does not apply to conferencing.
3. Only available with ABP when AT&T Voice Power adjunct (AUDIX, AUDIX Voice Power, AUDIX Voice Power Lodging, and DEFINITY AUDIX) are used.
4. The maximum number of users per coverage path equals the number of dial plan extensions (including hunt groups, TEGs, etc.).
5. 1,000 for G3iV3/5,000 for G3rV3
   10,604 for G3V3
6. The CDRU adjunct capacity is 40,000 calls per hour, and exceeds the system call capacity for all systems except for G3r.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Call Forwarding (Follow-me)					
Call Forwarded Digits (off-net)	16	16	16	16	16
Call Forwarded Numbers	1,600	200	200	2,400	25,000
Call Park					
Attendant Group Common Shared Extension Numbers	10	10	10	10	80
Number of Parked Calls	723	180	180	723	5,302
Call Pickup Groups					
Call Pickup Members per Group	50	50	50	50	50
Call Pickup Members per System	1,600	200	200	2,400	25,000
Number of Groups	800	100	100	800	5,000
With Hospitality Parameter Reduction	5	NA/5	NA/5	5	5
Call Vectoring/Call Prompting					
Expert Agent Selection					
Skill Groups	NA	NA/24	NA/24	99	255
VDN Skill Preferences	NA	NA/3	NA/3	3	3
Multiple Skills per Call	NA	NA/3	NA/3	3	3
Multiple Skills per Agent	4	NA/4	NA/4	4	4
Agent Login IDs	NA	NA/450	NA/450	1,500	10,000
Multiple Splits per Call	3	NA/3	NA/3	3	3
Priority Levels	4	NA/4	NA/4	4	4
Recorded Announcement	128	NA/128	NA/128	128	256
Steps per Vector	32	NA/32	NA/32	32	32
Vector Directory Numbers[1]	500	NA/100	NA/100	512	20,000
Measured VDNs	500	NA/100	NA/100	512	2,000
Vectors per System	256	NA/48	NA/48	256	512

1. The total of VDN, Station, and Login ID extensions cannot exceed 25,000.
   2,000 for G3rV3
   7,084 for G3rV3 for "Simultaneous 3-way Conference Calls"

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
CallVisor ASAI					
Active Station Controlling Association	2,000	NA	NA/250	2,000	6,000
Call Controllers per Call	1	NA	NA/1	1	1
Call Monitors per Call	1	NA	NA/1	1	1
Extension Controllers per Station Domain	2	NA	NA/2	2	2
Maximum Simultaneous Call Classifications	40	NA	NA/40	40	400
Number of ASAI Links	8	NA	NA/4	8	8
Notification Requests[1]	170	NA	NA/50	170	460
Simultaneous Active Adjunct Controlled Calls	300	NA	NA/75	300	3,000
Switch to Adjunct Associations	127	NA	NA/127	127	127
Conference Parties	6	6	6	6	6
Simultaneous 3-way Conference Calls[1,2]	483	161	161	483	3,542
Simultaneous 6-way Conference Calls[1,3]	240	80	80	240	1,760
Data Parameters					
Administered Connections	128	NA/24	NA/24	128	128
Permanent Switched Call	NA	NA	NA	NA	NA
Alphanumeric Dialing					
Maximum Entries	200	50	50	200	1,250
Characters per Entry	22	22	22	22	22
Digital Data Endpoints	800	75	75	800	7,500

1. The total of VDN, Station, and Login ID extensions cannot exceed 25,000.
   2,000 for G3rV3
   7,084 for G3rV3 for "Simultaneous 3-way Conference Calls"
   3,520 for G3rV3 for "Simultaneous 6-way Conference Calls"
2. Simultaneous 3-way Conference Call = (483 / 3)* number PNs. Simultaneous 3-way Conference Call is limited by the number of Simultaneous
3. Simultaneous 6-way Conference Call = (483 / 6)* number PNs.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Dial Plan					
DID LDNs	8	8	8	8	20
Extensions[1]	2900	500	500	3,500	36,065
Extension Number Portability[2]	240	NA/240	NA/240	10,000	50,000
Feature Dial Access Codes					
Number of Access Codes	70	70	70	70	70
Number of Digits	1-4	1-4	1-4	1-4	1-4
Integrated Directory Entries	1,616	207	207	2,416	25,028
Maximum Extension Size	5	5	5	5	5
Minimum Extension Size	1	1	1	1	1
Miscellaneous Extensions[3]	900	150	150	900	3317
Names					
Number of names[4]	3,615	448/464	448/464	4,215	36,511
Number of characters in name	15	15	15	15	15
Non-DID LDNs	50	50	50	50	666
Prefix Extensions	Yes	Yes	Yes	Yes	Yes
Trunk Dial Access Codes					
Number of Access Codes	157	49/65	49/65	317	884
Number of digits	1-4	1-4	1-4	1-4	1-4
Do Not Disturb (DND)					
DND Requests per System	1,600	200	200	2,400	25,000
Simultaneous Display Requests	30	10	10	10	30

1. Extensions include stations, data endpoints, hunt groups, announcements, TEGs, VDNs, common shared extensions, and code calling IDs.
2. The numbers shown in ‘‘Extension Number Portability’’ are Uniform Dialing Plan (UDP) entries.
3. Used for PCOL groups, common shared extensions, access endpoints, administered TSCs, code calling IDs, VDNs, LDNs, hunt groups, announcements, and TEGs.
4. The Number of Names = number of stations + attendant consoles + trunk groups + digital data endpoints + miscellaneous extensions.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Facility Busy Indicators					
Buttons per Tracked Resource	100	100	100	100	100
Number of Indicators (Station and Trunk Groups)	2400	450	450	3,600	5,000
Hunt Groups or Splits					
Announcements per Group	2	2	2	2	2
Announcements per System	128	128	128	128	256
Groups and/or Splits	99	12/24	12/24	99	255
With Hospitality Parameter Reduction	5	NA/5	NA/5	5	5
Group Members per Group/Split	200	150	150	200	999
Group Members per System	500	150	150	500	5200
Measured ACD Agents (Switch Limits)					
Agents Logged in per System	400	150	150	500	5200
Logged-In Splits per Agent	4	4	4	4	4
ACD Supervisor Assist Per System[1]	99	12/24	12/24	99	255
Queue Slots per Group	200	200	200	200	999
Queue Slots per System	1000	200	200	1,000	10,500
Intercom Translation Table (ICOM)					
Automatic/Manual and Dial ICOM Groups per System	32	10	10	32	256
Auto/Manual	32	10	10	32	256
Dial	32	10	10	32	256
Members per ICOM group					
Auto	32	32	32	32	32
Dial	32	32	32	32	32
Members per System	1,024	320	320	1,024	8,192

1. One supervisor assist per split.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Last Number Dialed					
Entries per System[1]	2,416	282	282	3,216	32,528
Number of Digits	24	24	24	24	24
Leave Word Calling (Switch-Based)[2]					
Messages Stored	2,000	450	450	2,000	2,000
Messages per User	10	10	10	10	16
Remote Message Waiting Indicators					
Per Extension	80	80	80	80	80
Per System	80	80	80	80	500
Simultaneous Message Retrievers	60	60	60	60	400
System-Wide Message Retrievers	10	10	10	10	10
Malicious Call Trace					
Maximum Simultaneous Traces	16	16	16	16	16
MLDN					
Via DID	8	8	8	8	20
Via CO	50	50	50	50	50
Modem Pool Groups					
Mode 2/Analog					
Group Members per System	160	64	64	160	2,016
Number of Groups	5	2	2	5	63
Members per Group	32	32	32	32	32
Networking					
CAS Nodes	99	NA/99	NA/99	99	99
DCS Nodes					
BX.25	20	NA/20	NA/20	20	20
ISDN PRI	63	NA/63	NA/63	63	63
Hybrid	63	NA/63	NA/63	63	63
UDP Nodes	240	NA/240	NA/240	240	999

1. Last Number Dialed Entries = Stations + Digital Data Endpoints.
2. Available with G3s ABP only if the Voice Mail Application Support Option is purchased.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Paging					
Code Calling IDs	125	125	125	125	125
Loudspeaker Zones	9	9	9	9	9
Partitions					
Attendant Partition	1	1	1	1	1
Extension Partition Group	8	8	8	8	8
Extension Partition	8	8	8	8	8
Personal CO Lines (PCOL)					
PCOL Appearances	4	4	4	4	16
PCOL Lines (Trunk Groups)	40	15	15	200	200
PCOL Trunks Per Trunk Group	1	1	1	1	1
Port Circuit Pack Slots[1]					
Per Expansion Port Network (EPN)					
MCC Simplex	99	NA	NA	99	99
MCC Duplex	98	NA	NA	98	98
SCC Simplex	71	NA	NA	71	71
SCC Duplex	70	NA	NA	70	70
Small Cabinet Simplex (Upgrade only)	39	NA	NA	39	39
Small Cabinet Duplex (Upgrade only)	38	NA	NA	38	38
Per Processor Port Network (PPN)					
MCC Simplex	89	NA	NA	89	80
MCC Duplex	78	NA	NA	78	60
SCC Simplex	64	NA	NA	64	NA
SCC Duplex	56	NA	NA	56	NA
ESCC Simplex	NA	70	70	70	NA
ESCC Duplex	NA	NA	NA	68	NA
CSCC Simplex	NA	10	NA	NA	NA

1. Only port slots are included in this count. For example, there are 99 port slots per MCC EPN cabinet. One slot in the cabinet is already dedicated for the Tone/Clock board. Other service circuits may be required which would further reduce the number of port slots available. In G3 carriers, the 21st slot of MCC port carriers may be equipped with service boards that do not require tip and ring connections.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Recorded Announcements					
Analog Queue Slots per Announcement	150	50	50	150	1,000
Analog Queue Slots per System	150	50	50	150	1,000
Calls Connected per Announcement					
Integrated Announcement or Auxiliary Trunk	5	5	5	25	255
Analog Ports	5	5	5	25	128
Channels per Integrated Announcement Circuit Pack	16	16	16	16	16
Integrated Announcement Circuit Packs	1	1	1	1	1
Integrated Announcement Recording Time (Minutes:Seconds)					
16 KB recording	8:32	8:32	8:32	8:32	8:32
32 KB recording	4:16	4:16	4:16	4:16	4:16
Integrated Queue Slots per System	50	50	50	50	1,000
Recorded Announcements	128	128	128	128	256
System Administration					
Administrable History File Entries	250	50	50	250	1,250
Simultaneous Administration Command	1	1	1	1	5
Simultaneous Maintenance Command	1	1	1	1	5
Simultaneous SM Sessions	5	3	3	5	8
Printer Queue Size	50	50	50	50	50
Speech Synthesis Circuit Packs	6	6	6	6	40
Channels per Speech Circuit Pack	4	4	4	4	4
Terminating Extension Groups (TEG)					
TEGs	32	32	32	32	32
Users That May Share a TEG	4	4	4	4	4

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Time Slots					
Simultaneous Circuit Switched Calls[1]	723	180	180	723	5,302
Total Slots[2]	1536	512	512	1,536	11,264
Time Slots for Voice & Data[3]	1449	483	483	1,449	10,604
Time Slots per Port Network	512	512	512	512	512
Tone Classifiers					
Call Classifier Boards	10	NA/10	NA/10	10	50
Call Progress/Touch Tone Receivers	80	NA/80	NA/80	80	400
Tone Detector Boards	20	10	20	20	50
General Purpose Tone Detectors	40	20	40	40	100
Touch-Tone Receivers	80	80	80	80	200
Prompting TTR Queue Size	80	NA/80	NA/80	80	80
TTR Queue Size	4	4	4	4	4

1. 241 Simultaneous Circuit-Switched Calls per port network, except for G3s, where the maximum is 180.
2. 512 time slots per port network.
3. 483 time slots for Voice and Data per port network. Even though an EPN is supported in G3sV2, giving a total of two port networks, G3sV2 is engineered to support only 180 Simultaneous Circuit-Switched Calls.

Table A-2. Maximum System Parameters for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Trunks					
DS1 Circuit Packs	30	8	8	30	166
Queue Slots for Trunks	198	32/64	32/64	198	1,332
PRI Interfaces via PI[1]	8	NA/4	NA/4	8	NA
PRI Interfaces via PKTINT	NA	NA	NA	NA	166
PRI Temporary Signaling Connections					
TSCs in System	656	NA/164	NA/164	656	4,256
Call Associated TSCs	400	NA/100	NA/100	400	4,000
Non Call Associated TSCs	256	NA/64	NA/64	256	256
Administered TSCs	128	NA/32	NA/32	128	128
Ringback Queue Slots	198	32/64	32/64	198	1,332
Total PRI Interfaces[2]	8	NA/4	NA/4	8	166
Trunk Groups Hourly Measurements	25	25	25	25	75
Trunk Groups in the System	99	16/32	16/32	99	666
Trunk Members in a Trunk Group	99	50/99	50/99	99	255
Trunks in System (Including Remote Access)[3]	400	50/100	50/100	400	4,000
With Hospitality Parameter Reduction	50	NA/50	NA/50	50	50
Measured Trunks in System	400	50/100	50/100	400	4000

1. Only one PI board is supported in G3vs/G3s (both MCC and SCC), and therefore a total of four physical links, used for BX.25 or PRI, are available.
   In G3i, two PI boards can be supported in the MCC, and therefore a total of eight physical links (used for BX25 or PRI) are available. Since the SCC can only support one PI board, a total of four physical links (used for BX25 or PRI) are available in the SCC 286 and Medium configurations.
2. All digital stations can be display stations.
3. G3vs has the same software capacities for stations and trunks as does G3s. However, these software capacities are limited by the cabinet hardware. A typical switch would probably have 20 to 50 stations with 10 to 20 trunks. Station capacities can be reached only by administration without hardware (AWCH). This includes extensions administered without hardware.

Table A-2. Maximum System Parameters for Hardware and Software Items for G3V1.1 or V2 — *continued*

ITEM	G3iV1.1-286	G3vsV2 ABP/PBP	G3sV2 ABP/PBP	G3iV2-386	G3rV2
Voice Terminals					
Associated Data Modules (such as DTDMs)	800	75	75	800	7,500
BRI Stations[1]	1,000	NA	50	1,000	7,000
Digital Stations	1,600	200	200	2,400	25,000
Stations	1,600	200	200	2,400	25,000
Station Button Capacity (K Units)[2]	410.4	68.4	68.4	547.2	5,260

1. All BRI stations can be display stations.
2. In G3, ‘‘Station Button Capacity (units) ‘replaces’ Maximum Button Modules.’’

The following examples show how these units can be used. The assumption is that only three call appearances are assigned to the sets (except analog sets which have no call appearance).

— Analog sets (for example, 7104A): G3r, 76 units; all other releases, 62 units.

— Digital sets with 10 buttons (for example, 7403D): G3r, 124 units; all other releases, 102 units.

— Digital sets with 34 buttons, without display (for example, 7405D): G3r, 412 units; all other releases, 342 units.

— Digital sets with 34 buttons, with display (for example, 7405D): G3r, 568 units; all other releases, 472 units.

— 7406D Digital sets with display: G3r, 412 units; all other releases, 342 units.

— BRI sets with 17 buttons, with display (for example, 7506D): G3r, 304 units; all other releases, 250 units.

The station button capacity can support all stations equipped as 7406D digital sets with display. For example, a total of 342* 1200 + 410.4K units for the G3iV1.1-286.

Table A-3. Maximum System Parameters for G3V3

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Abbreviated Dialing (AD)				
AD Lists Per System	200	200	2,400	5,000
AD List Entry Size	24	24	24	24
AD Entries Per System	2,000	2,000	12,000	100,000
Auto Dialing Button[1]				
Entries per System	NA	NA	NA	NA
Enhanced List (System List)	NA/1	NA/1	1	1
Maximum Entries	NA/1,000	NA/1,000	1,000	1,000
Group Lists	100	100	100	1,000
Maximum Entries	90	90	90	90
Group Lists per Extension	3	3	3	3
System List	1	1	1	1
Maximum Entries	90	90	90	90
Personal Lists	200	200	2,400	5,000
Maximum Entries	10	10	10	10
Personal Lists/Extension	3	3	3	3
Applications Adjuncts				
CallVisor ASAI Adjuncts	NA	NA/4	8	8
Asynchronous Links (RS-232)	5	5	5	10
SMDR Output Devices	2	2	2	2
Journal:System Printer	2:1	2:1	2:1	2:1
Property Management Systems	1	1	1	1
BX.25 Physical Links[2]	4	4	8	16
Application Processors (such as 3B2-MCS)	1	1	1	7
AUDIX Adjuncts	1	1	1	8
CMS Adjuncts	1	1	1	1
ICM Adjuncts (ISDN Gateway)	NA/1	NA/1	1	1
BX.25 Processor Channels	64	64	64	128
Hop Channels	64	64	64	128

1. No limit on the maximum number of auto dial buttons (other than the system limit on button capacity).
2. In the case of SCC/ESCC/CSCC, only four BX.25 physical links are supported in the configuration.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Automatic Call Distribution (ACD)				
Announcements per Split	2	2	2	2
Announcements per System	128	128	128	256
Splits[1]	12/24	12/24	99	255
ACD Members per Split	150	150	200	999
Split Members per System Measured ACD Agents (Switch Limits)				
Logged-In Splits per Agent[2]				
No CMS	4	4	4	4
R2 CMS	3	3	3	3
R3 CMS	3	3	3	3
R3V2 CMS	4	4	4	4
R3V4 CMS[3]	4	4	4	4
Queue Slots per Group	200	200	200	999
Queue Slots per System	200	200	1,000	10,500
ARS/AAR[4]				
AAR/ARS Patterns (Shared)	20/40	20/40	254	640
ARS/AAR Table Entries (NPA, NXX, RXX, HNPA, FNPA)	2,000	2,000	2,000	2,000
Choices per RHNPA Table	12	12	12	12
Digit Conversion Entries	400	400	400	400
AAR/ARS Digit Conversion				
Digits Deleted for ARS/AAR[5]	23	23	23	23
Digits Inserted for ARS/AAR	18	18	18	18
AAR/ARS Sub-Net Trunking				
Digits Deleted for ARS/AAR	23	23	23	23
Digits Inserted for ARS/AAR	36	36	36	36

1. All references to Hospitality Parameter Reduction on the Customer Option form have been removed.
2. In the case where going from 4 to 3 login maximums, a change to the hunt group form will also be required, which in turn would require all agents to be logged-out. In one extreme case, this is potentially avoided and R2 & R3 CMS will handle the fourth login as UNSTAFFED, appropriately.
3. R3V3 CMS was renamed to R3V4 CMS to match the DEFINITY switch numbering.
4. AAR is **not an optional feature** in the G3vs/G3s ABP.
5. + up to 7 inter-exchange carrier (IXC) digits.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
ARS/AAR (Continued)				
Entries in HNPA & RHNPA Tables	1,000	1,000	1,000	1,000
FRLs	8	8	8	8
Inserted Digit Strings[1]	450	450	1,200	3,000
Patterns for Measurement				
Shared Patterns for Measurement	20	20	20	25
RHNPA Tables	32	32	32	32
Routing Plans	8	8	8	8
Toll Tables	32	32	32	32
Entries per Toll Table	800	800	800	800
Trunk Groups in an ARS/AAR Pattern	6	6	6	16
UDP (Entries)	NA/240	NA/240	10,000	50,000
TOD Charts	8	8	8	8
Attendant Service				
Attendant Consoles (day:night)[2]	4:1	6:1	15:1	27:1
Attendant Console 100s Groups per Attendant	20	20	20	20
Attendant Control Restriction Groups	96	96	96	96
Centralized Attendant Service				
Release Link Trunks at Branch	NA/99	NA/99	99	255
Release Link Trunk Group at Branch	NA/1	NA/1	1	1
Release Link Trunks at Main	NA/100	NA/100	400	4,000
Release Link Trunk Groups at Main[3]	NA/32	NA/32	99	666

1. This is the number of 12 character inserted-digit-strings available for AAR/ARS preferences.
2. The number for G3vs V2/V3 (4) is the recommended number of consoles that should be supported due to power limitations. Of the four consoles, one may be used as a night console. The software actually supports 6:1 day/night attendant consoles.
3. The number of ‘‘Release Link Trunk Groups at Main’’ is the same as the number of trunk groups in the system.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Attendant Service (Continued)				
Other Access Queues				
Maximum Number of Queues	12	12	12	12
Maximum Number of Queue Slots[1]	30	30	80	80
Size Range of Reserved Queue	2-25	2-25	2-75	2-75
Reserved Queue Default Size	5	5	5	5
Queue Length	30	30	80	300
Switched Loops/Console	6	6	6	6
Authorization				
Authorization Codes	1,500	1,500	5,000	90,000
Classes of Restriction	96	96	96	96
Classes of Service	16	16	16	16
Length of Authorization Code	4-7	4-7	4-7	4-7
Length of Barrier Code	4-7	4-7	4-7	4-7
Length of Forced Entry Account Codes	NA/1-15	NA/1-15	1-15	1-15
Restricted Call List	1	1	1	1
Remote Access Barrier Codes	10	10	10	10
SMDR Forced Entry Account Code List	NA/1	NA/1	1	1
Toll Call List	1	1	1	1
Unrestricted/Allowed Call Lists	10	10	10	10
Total Call List Entries	1,000	1,000	1,000	1,000
Automatic Callback Calls	20	20	240	1,500
Automatic Wakeup				
Simultaneous Display Requests	10	10	10	30
Wakeup Requests per System	200	200	2,400	15,000
Wakeup Request per Extension	1	1	1	1
Wakeup Requests per 15-minute Interval	150	150	450	950

1. Referred to as ‘‘emergency access queue length’’ in G3i.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Basic CMS				
Measured Agents or Login IDs	75	75	200	600
Measured Splits	12/24	12/24	99	99
Measured Trunk Groups	16/32	16/32	32	32
Measured VDNs	NA/24	NA/24	99	512
Reporting Periods				
Intervals	25	25	25	25
Days	7	7	7	7
Cabinets				
Expansion Port Network (EPN)				
Multi-Carrier Cabinet (MCC)[1]	NA	NA	2	43
Single-Carrier Cabinet (SCC)[1]	NA	NA	8	164
Small (Upgrades only)[2]	NA	NA	2	41
Inter-Port Network Connectivity				
Port Networks	1	1	3	44
Maximum Number of Port Networks per Cabinet	1	1	1	2
Switch Nodes (Simplex)	NA	NA	NA	3
Switch Nodes (Duplex)	NA	NA	NA	6
DS1 Converter Complex (Simplex)	NA	NA	NA	41
DS1 Converter Complex (Duplex)	NA	NA	NA	82
Processor Port Network (PPN)				
Multi-Carrier Cabinet (MCC)[3]	NA	NA	1	1
Single-Carrier Cabinet (SCC) or Enhanced Single-Carrier Cabinet (ESCC)	NA	4	4	NA
Compact Single-Carrier Cabinet (CSCC)	1	NA	NA	NA

1. Only EPNs in G3r can be DS1-remote EPNs.
2. Small systems refer to the 2-carrier cabinet systems that are no longer sold to new customers.
3. MCC includes Medium Cabinet.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Call Appearances				
Bridged Images per Appearance	7	7	7	15
Call Appearances per Station[1]	54	54	54	54
Maximum Appearances per Extension	10	10	10	10
Minimum Appearances per Extension	2	2	2	2
Total Bridged Appearances	200	200	2,400	25,000
Maximum Simultaneous Off-Hook per Call[2]	5	5	5	5
Call Coverage				
Coverage Answer Groups (CAG)	30	30	200	750
Coverage Paths	150	150	600	7,500
Coverage Paths Included in Call Coverage Report	100	100	100	100
Coverage Path per Station	4	4	4	4
Coverage Points in a Path	3	3	3	3
Maximum Users per Coverage Path[3]	500	500	3,500	36,065
Members per CAG	8	8	8	8
Number of Coverage Paths for which Each Station Can Be a Member	NA	NA	NA	NA
Call Detail Recording				
CDRU Trackable Extensions	200	200	2,400	25,000
Intra-Switch Call Trackable Extensions	100	100	1,000	5,000
Number of CDRUs/System[4]	1	1	1	1
Maximum Number of CDR Records that Can Be Buffered in the Switch	300	300	300	1,900
No. of Records Buffered for the Primary Output Device that will Cause Secondary Device to be Busied Out for 2 Minutes	200	200	200	1,800

1. The number of call appearances is the sum of primary and bridged appearances; at most 10 can be primary. A maximum of 54 administrable buttons are supported for the 7434D terminal — 34 buttons in the basic terminal and an additional 20 buttons in the coverage module.
2. Does not apply to conferencing.
3. The maximum number of users per coverage path is equal to the number of extensions.
4. The CDRU adjunct capacity is 40,000 calls per hour, and it exceeds the system call capacity for all systems except for G3r.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Call Forwarding (Follow-me)				
Call Forwarded Digits (off-net)	16	16	16	16
Call Forwarded Numbers	200	200	2,400	25,000
Call Park				
Attendant Group Common Shared Extension Numbers	10	10	80	80
Number of Parked Calls	180	180	723	10,604
Call Pickup Groups				
Call Pickup Members per Group	50	50	50	50
Call Pickup Members per System	200	200	2,400	25,000
Number of Groups	100	100	800	5,000
Call Vectoring				
Maximum Skills a Call Can Simultaneously Queue to	NA/3	NA/3	3	3
Priority Levels	NA/4	NA/4	4	4
Recorded Announcement	NA/128	NA/128	128	256
Steps per Vector	NA/32	NA/32	32	32
Vector Directory Numbers	NA/100	NA/100	512	20,000
CMS Measured VDNs[1]	NA/100	NA/100	512	2,000
Vectors per System	NA/48	NA/48	256	512
Number of Collected Digits for Call Prompting	NA/16	NA/16	16	16
Number of Dial-Ahead Digits for Call Prompting	NA/24	NA/24	24	24
Vector Routing Tables	NA	NA	NA	NA
CallVisor ASAI				
Active Station Control Association	NA	NA/250	2,000	6,000
Call Controllers per Call	NA	NA/1	1	1
Call Monitors per Call	NA	NA/14	14	14
Station Controllers per Station	NA	NA/2	2	2
Maximum Simultaneous Call Classifications	NA	NA/40	40	400

1. Measured limits depend on the CMS release used.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
CallVisor ASAI (Continued)				
Number of CallVisor ASAI Links (Open & Proprietary)[1]	NA	NA/4	8	8
Notification Requests (Monitors)	NA	NA/50	170	2,000
Simultaneous Active Call Controlled Calls	NA	NA/75	300	3,000
Switch to Adjunct Associations (Routing)	NA	NA/127	127	127
Number of Open MultiQuest Billing Requests	NA	NA	NA	NA
Conference Parties	6	6	6	6
Simultaneous 3-way Conference Calls[2]	161	161	483	7,084
Simultaneous 6-way Conference Calls[3]	80	80	240	3,520
Data Parameters				
Administered Connections	NA/24	NA/24	128	128
Alphanumeric Dialing				
Maximum Entries	50	50	200	1,250
Characters/Entry	22	22	22	22
PRI Endpoints (PE)	NA/25	NA/25	25	50
Access Endpoints (Number of Trunks)	50/100	50/100	400	4,000
Digital Data Endpoints	75	75	800	7,500
Dial Plan				
DID LDNs	8	8	8	20
EAS Agent Login IDs[4]	NA/450	NA/450	1,500	10,000
Extensions	500	500	3,500	36,065
Extension Number Portability[5]	NA/240	NA/240	10,000	50,000
Feature Dial Access Codes				
Number of Access Codes	70	70	70	70
Number of Digits	1-4	1-4	1-4	1-4

1. Proprietary, exists in G3V3 only.
2. Simultaneous 3-way Conference Call = (483 / 3)* number PNs.
3. Simultaneous 6-way Conference Call = (483 / 6)* number PNs.
4. Login IDs count against the “Extensions” switch capacity.
5. These are Uniform Dialing Plan (UDP) entries.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Dial Plan (Continued)				
Integrated Directory Entries[1]	207	207	2,416	25,028
Maximum Extension Size	5	5	5	5
Minimum Extension Size	1	1	1	1
Miscellaneous Extensions[2]	150	150	900	3,317
Names				
Number of Names[3]	448/464	448/464	4,215	36,511
Number of Characters in a Name	15	15	15	15
Non-DID LDNs	50	50	50	666
Prefix Extensions	Yes	Yes	Yes	Yes
Trunk Dial Access Codes				
Number of Access Codes	49/65	49/65	317	884
Number of Digits	1-4	1-4	1-4	1-4
Do Not Disturb (DND)				
DND Requests per System	200	200	2,400	25,000
Simultaneous Display Requests	10	10	10	30
Expert Agent Selection (EAS)				
Skill Groups	NA/24	NA/24	99	255
VDN Skill Preferences	NA/3	NA/3	3	3
Maximum Skills a Call Can Simultaneously Queue to	NA/3	NA/3	3	3
Agent Login IDs	NA/450	NA/450	1,500	10,000
Maximum Skills per Agent	NA/4	NA/4	4	4
Maximum Agents that can be Logged-In				
When Each Has 4 Skills Assigned	NA/37	NA/37	125	1,300
When Each Has 1 Skill Assigned	NA/150	NA/150	500	5,200

1. The Integrated Directory Entries = Stations + Attendant Consoles.
2. Used for PCOL groups, common shared extensions, access endpoints, administered TSCs, code calling IDs, LDNs, hunt groups, announcements, and TEGs.
3. The Number of Names = number of stations + attendant consoles + trunk groups + digital data endpoints + miscellaneous extensions.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Facility Busy Indicators				
Buttons per Tracked Resource	100	100	100	100
Number of Indicators (Station and Trunk Groups)	450	450	3,600	5,000
Hunt Groups				
Announcements per Group	2	2	2	2
Announcements per System	128	128	128	256
Groups	12/24	12/24	99	255
Group Members per Group	150	150	200	999
Group Members per System	150	150	500	5,200
Queue Slots per Group	200	200	200	999
Queue Slots per System	200	200	1,000	10,500
Number of Queue Warning Lamps per Split	100	100	100	100
Number of Queue Warning Lamps per System	150	500	500	5,200
Intercom Translation Table (ICOM)				
Automatic/Manual and Dial				
ICOM groups per system	10	10	32	256
Auto/Manual	10	10	32	256
Dial	10	10	32	256
Members per ICOM group				
Auto	32	32	32	32
Dial	32	32	32	32
Members per System	320	320	1,024	8,192
Last Number Dialed				
Entries per System[1]	282	282	3,216	32,528
Number of Digits	24	24	24	24

1. The Last Number Dialed Entries = Stations + Digital Data Endpoints + Attendant Consoles.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Leave Word Calling (Switch-Based)[1]				
Messages Stored	450	450	2,000	6,000
Messages per User	10	10	10	16
Remote Message Waiting Indicators				
Per Extension	80	80	80	80
Per System	240	240	240	1,250
Simultaneous Message Retrievers	60	60	60	400
System-Wide Message Retrievers	10	10	10	10
Malicious Call Trace				
Maximum Simultaneous Traces	16	16	16	16
MLDN				
Via DID	8	8	8	20
Via CO	50	50	50	50
Modem Pool Groups				
Mode 2/Analog				
Group Members per System	64	64	160	2,016
Number of Groups	2	2	5	63
Members per Group	32	32	32	32
Networking				
CAS Nodes	NA/99	NA/99	99	99
DCS Nodes[2]				
BX.25	NA/20	NA/20	20	20
ISDN PRI	NA/20	NA/20	20	20
Hybrid	NA/20	NA/20	20	20
ENP Nodes[3]	NA/999	NA/999	999	999
Paging				
Code Calling IDs	125	125	125	125
Loudspeaker Zones	9	9	9	9

1. Leave Word Calling is available in the ABP only if the Voice Mail Option is purchased.
2. The actual limit in the software is 63, but due to performance considerations the recommended number of DCS Nodes is 20.
3. The numbers here are node number addresses.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Partitions[1]				
Attendant Group	1	1	1	1
Extension Partition Group	8	8	8	8
Extension Partition	8	8	8	8
Tenant Partition	NA	NA	NA	NA
Personal CO Lines (PCOL)				
PCOL Appearances	4	4	4	16
PCOL Lines (Trunk Groups)	15	15	200	200
PCOL Trunks Per Trunk Group	1	1	1	1
Port Circuit Pack Slots[2]				
Per Expansion Port Network (EPN)				
Multi-Carrier Cabinet (MCC) Standard Reliability	NA	NA	99	99
Single-Carrier Cabinet (SCC) Standard Reliability	NA	NA	71	71
Small Cabinet Standard Reliability (Upgrade only)	NA	NA	39	39
Per Processor Port Network (PPN)				
Multi-Carrier Cabinet (MCC) Standard Reliability	NA	NA	89	80
Single-Carrier Cabinet (SCC) Standard Reliability	NA	NA	64	NA
Single-Carrier Cabinet (ESCC) Standard Reliability	NA	70	70	NA
Single-Carrier Cabinet (CSCC) Standard Reliability	10	NA	NA	NA

1. G3V2 AND G3V3 do not support Tenant Partitioning.
2. Only port slots are included in this count. For example, there are 100 port slots per MCC EPN cabinet. One slot in the cabinet is already dedicated for the Tone/Clock board. Other service circuits may be required which would further reduce the number of port slots available. In G3r and G3i carriers, the service slot may be equipped with service boards that do not require tip and ring connections

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Recorded Announcements				
Analog and Auxiliary Trunk Announcements				
Analog and Auxiliary Trunk Queue Slots per Announcement	50	50	150	1,000
Analog and Auxiliary Trunk Queue Slots per System	50	50	150	1,000
Calls Connected per Announcement				
Auxiliary Trunk	5	5	25	255
Analog Port	5	5	25	128
Integrated Announcements				
Integrated Announcement Circuit Pack	1	1	1	1
Channels Connected per Integrated Announcement Circuit Pack	16	16	16	16
Calls Connected per Integrated Announcement	5	5	25	255
Integrated Announcement Recording Time (Minutes:Seconds)				
16 KB recording	8:32	8:32	8:32	8:32
32 KB recording	4:16	4:16	4:16	4:16
64 KB recording	2:8	2:8	2:8	2:8
Integrated Queue Slots per System[1]	50	50	50	1,000
Total Recorded Announcements	128	128	128	256
System Administration				
Number of Logins	15	15	15	15
Administrable History File Entries	50	50	250	1,250
Simultaneous Administration Command	1	1	1	5
Simultaneous Maintenance Command	1	1	1	5
Simultaneous SM Sessions	3	3	5	8
Printer Queue Size	50	50	50	50

1. The capacity for the G3vs and G3s was reduced to make the capacity proportional to that provided in the larger sizes (about 25% of the maximum number of system trunks for one board). G3i integrated queue slots should be increased to 100 for one board (200 for 5 boards) but can not be done in this release due to memory limitation (each queue slot requires 18 bytes). The G3r has been resized to 4000 queue slots for the 10 boards maximum (only 1,000 would have been needed for one board), since the common pool architecture requires a greater number of total queue slots.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Speech Synthesis Circuit Packs	6	6	6	40
Channels per Speech Circuit Pack	4	4	4	4
Terminating Extension Groups (TEG)				
TEGs	32	32	32	32
Users That May Share a TEG	4	4	4	4
Time Slots				
Simultaneous Circuit Switched Calls[1]	180	180	723	7,712
Total Slots	512	512	1,536	22,528
Time Slots for Voice and Data[2]	483	483	1,449	21,208
Time Slots per Port Network	512	512	512	512
Tone Classifiers				
Tone Receivers (General)[3]	80	200	200	840
Call Classifier Boards	NA	NA	NA	NA
Classifiers / Prompting TTRs	NA	NA	NA	NA
Tone Detector Boards	NA	NA	NA	NA
General Purpose Tone Detectors	NA	NA	NA	NA
Touch-Tone Receivers	NA	NA	NA	NA
TTR Queue Size	4	4	4	4
Prompting TTR Queue Size	NA/80	NA/80	80	80

1. 241 Simultaneous Circuit-Switched Calls per port network, except for G3vs and G3s which are 180 Simultaneous Circuit Switched Calls and G3r which has a total of 7,712 (limited by the number of call records supported).
2. There are 483 time slots for Voice and Data per port network.
3. G3V3 Release 3.0 or later, or G3V4, use TN744C Call Classifier-Detector for basic TTR usage as well as call prompting/call classification/MFC. In addition, the TN2182 Tone/Clock/Detector is used for multiple tone detection functions. The number of TN748, TN420, or TN744 boards is limited only by the number of available slots. There is a single limit on the total number of tone receiver (classifier) ports for the system. For G3V3 Release 3.0 or later, or G3V4: TN748/TN420 have 4 ports for TTR use, TN748/TN420 have 2 ports for GPTD use, TN744 has 8 ports for call prompting/call classification/MFC/TTR/GPTD use, and TN2182 has 8 ports for call prompting/call classification/MFC/TTR/GPTD use.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Trunks				
DS1 Circuit Packs	8	8	30	166
Queue Slots for Trunks	32/64	32/64	198	1,332
PRI Interfaces via PI[1]	NA/4	NA/4	8	NA
PRI Interfaces via PACCON[2]	NA	NA/8	30	NA
PRI Interfaces via PKTINT	NA	NA	NA	166
PRI Temporary Signaling Connections				
TSCs in System	NA/164	NA/164	656	4,256
Call Associated TSCs	NA/100	NA/100	400	4,000
Non Call Associated TSCs	NA/64	NA/64	256	256
Administered TSCs	NA/32	NA/32	128	128
Ringback Queue Slots	32/64	32/64	198	1,332
Total PRI Interfaces[3]	NA/4	NA/8	30	166
Trunk Groups Hourly Measurements	25	25	25	75
Trunk Groups in the System	16/32	16/32	99	666
Trunk Members in a Trunk Group	50	50/99	99	255
Trunks in System (Including Remote Access)[4]	50/100	50/100	400	4,000
Measured Trunks in System	50	50/100	400	4,000

1. Only one Processor Interface (PI) board is supported in G3vs (CSCC) and G3s (ESCC) configurations, and therefore a total of four physical links (used for BX.25 or PRI) are available. PRI interface via the PI is not available in Germany. PRI interface via the Packet Control must be used.
2. PRI interface via the Packet Control is not available on G3vs. Therefore, PRI is not available on G3vs in Germany. Other Countries must use the PI when they have the G3vs configuration.
3. In the 286 or the G3i configuration, 2 PI boards can be supported in the MCC, and therefore a total of 8 physical links (used for BX.25 or PRI) is available. Since the SCC/ESCC/CSCC can only support 1 PI board, a total of 4 physical links (used for BX.25 or PRI) is available in the G3vs and the G3s configurations. When using the Packet Control, the G3s and the G3i limit is bounded by the DS1 CP limit.
4. G3vs has the same software capacities for stations and trunks as does G3s. However, these software capacities are limited by the cabinet hardware. A typical switch would probably have 20 to 50 stations with 10 to 20 trunks. Station capacities can be reached only by administration without hardware (AWCH). This includes extensions administered without hardware.

Table A-3. Maximum System Parameters for G3V3 — *continued*

ITEM	G3vsV3 ABP/PBP	G3sV3 ABP/PBP	G3iV3	G3rV3
Voice Terminals[1]				
Associated Data Modules (such as DTDMs)	75	75	800	7,500
BRI Stations[2]	NA	50	1,000	7,000
Digital Stations[3]	80	200	2,400	25,000
Display Stations	200	200	2,400	10,000
Stations[4]	200	200	2,400	25,000
Station Button Capacity (K Units)	68.4	68.4	700.8	5,260
VuStats				
Measured Agents or Login IDs	75	75	400	2,000
Measured Splits	12/24	12/24	99	255
Measured Trunk Groups	16/32	16/32	32	32
Measured VDNs	12/24	12/24	99	512
Reporting Periods				
Intervals	1	1	1	1
Days	NA	NA	NA	NA
Display Formats	25	25	25	25
Simultaneous Updating Displays	100	100	100	500

1. The following items detract from the total number of available “Stations” on a given switch:
   - — Analog Music-On-Hold
   - — Attendants
   - — Modem Pool Conversion Resources
   - — TAAS Port
   - — Stations (Digital, Display, BRI, etc.)
   - — Analog Announcements
   - — Analog External Alarm Port
   - — Agent Login IDs
   - — ACD Agents
2. All BRI stations can be display stations (G3vs does not support BRI).
3. The software limit for digital stations in G3vs is 200 stations, but due to power limitations the recommended limit is 80 digital stations.
4. Including extensions administered without associated hardware (for the G3s, G3i and G3r Configurations). The Station Capacity for G3vs (200) is a software limit. The physical capacity of the CSCC (10 port slots) limits the G3vs configuration from reaching the software limit.

Table A-4. Maximum System Parameters for G3V4 Release 1.0

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Abbreviated Dialing (AD)				
AD Lists Per System	200	200	2,400	5,000
AD List Entry Size	24	24	24	24
AD Entries Per System	2,000	2,000	12,000	100,000
Auto Dialing Button[1]				
Entries per System	NA	NA	NA	NA
Enhanced List (System List)	NA/1	NA/1	1	1
Maximum Entries	NA/2,000	NA/2,000	10,000	10,000
Group Lists	100	100	100	1,000
Maximum Entries	100	100	100	100
Group Lists per Extension	3	3	3	3
System List	1	1	1	1
Maximum Entries	100	100	100	100
Personal Lists	200	200	2,400	5,000
Maximum Entries	100	100	100	100
Personal Lists per Extension	3	3	3	3
Applications Adjuncts				
CallVisor ASAI Adjuncts	NA	NA/4	8	8
Asynchronous Links (RS-232)	5	5	5	10
SMDR Output Devices	2	2	2	2
Journal:System Printer	2:1	2:1	2:1	2:1
Property Management Systems	1	1	1	1
BX.25 Physical Links[2]	4	4	8	16
Application Processors (such as 3B2-MCS)	1	1	1	7
AUDIX Adjuncts	1	1	1	8
CMS Adjuncts	1	1	1	1
ICM Adjuncts (ISDN Gateway)	NA/1	NA/1	1	1
BX.25 Processor Channels	64	64	64	128
Hop Channels	64	64	64	128

1. No limit on the maximum number of auto dial buttons (other than the system limit on button capacity).
2. In the case of SCC/ESCC/CSCC, only four BX.25 physical links are supported in the configuration.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Automatic Call Distribution (ACD)				
Announcements per Split	2	2	2	2
Announcements per System	128	128	128	256
Splits[1]	12/24	12/24	99	255
ACD Members per Split	150	150	200	999
Split Members per System Measured ACD Agents (Switch Limits)				
Logged-In Splits per Agent[2]				
No CMS	4	4	4	4
R2 CMS	3	3	3	3
R3 CMS	3	3	3	3
R3V2 CMS	4	4	4	4
R3V4 CMS[3]	4	4	4	4
Queue Slots per Group	200	200	200	999
Queue Slots per System	200	200	1,000	10,500
ARS/AAR[4]				
AAR/ARS Patterns (Shared)	20/40	20/40	254	640
ARS/AAR Table Entries (NPA, NXX, RXX, HNPA, FNPA)	2,000	2,000	2,000	2,000
Choices per RHNPA Table	12	12	12	12
Digit Conversion Entries	400	400	400	400
AAR/ARS Digit Conversion				
Digits Deleted for ARS/AAR[5]	28	28	28	28
Digits Inserted for ARS/AAR	18	18	18	18
AAR/ARS Sub-Net Trunking				
Digits Deleted for ARS/AAR	28	28	28	28
Digits Inserted for ARS/AAR	36	36	36	36
Digits Sent for ARS/AAR	40	56	31	68

1. All references to Hospitality Parameter Reduction on the Customer Option form have been removed.
2. In the case where going from 4 to 3 login maximums, a change to the hunt group form is required, forcing all agents to be logged-out. In one extreme case, this is potentially avoided and R2 & R3 CMS handles the fourth login as UNSTAFFED appropriately.
3. R3V3 CMS was renamed to R3V4 CMS to match the DEFINITY switch numbering.
4. AAR is **not an optional feature** in the G3vs/G3s ABP.
5. + up to 7 inter-exchange carrier (IXC) digits.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
ARS/AAR (continued)				
Entries in HNPA & RHNPA Tables	1,000	1,000	1,000	1,000
FRLs	8	8	8	8
Inserted Digit Strings[1]	450	450	1,200	3,000
Patterns for Measurement				
Shared Patterns for Measurement	20	20	20	25
RHNPA Tables	32	32	32	32
Routing Plans	8	8	8	8
Toll Tables	32	32	32	32
Entries per Toll Table	800	800	800	800
Trunk Groups in an ARS/AAR Pattern	6	6	6	16
UDP (Entries)	NA/240	NA/240	10,000	50,000
TOD Charts	8	8	8	8
Attendant Service				
Attendant Consoles (day:night)[2]	4:1	6:1	15:1	27:1
Attendant Console 100s Groups per Attendant	20	20	20	20
Attendant Control Restriction Groups	96	96	96	96
Centralized Attendant Service				
Release Link Trunks at Branch	NA/99	NA/99	99	255
Release Link Trunk Group at Branch	NA/1	NA/1	1	1
Release Link Trunks at Main	NA/100	NA/100	400	4,000
Release Link Trunk Groups at Main[3]	NA/32	NA/32	99	666
Other Access Queues				
Maximum Number of Queues	12	12	12	12
Maximum Number of Queue Slots[4]	30	30	80	80
Size Range of Reserved Queue	2-25	2-25	2-75	2-75
Reserved Queue Default Size	5	5	5	5

1. This is the number of 12 character inserted-digit-strings available for AAR/ARS preferences.
2. The number for G3vs V2/V3 (4) is the recommended number of consoles that should be supported due to power limitations. Of the four consoles, one may be used as a night console. The software actually supports 6:1 day/night attendant consoles.
3. This number is the same as the number of trunk groups in the system.
4. Referred to as ‘‘emergency access queue length’’ in G3i.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Attendant Service (continued)				
Queue Length	30	30	80	300
Switched Loops per Console	6	6	6	6
Authorization				
Authorization Codes	1,500	1,500	5,000	90,000
Classes of Restriction	96	96	96	96
Classes of Service	16	16	16	16
Length of Authorization Code	4-7	4-7	4-7	4-7
Length of Barrier Code	4-7	4-7	4-7	4-7
Length of Forced Entry Account Codes	NA/1-15	NA/1-15	1-15	1-15
Restricted Call List	1	1	1	1
Remote Access Barrier Codes	10	10	10	10
SMDR Forced Entry Account Code List	NA/1	NA/1	1	1
Toll Call List	1	1	1	1
Unrestricted/Allowed Call Lists	10	10	10	10
Total Call List Entries	1,000	1,000	1,000	1,000
Automatic Callback Calls	20	20	240	1,500
Automatic Wakeup				
Simultaneous Display Requests	10	10	10	30
Wakeup Requests per System	200	200	2,400	15,000
Wakeup Request per Extension	1	1	1	1
Wakeup Requests per 15-minute Interval	150	150	450	950
Basic CMS				
Measured Agents or Login IDs	75	75	400	2,000
Measured Splits	12/24	12/24	99	255
Measured Trunk Groups	16/32	16/32	32	32
Measured VDNs	NA/24	NA/24	99	512
Reporting Periods				
Intervals	25	25	25	25
Days	7	7	7	7

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Cabinets				
Expansion Port Network (EPN)				
Multi-Carrier Cabinet (MCC)[1]	NA	NA	2	43
Single-Carrier Cabinet (SCC)[1]	NA	NA	8	164
Small (Upgrades only)[2]	NA	NA	2	41
Inter-Port Network Connectivity				
Port Networks	1	1	3	44
Maximum Port Networks per Cabinet	1	1	1	2
Switch Nodes (Simplex)	NA	NA	NA	3
Switch Nodes (Duplex)	NA	NA	NA	6
DS1 Converter Complex (Simplex)	NA	NA	NA	41
DS1 Converter Complex (Duplex)	NA	NA	NA	82
Processor Port Network (PPN)				
Multi-Carrier Cabinet (MCC)[3]	NA	NA	1	1
SCC/ESCC	NA	4	4	NA
CSCC	1	NA	NA	NA
Call Appearances				
Bridged Images per Appearance	7	7	7	15
Call Appearances per Station[4]	54	54	54	54
Maximum Appearances per Extension	10	10	10	10
Minimum Appearances per Extension	0	0	0	0
Total Bridged Appearances	200	200	2,400	25,000
Maximum Simultaneous Off-Hook per Call[5]	5	5	5	5
Call Coverage				
Coverage Answer Groups (CAG)	30	30	200	750
Coverage Paths	150	150	600	7,500

1. Only EPNs in G3r can be DS1-remote EPNs.
2. Small systems refer to the 2-carrier cabinet systems that are no longer sold to new customers.
3. MCC includes Medium Cabinet.
4. The number of call appearances is the sum of primary and bridged appearances; at most 10 can be primary. A maximum of 54 administrable buttons are supported for the 7434D terminal — 34 buttons in the basic terminal and an additional 20 buttons in the coverage module.
5. Does not apply to conferencing.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Call Coverage (Continued)				
Coverage Paths Included in Call Coverage Report	100	100	100	100
Coverage Path per Station	4	4	4	4
Coverage Points in a Path	3	3	3	3
Maximum Users/Coverage Path[1]	500	500	3,500	36,065
Members per CAG	8	8	8	8
Number of Coverage Paths for which Each Station Can Be a Member	300	300	300	300
Call Detail Recording				
CDRU Trackable Extensions	200	200	2,400	25,000
Intra-Switch Call Trackable Extensions	100	100	1,000	5,000
Number of CDRUs/System[2]	1	1	1	1
Maximum Number of CDR Records Buffered in the Switch	300	300	300	1,900
No. of Records Buffered for the Primary Output Device Causing Secondary Device to be Busied Out for 2 Minutes	200	200	200	1,800
Call Forwarding (Follow-me)				
Call Forwarded Digits (off-net)	16	16	16	16
Call Forwarded Numbers	200	200	2,400	25,000
Call Park				
Attendant Group Common Shared Extension Numbers	10	10	80	80
Number of Parked Calls	180	180	723	10,604
Call Pickup Groups				
Call Pickup Members per Group	50	50	50	50
Call Pickup Members per System	200	200	2,400	25,000
Number of Groups	100	100	800	5,000

1. The maximum number of users per coverage path is equal to the number of extensions.
2. The CDRU adjunct capacity is 40,000 calls per hour, and it exceeds the system call capacity for all systems except for G3r.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Call Vectoring				
Maximum Skills a Call Can Simultaneously Queue to	NA/3	NA/3	3	3
Priority Levels	NA/4	NA/4	4	4
Recorded Announcement	NA/128	NA/128	128	256
Steps per Vector	NA/32	NA/32	32	32
Vector Directory Numbers	NA/100	NA/100	512	20,000
CMS Measured VDNs[1]	NA/100	NA/100	512	2,000
Vectors per System	NA/48	NA/48	256	512
Number of Collected Digits for Call Prompting	NA/16	NA/16	16	16
Number of Dial-Ahead Digits for Call Prompting	NA/24	NA/24	24	24
Vector Routing Tables	NA/5	NA/5	10	100
CallVisor ASAI				
Active Station Control Association	NA	NA/250	2,000	6,000
Call Controllers per Call	NA	NA/1	1	1
Call Monitors per Call	NA	NA/14	14	14
Station Controllers per Station	NA	NA/2	2	2
Maximum Simultaneous Call Classification	NA	NA/40	40	400
Number of CallVisor ASAI Links (Open and Proprietary)[2]	NA	NA/4	8	8
Notification Requests (Monitors)	NA	NA/50	170	2,000
Simultaneous Active Call Controlled Calls	NA	NA/75	300	3,000
Switch to Adjunct Associations (Routing)	NA	NA/127	127	127
Number of Open MultiQuest Billing Requests	NA	25	100	1,000
Conference Parties	6	6	6	6
Simultaneous 3-way Conference Calls[3]	161	161	483	7,084
Simultaneous 6-way Conference Calls[4]	80	80	240	3,520

1. Measured limits depend on the CMS release used.
2. Proprietary, exists in G3V3 only.
3. Simultaneous 3-way Conference Call = (483 / 3)* number PNs.
4. Simultaneous 6-way Conference Call = (483 / 6)* number PNs.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Data Parameters				
Administered Connections	NA/24	NA/24	128	128
Alphanumeric Dialing				
Maximum Entries	50	50	200	1,250
Characters/Entry	22	22	22	22
PRI Endpoints (PE)	NA/25	NA/25	25	50
Access Endpoints (Number of Trunks)	50/100	50/100	400	4,000
Digital Data Endpoints	75	75	800	7,500
Dial Plan				
DID LDNs	8	8	8	20
EAS Agent Login IDs[1]	NA/450	NA/450	1,500	10,000
Extensions	500	500	3,500	36,065
Extension Number Portability[2]	NA/240	NA/240	10,000	50,000
Feature Dial Access Codes				
Number of Access Codes	70	70	70	70
Number of Digits	1-4	1-4	1-4	1-4
Integrated Directory Entries[3]	207	207	2,416	25,028
Maximum Extension Size	5	5	5	5
Minimum Extension Size	1	1	1	1
Miscellaneous Extensions[4]	150	150	900	3,317
Names[5]	448/464	448/464	4,215	36,511
Number of Characters in a Name	15	15	15	15
Non-DID LDNs	50	50	50	666
Prefix Extensions	Yes	Yes	Yes	Yes
Trunk Dial Access Codes				
Number of Access Codes	49/65	49/65	317	884
Number of Digits	1-4	1-4	1-4	1-4

1. Login IDs count against the "Extensions" switch capacity.
2. The numbers shown in ''Extension Number Portability'' are Uniform Dialing Plan (UDP) entries.
3. The Integrated Directory Entries = Stations + Attendant Consoles.
4. Used for PCOL groups, common shared extensions, access endpoints, administered TSCs, code calling IDs, LDNs, hunt groups, announcements, and TEGs.
5. The Number of Names = number of stations + attendant consoles + trunk groups + digital data endpoints + miscellaneous extensions.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Do Not Disturb (DND)				
DND Requests per System	200	200	2,400	25,000
Simultaneous Display Requests	10	10	10	30
Expert Agent Selection (EAS)				
Skill Groups	NA/24	NA/24	99	255
VDN Skill Preferences	NA/3	NA/3	3	3
Maximum Skills a Call Can Simultaneously Queue to	NA/3	NA/3	3	3
Agent Login IDs	NA/450	NA/450	1,500	10,000
Maximum Skills per Agent	NA/4	NA/4	4	4
Maximum Agents that can be Logged-In				
When Each Has 4 Skills Assigned	NA/37	NA/37	125	1,300
When Each Has 1 Skill Assigned	NA/150	NA/150	500	5,200
Facility Busy Indicators				
Buttons per Tracked Resource[1]	100	100	100	500
Number of Indicators (Station and Trunk Groups)	450	450	3,600	10,000
Hunt Groups				
Announcements per Group	2	2	2	2
Announcements per System	128	128	128	256
Groups	12/24	12/24	99	255
Group Members per Group	150	150	200	999
Group Members per System	150	150	500	5,200
Queue Slots per Group	200	200	200	999
Queue Slots per System	200	200	1,000	10,500
Number of Queue Warning Lamps per Split	100	100	100	100
Number of Queue Warning Lamps per System	150	500	500	5,200

1. With G3V4 and later releases this limit is enforced. However, customers upgrading to V4 or a later release are not forced to decrease the number of buttons.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Intercom Translation Table (ICOM)				
Automatic/Manual and Dial				
ICOM groups per system	10	10	32	256
Auto/Manual	10	10	32	256
Dial	10	10	32	256
Members per ICOM group				
Auto	32	32	32	32
Dial	32	32	32	32
Members per System	320	320	1,024	8,192
Last Number Dialed				
Entries/System[1]	282	282	3,216	32,528
Number of Digits	24	24	24	24
Leave Word Calling (Switch-Based)[2]				
Messages Stored	450	450	2,000	6,000
Messages per User	125	125	125	125
Remote Message Waiting Indicators				
Per Extension	80	80	80	80
Per System	240	240	240	1,250
Simultaneous Message Retrievers	60	60	60	400
System-Wide Message Retrievers	10	10	10	10
Malicious Call Trace				
Maximum Simultaneous Traces	16	16	16	16
MLDN				
Via DID	8	8	8	20
Via CO	50	50	50	50
Modem Pool Groups				
Mode 2/Analog				
Group Members per System	64	64	160	2,016
Number of Groups	2	2	5	63
Members per Group	32	32	32	32

1. The Last Number Dialed Entries = Stations + Digital Data Endpoints + Attendant Consoles.
2. Leave Word Calling is available in the ABP only if the Voice Mail Option is purchased.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Networking				
CAS Nodes	NA/99	NA/99	99	99
DCS Nodes[1]				
BX.25	NA/20	NA/20	20	20
ISDN PRI	NA/20	NA/20	20	20
Hybrid	NA/20	NA/20	20	20
ENP Nodes[2]	NA/999	NA/999	999	999
Paging				
Code Calling IDs	125	125	125	125
Loudspeaker Zones	9	9	9	9
Partitions[3]				
Attendant Group	15	15	15	27
Extension Partition Group	8	8	8	8
Extension Partition	8	8	8	8
Tenant Partition	20	20	20	100
Personal CO Lines (PCOL)				
PCOL Appearances	4	4	4	16
PCOL Lines (Trunk Groups)	15	15	200	200
PCOL Trunks Per Trunk Group	1	1	1	1
Port Circuit Pack Slots[4]				
Per Expansion Port Network (EPN)				
MCC Standard Reliability	NA	NA	99	99
SCC Standard Reliability	NA	NA	71	71
Small Cabinet Standard Reliability (Upgrade only)	NA	NA	39	39

1. The actual limit in the software is 63, but due to performance considerations, the recommended number of DCS Nodes is 20.
2. The numbers here are node number addresses.
3. G3V2 AND G3V3 do not support Tenant Partitioning.
4. Only port slots are included in this count. For example, there are 100 port slots per MCC EPN cabinet. One slot in the cabinet is already dedicated for the Tone/Clock board. Other service circuits may be required which would further reduce the number of port slots available. In G3r and G3si, the service slot may be equipped with service boards that do not require tip and ring connections

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Port Circuit Pack Slots (Continued)				
Per Processor Port Network (PPN)				
MCC Standard Reliability	NA	NA	89	80
SCC Standard Reliability	NA	NA	64	NA
ESCC Standard Reliability	NA	70	70	NA
CSCC Standard Reliability	10	NA	NA	NA
Recorded Announcements				
Analog and Aux. Trunk Announcements				
Analog and Auxiliary Trunk Queue Slots per Announcement	50	50	150	1,000
Analog and Auxiliary Trunk Queue Slots per System	50	50	150	1,000
Calls Connected per Announcement				
Auxiliary Trunk	50	50	150	1,000
Analog Port	50	50	150	1,000
Integrated Announcements				
Integrated Announcement Circuit Packs	1	1	5	10
Channels Connected per Integrated Announcement Circuit Pack	16	16	16	16
Calls Connected per Integrated Announcement	25	25	50	1,000
Integrated Announcement Recording Time (Minutes:Seconds)				
16 KB recording	8:32	8:32	8:32	8:32
32 KB recording	4:16	4:16	4:16	4:16
64 KB recording	2:8	2:8	2:8	2:8
Integrated Queue Slots per System[1]	25	25	50	4,000
Total Recorded Announcements	128	128	128	256

1. This capacity for the G3vs and G3s was reduced to make the capacity proportional to that provided in the larger sizes (about 25% of the maximum number of system trunks for one board). G3i integrated queue slots should be increased to 100 for one board (200 for 5 boards) but can not be done in this release due to memory limitation (each queue slot requires 18 bytes). The G3r has been resized to 4000 queue slots for the 10 boards maximum (only 1,000 would have been needed for one board), since the common pool architecture requires a greater number of total queue slots.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
System Administration				
Number of Logins	15	15	15	15
Administrable History File Entries	50	50	500	1,250
Simultaneous Administration Command	1	1	1	5
Simultaneous Maintenance Command	1	1	1	5
Simultaneous SM Sessions	3	3	5	8
Printer Queue Size	50	50	50	50
Speech Synthesis Circuit Packs	6	6	6	40
Channels per Speech Circuit Pack	4	4	4	4
Terminating Extension Groups (TEG)				
TEGs	32	32	32	32
Users That May Share a TEG	4	4	4	4
Time Slots				
Simultaneous Circuit Switched Calls[1]	180	180	723	7,712
Total Slots	512	512	1,536	22,528
Time Slots for Voice & Data[2]	483	483	1,449	21,208
Time Slots per Port Network	512	512	512	512
Tone Classifiers				
Tone Receivers (General)[3]	80	200	200	840
Call Classifier Boards	NA	NA	NA	NA
Classifiers / Prompting TTRs	NA	NA	NA	NA
Tone Detector Boards	NA	NA	NA	NA
General Purpose Tone Detectors	NA	NA	NA	NA
Touch-Tone Receivers	NA	NA	NA	NA

1. 241 Simultaneous Circuit-Switched Calls per port network, except for G3vs and G3s with 180 and G3r with f 7,712 (limited by the number of call records supported).
2. There are 483 time slots for Voice and Data per port network.
3. G3V3 Release 3.0 or later, or G3V4, use TN744 Call Classifier for basic TTR usage as well as call prompting/call classification/MFC. Also, the TN2182 Tone/Clock/Detector is used for multiple tone detection functions. The number of TN748, TN420, or TN744 boards is limited only by the number of available slots. There is a single limit on the total number of tone receiver (classifier) ports for the system. For G3V3 Release 3.0 or later, or G3V4: TN748/TN420 have 4 ports for TTR use, TN748/TN420 have 2 ports for GPTD use, TN744 has 8 ports for call prompting/call classification/MFC/TTR/GPTD use, and TN2182 has 8 ports for call prompting/call classification/MFC/TTR/GPTD use.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Tone Classifiers (Continued)				
TTR Queue Size	4	4	4	4
Prompting TTR Queue Size	NA/80	NA/80	80	80
Trunks				
DS1 Circuit Packs	8	8	30	166
Queue Slots for Trunks	32/64	32/64	198	1,332
PRI Interfaces via PI[1]	NA/4	NA/4	8	NA
PRI Interfaces via Packet Control[2]	NA	NA/8	30	NA
PRI Interfaces via PKTINT	NA	NA	NA	166
PRI Temporary Signaling Connections				
TSCs in System	NA/164	NA/164	656	4,256
Call Associated TSCs	NA/100	NA/100	400	4,000
Non Call Associated TSCs	NA/64	NA/64	256	256
Administered TSCs	NA/64	NA/64	128	128
Ringback Queue Slots	32/64	32/64	198	1,332
Total PRI Interfaces[3]	NA/4	NA/8	30	166
Trunk Groups Hourly Measurements	25	25	25	75
Trunk Groups in the System	16/32	16/32	99	666
Trunk Members in a Trunk Group	50	50/99	99	255
Trunks in System (Including Remote Access)[4]	50/100	50/100	400	4,000
Measured Trunks in System	50	50/100	400	4,000

1. Only one Processor Interface (PI) is supported in G3vs (CSCC) and G3s (ESCC) configurations, therefore a total of 4 physical links (used for BX.25 or PRI) are available. PRI interface via the PI is not available in Germany. PRI interface via the Packet Control must be used.
2. PRI interface via the Packet Control is not available on G3vs. PRI is not available on G3vs in Germany. Other Countries must use the PI when they have the G3vs configuration.
3. In the 286 or the G3i configuration, 2 PI boards can be supported in the MCC, a total of 8 physical links (used for BX.25 or PRI) is available. Since the SCC/ESCC/CSCC can only support 1 PI board, a total of 4 physical links (used for BX.25 or PRI) is available in the G3vs and the G3s configurations. When using the Packet Control, the G3s and G3i limit is bounded by the DS1 CP limit.
4. G3vs has the same software capacities for stations and trunks as does G3s. However, these software capacities are limited by the cabinet hardware. A typical switch would have 20 to 50 stations with 10 to 20 trunks. Station capacities can be reached only by administration without hardware (AWOH). This includes extensions administered without hardware.

Table A-4. Maximum System Parameters for G3V4 Release 1.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3sV4 ABP/PBP	G3iV4	G3rV4
Voice Terminals[1]				
Associated Data Modules (DTDMs)	75	75	800	7,500
BRI Stations[2]	NA	50	1,000	7,000
Digital Stations[3]	80	200	2,400	25,000
Display Stations	200	200	2,400	10,000
Stations[4]	200	200	2,400	25,000
Station Button Capacity (K Units)	68.4	68.4	700.8	5,260
VuStats				
Measured Agents or Login IDs	75	75	400	2,000
Measured Splits	12/24	12/24	99	255
Measured Trunk Groups	16/32	16/32	32	32
Measured VDNs	12/24	12/24	99	512
Reporting Periods				
Intervals	25	25	25	25
Days	1	1	1	1
Display Formats	25	25	25	25
Simultaneous Updating Displays	100	100	100	500

1. The following items detract from the total number of available “Stations” on a given switch:
   - Analog Music-On-Hold
   - Attendants
   - Modem Pool Conversion Resources
   - TAAS Port
   - Stations (Digital, Display, BRI, etc.)
   - Analog Announcements
   - Analog External Alarm Port
   - Agent Login IDs
   - ACD Agents
2. All BRI stations can be display stations (G3vs does not support BRI).
3. The software limit for digital stations in G3vs is 200 stations, but due to power limitations the recommended limit is 80 digital stations.
4. Including extensions administered without associated hardware (for the G3s, G3i and G3r Configurations). The Station Capacity for G3vs (200) is a software limit. The physical capacity of the CSCC (10 port slots) limits the G3vs configuration from reaching the software limit.

Table A-5. Maximum System Parameters for G3V4 Release 3.0

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Abbreviated Dialing (AD)				
AD Lists Per System	400	400	2,400	5,000
AD List Entry Size	24	24	24	24
AD Entries Per System	2,000	2,000	12,000	100,000
Auto Dialing Button[1]				
Entries per System[1]	NA	N/A	N/A	N/A
Enhanced List (System List)	NA/1	1	1	1
Maximum Entries	NA/2,000	2,000	10,000	10,000
Group Lists	100	100	100	1,000
Maximum Entries	100	100	100	100
Group Lists per Extension	3	3	3	3
System List	1	1	1	1
Maximum Entries	100	100	100	100
Personal Lists	400	400	2,400	5,000
Maximum Entries	100	100	100	100
Personal Lists per Extension	3	3	3	3
Applications Adjuncts				
CallVisor ASAI Adjuncts	N/A	4	8	8
Asynchronous Links (RS-232)	5	5	5	10
CDR Output Devices	2	2	2	2
Journal: System Printer	2:1	2:1	2:1	2:1
Property Management Systems	1	1	1	1
BX.25 Physical Links[2]	4	4	8	16
Application Processors (such as 3B2-MCS)	1	1	1	7
AUDIX Adjuncts	1	1	1	8
CMS Adjuncts	1	1	1	1
ICM Adjuncts (ISDN Gateway)	NA/1	1	1	1
BX.25 Processor Channels	64	64	64	128
Hop Channels	64	64	64	128

1. No limit on maximum number of auto dial buttons (other than system limit on button capacity).
2. In the case of SCC/ESCC/CSCC, only four BX.25 physical links are supported in the configuration.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Automatic Call Distribution (ACD)				
Announcements per Split	2	2	2	2
Announcements per System	128	128	128	256
Splits[1]	12/24	24	99	255
ACD Members per Split	150	150	200	999
Split Members per System Measured ACD Agents (Switch Limits)				
Logged-In Splits per Agent[2]				
No CMS	4	4	4	4
R2 CMS	3	3	3	3
R3 CMS	3	3	3	3
R3V2 CMS	4	4	4	4
R3V4 CMS[3]	4	4	4	4
Queue Slots per Group	200	200	200	999
Queue Slots per System	200	200	1,000	10,500
ARS/AAR[4]				
AAR/ARS Patterns (Shared)	20/40	40	254	640
ARS/AAR Analysis Tables	2,000	2,000	2,000	2,000
Choices per RHNPA Table	12	12	12	12
Digit Conversion Entries	400	400	400	400
AAR/ARS Digit Conversion				
Digits Deleted for ARS/AAR[5]	28	28	28	28
Digits Inserted for ARS/AAR	18	18	18	18
AAR/ARS Sub-Net Trunking				
Digits Deleted for ARS/AAR	28	28	28	28
Digits Inserted for ARS/AAR	36	36	36	36
Digits Sent for ARS/AAR	40	56	31	68

1. All references to Hospitality Parameter Reduction on the Customer Option form have been removed from the Capacities Tables.
2. When going from 4 to 3 login maximums, a change to the hunt group form is required. This mean all agents must be logged-out. In one extreme case, this is potentially avoided and R2 & R3 CMS handles the fourth login as UNSTAFFED, appropriately.
3. R3V3 CMS was renamed to R3V4 CMS to match the DEFINITY switch numbering.
4. AAR is **not an optional feature** in the G3vs ABP.
5. Plus up to seven inter-exchange carrier (IXC) digits.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
ARS/AAR (Continued)				
Entries in each RHNPA Table	1,000	1,000	1,000	1,000
FRLs	8	8	8	8
Inserted Digit Strings[1]	450	450	1,200	3,000
Patterns for Measurement				
Shared Patterns for Measurement	20	20	20	25
RHNPA Tables	32	32	32	32
Routing Plans	8	8	8	8
ARS Toll Tables	32	32	32	32
Entries per Toll Table	800	800	800	800
Trunk Groups in an ARS/AAR Pattern	6	6	6	16
UDP (Entries)	NA/240	240	10,000	50,000
TOD Charts	8	8	8	8
Toll Analysis Table Entries	1,000	1,000	1,000	1,000
Attendant Service				
Attendant Consoles (day:night)[2]	4	6:1	15:1	27:1
Attendant Console 100s Groups per Attendant	20	20	20	20
Attendant Control Restriction Groups	96	96	96	96
Centralized Attendant Service				
Release Link Trunks at Branch	NA/99	99	99	255
Release Link Trunk Group at Branch	NA/1	1	1	1
Release Link Trunks at Main	NA/100	100	400	4,000
Release Link Trunk Groups at Main[3]	NA/32	32	99	666
Other Access Queues				
Maximum Number of Queues	12	12	12	12
Maximum Number of Queue Slots[4]	30	30	80	80
Size Range of Reserved Queue	2-25	2-25	2-75	2-75
Reserved Queue Default Size	5	5	5	5
Queue Length	30	30	80	300
Switched Loops per Console	6	6	6	6

1. Number of available 12 character inserted-digit-strings available for AAR/ARS preferences.
2. Recommended number of consoles supported due to power limitations. Of the four consoles, one may be used as a night console. The software actually supports 6:1 day/night attendant consoles.
3. This is the same as the number of trunk groups in the system.
4. Referred to as ‘‘emergency access queue length’’ in G3s.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Authorization				
Authorization Codes	1,500	1,500	5,000	90,000
Station Security Code Length	4	4	4	4
Classes of Restriction	96	96	96	96
Classes of Service	16	16	16	16
Length of Authorization Code	4-7	4-7	4-7	4-7
Length of Barrier Code	4-7	4-7	4-7	4-7
Length of Account Codes	NA/1-15	1-15	1-15	1-15
Restricted Call List	1	1	1	1
Remote Access Barrier Codes	10	10	10	10
CDR Account Code List	NA/1	1	1	1
Toll Call List	1	1	1	1
Unrestricted/Allowed Call Lists	10	10	10	10
Total Call List Entries	1,000	1,000	1,000	1,000
Automatic Callback Calls	20	20	240	1,500
Automatic Wakeup				
Simultaneous Display Requests	10	10	10	30
Wakeup Requests per System	400	400	2,400	15,000
Wakeup Request per Extension	1	1	1	1
Wakeup Requests per 15-minute Interval	150	150	450	950
Basic CMS				
Measured Agents or Login IDs	75	75	400	2,000
Measured Splits	12/24	24	99	255
Measured Trunk Groups	16/32	32	32	32
Measured VDNs	NA/24	24	99	512
Reporting Periods				
Intervals	25	25	25	25
Days	7	7	7	7

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Cabinets				
Expansion Port Network (EPN)				
Multi-Carrier Cabinet (MCC[1])	N/A	N/A	2	43
Single-Carrier Cabinet (SCC[1])	N/A	N/A	8	164
Small (Upgrades only)[2]	N/A	N/A	2	41
Inter-Port Network Connectivity				
Port Networks	1	1	3	44
Maximum Number of Port Networks per Cabinet	1	1	1	2
Switch Nodes (Simplex)	N/A	N/A	NA	3
Switch Nodes (Duplex)	N/A	N/A	NA	6
DS1 Converter Complex (Simplex)	N/A	N/A	NA	41
DS1 Converter Complex (Duplex)	N/A	N/A	NA	82
Processor Port Network (PPN)				
Multi-Carrier Cabinet (MCC)[3]	N/A	N/A	1	1
Single-Carrier Cabinet (SCC) or Enhanced Single-Carrier Cabinet (ESCC)	N/A	4	4	NA
CSCC	1	N/A	NA	NA
Call Appearances				
Bridged Images per Appearance	7	7	7	15
Call Appearances per Station[4]	54	54	54	54
Maximum Appearances per Extension	10	10	10	10
Minimum Appearances per Extension	0	0	0	0
Total Bridged Appearances	400	400	2,400	25,000
Maximum Simultaneous Off-Hook per Call[5]	5	5	5	5

1. Only EPNs in G3r can be DS1-remote EPNs.
2. Small systems refer to the 2-carrier cabinet systems no longer sold to new customers.
3. MCC includes the Medium Cabinet.
4. The number of appearances is the sum of primary and bridged appearances; at most 10 can be primary. A maximum of 54 administrable buttons are supported for the 7434D terminal — 34 buttons in the basic terminal and 20 additional buttons in the coverage module.
5. Does not apply to conferencing.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Call Coverage				
Coverage Answer Groups (CAG)	30	30	200	750
Coverage Paths[1]	150	150	600	7,500
Coverage Paths Including in Call Coverage Report	100	100	100	100
Coverage Paths per Station	4	4	4	4
Coverage Points in a Path	3	3	3	3
Remote Coverage Points	225	225	225	225
Maximum Users per Coverage Path[2]	700	700	3,500	36,065
Members per Call Answer Group	8	8	8	8
Call Detail Recording				
Intra-Switch Call Trackable Extensions	100	100	1,000	5,000
Maximum Number of CDR Records Buffered in Switch	300	300	300	1,900
Number of Records Buffered for the Primary Output Device to Cause Secondary Device to be Busied Out for 2 Minutes	200	200	200	1,800
Call Forwarding				
Call Forwarded Digits (off-net)	16	16	16	16
Call Forwarded Numbers	400	400	2,400	25,000
Call Park				
Attendant Group Common Shared Extension Numbers	10	10	80	80
Number of Parked Calls	180	180	723	10,604
Call Pickup Groups				
Call Pickup Members per Group	50	50	50	50
Call Pickup Members per System	400	400	2,400	25,000
Number of Groups	100	100	800	5,000

1. All references to Hospitality Parameter Reduction on the Customer Option form have been removed from the Capacities Tables.
2. The maximum number of users per coverage path is equal to the number of extensions.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Call Vectoring				
Maximum Skills a Call Can Simultaneously Queue to	NA/3	3	3	3
Priority Levels	NA/4	4	4	4
Recorded Announcement/Analog	NA/128	128	128	256
Steps per Vector	NA/32	32	32	32
Vector Directory Numbers	NA/100	100	512	20,000
CMS Measured VDNs[1]	NA/100	100	512	2,000
Vectors per System	NA/48	48	256	512
Number of Collected Digits for Call Prompting	NA/16	16	16	16
Number of Dial-Ahead Digits for Call Prompting	NA/24	24	24	24
Vector Routing Tables	NA/5	5	10	100
CallVisor ASAI				
Active Station Control Association	N/A	250	2,000	6,000
Call Controllers per Call	N/A	1	1	1
Call Monitors per Call	N/A	14	14	14
Station Controllers per Station	N/A	2	2	2
Maximum Simultaneous Call Classifications	N/A	40	40	400
Number of CallVisor ASAI Links (Open and Proprietary)[2]	N/A	4	8	8
Notification Requests (Monitors)	N/A	50	170	2,000
Simultaneous Active Call Controlled Calls	N/A	75	300	3,000
Switch to Adjunct Associations (Routing)	N/A	127	127	127
Number of Open MultiQuest Billing Requests	N/A	25	100	1,000
Maximum Calls With Send DTMF Active	N/A	16	16	32
Selected Listen - Disconnect Paths	N/A	N/A	N/A	N/A
LAN Gateway Circuit Pack Maximum Links	N/A	4	4	4
Conference Parties	6	6	6	6
Simultaneous 3-way Conference Calls[3]	161	161	483	7,084
Simultaneous 6-way Conference Calls[4]	80	80	240	3,520

1. Measured limits depend on the CMS release used.
2. Proprietary, exists in G3V3 only.
3. Simultaneous 3-way Conference Call = (483 / 3)* number PNs.
4. Simultaneous 6-way Conference Call = (483 / 6)* number PNs.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Data Parameters				
Administered Connections	NA/24	24	128	128
Alphanumeric Dialing				
Maximum Entries	50	50	200	1,250
Characters per Entry	22	22	22	22
PRI Endpoints (PE)	NA/25	25	25	50
Access Endpoints (Number of Trunks)	50/100	100	400	4,000
Digital Data Endpoints	75	75	800	7,500
Dial Plan				
DID LDNs	8	8	8	20
EAS Agent Login IDs[1]	NA/450	450	1,500	10,000
Extensions	700	700	3,500	36,065
Extension Number Portability[2]	NA/240	240	10,000	50,000
Feature Dial Access Codes	77	77	77	77
Number of Access Codes	70	70	70	70
Number of Digits	1-4	1-4	1-4	1-4
Integrated Directory Entries[3]	407	407	2,416	25,028
Maximum Extension Size	5	5	5	5
Minimum Extension Size	1	1	1	1
Miscellaneous Extensions[4]	150	150	900	3,317
Names				
Number of Names[5]	648/664	664	4,215	36,511
Number of Characters in a Name	15	15	15	15
Non-DID LDNs	50	50	50	666
Prefix Extensions	Yes	Yes	Yes	Yes

1. Login IDs count against the “Extensions” switch capacity.
2. These numbers are Uniform Dialing Plan (UDP) entries.
3. The Integrated Directory Entries = Stations + Attendant Consoles.
4. Used for PCOL groups, common shared extensions, access endpoints, administered TSCs, code calling IDs, LDNs, hunt groups, announcements, and TEGs.
5. The Number of Names = number of stations + attendant consoles + trunk groups + digital data endpoints + miscellaneous extensions.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Dial Plan (Continued)				
Trunk Dial Access Codes				
Number of Access Codes	49/65	65	317	884
Number of Digits	1-4	1-4	1-4	1-4
Do Not Disturb (DND)				
DND Requests per System	400	400	2,400	25,000
Simultaneous Display Requests	10	10	10	30
Expert Agent Selection (EAS)				
Skill Groups	NA/24	24	99	255
VDN Skill Preferences	NA/3	3	3	3
Maximum Skills a Call Can Simultaneously Queue to	NA/3	3	3	3
Agent Login IDs	NA/450	450	1,500	10,000
Maximum Skills per Agent	NA/4	4	4	4
Maximum Agents that can be Logged-In				
When Each Has 4 Skills Assigned	NA/37	37	125	1,300
When Each Has 1 Skill Assigned	NA/150	150	500	5,200
Facility Busy Indicators				
Buttons per Tracked Resource	100	100	100	500
Number of Indicators (Station and Trunk Groups)	450	450	3,600	10,000
Hunt Groups				
Announcements per Group	1	1	1	1
Announcements per System	128	128	128	256
Groups[1]	12/24	24	99	255
Group Members per Group	150	150	200	999
Group Members per System	150	150	500	5,200
Queue Slots per Group	200	200	200	999
Queue Slots per System	200	200	1,000	10,500

1. All references to Hospitality Parameter Reduction on the Customer Option form have been removed from the Capacities Tables.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Intercom Translation Table (ICOM)				
Automatic/Manual and Dial				
ICOM groups per system	10	10	32	256
Auto/Manual	10	10	32	256
Dial	10	10	32	256
Members per ICOM group				
Auto	32	32	32	32
Dial	32	32	32	32
Members per System	320	320	1,024	8,192
Last Number Dialed				
Entries per System[1]	482	482	3,216	32,518
Number of Digits	24	24	24	24
Leave Word Calling (Switch-Based)[2]				
Messages Stored	650	650	2,000	6,000
Messages per User	125	125	125	125
Remote Message Waiting Indicators				
Per Extension	80	80	80	80
Per System	240	240	240	1,250
Simultaneous Message Retrievers	60	60	60	400
System-Wide Message Retrievers	10	10	10	10
Malicious Call Trace				
Maximum Simultaneous Traces	16	16	16	16
MLDN				
Via Direct Inward Dialing	8	8	8	20
Via Direct Inward Dialing with Tenant Partition	20	20	20	100
Via CO	99	99	99	99
Modem Pool Groups				
Mode 2/Analog				
Group Members per System	64	64	160	2,016
Number of Groups	2	2	5	63
Members per Group	32	32	32	32

1. The Last Number Dialed Entries = Stations + Digital Data Endpoints + Attendant Consoles.
2. Leave Word Calling is available in the ABP only if the Voice Mail Option is purchased.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Networking				
CAS Nodes	NA/99	99	99	99
DCS Nodes[1]				
BX.25	NA/20	20	20	20
ISDN PRI	NA/20	20	20	20
Hybrid	NA/20	20	20	20
ENP Nodes[2]	NA/999	999	999	999
Paging				
Code Calling IDs	125	125	125	125
Loudspeaker Zones	9	9	9	9
Partitions				
Attendant Group	15	15	15	27
Ext. Partition Group	8	8	8	8
Extension Partition	8	8	8	8
Tenant Partition	20	20	20	100
Personal CO Lines (PCOL)				
PCOL Appearances	4	4	4	16
PCOL Lines (Trunk Groups)	15	15	200	200
PCOL Trunks Per Trunk Group	1	1	1	1
Port Circuit Pack Slots[3]				
Per Expansion Port Network (EPN)				
MCC Standard Reliability	N/A	N/A	99	99
SCC Standard Reliability	N/A	N/A	71	71
Small Cabinet Standard Reliability (Upgrade only)	N/A	N/A	39	39

1. The actual limit in the software is 63, but due to performance considerations the recommended number of DCS Nodes is 20.
2. These are node number addresses.
3. Only port slots are included in this count. For example, there are 100 port slots per MCC EPN cabinet. One slot in the cabinet is already dedicated for the Tone/Clock board. Other service circuits may be required that would further reduce the number of port slots available. In G3r and G3si carriers, the service slot may be equipped with service boards that do not require tip and ring connections

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Port Circuit Pack Slots (Continued)				
Per Processor Port Network (PPN)				
MCC Standard Reliability	N//A	N/A	89	80
SCC Standard Reliability	NA	N/A	64	NA
ESCC Standard Reliability	N/A	70	70	NA
CSCC Standard Reliability	10	N/A	NA	NA
Recorded Announcements				
Analog and Auxiliary Trunk Announcements				
Analog and Auxiliary Trunk Queue Slots per Announcement	50	50	150	1,000
Analog & Auxiliary Trunk Queue Slots per System	50	50	150	1,000
Calls Connected per Announcement				
Auxiliary Trunk	50	50	150	1,000
Analog Port	50	50	150	1,000
Integrated Announcements				
Integrated Announcement Circuit Packs	1	1	5	10
Channels Connected per Integrated Announcement Circuit Pack	16	16	16	16
Calls Connected per Integrated Announcement	25	25	50	1,000
Integrated Announcement Recording Time (Minutes:Seconds)				
16 KB recording	8:32	8:32	8:32	8:32
32 KB recording	4:16	4:16	4:16	4:16
64 KB recording	2:8	2:8	2:8	2:8
Integrated Queue Slots per System[1]	25	25	50	4,000
Total Recorded Announcements	128	128	128	256

1. The G3r has been resized to 4000 queue slots for the 10 boards maximum (only 1,000 is needed for one board), since common pool architecture requires a greater number of total queue slots.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
System Administration				
Number of Logins	15	15	15	15
Administrable History File Entries	50	50	500	1,250
Simultaneous Administration Command	1	1	1	5
Simultaneous Maintenance Command	1	1	1	5
Simultaneous SM Sessions	3	3	5	8
Printer Queue Size	50	50	50	50
Speech Synthesis Circuit Packs	6	6	6	40
Channels per Speech Circuit Pack	4	4	4	4
Terminating Extension Groups (TEG)				
TEGs	32	32	32	32
Users That May Share a TEG	4	4	4	4
Time Slots				
Simultaneous Circuit-Switched Calls[1]	180	180	723	7,712
Total Slots	512	512	1,536	22,528
Time Slots for Voice & Data	483	483	1,449	21,208
Time Slots per Port Network	512	512	512	512
Tone Classifiers				
Tone Receivers (General)[2]	80	200	200	840
Call Classifier Boards	N/A	N/A	NA	NA
Classifiers / Prompting TTRs	N/A	N/A	NA	NA
Tone Detector Boards	N/A	N/A	NA	NA
General Purpose Tone Detectors	N/A	N/A	NA	NA
Touch-Tone Receivers	N/A	N/A	NA	NA
TTR Queue Size	4	4	4	4
Prompting TTR Queue Size	NA/80	80	80	80

1. 241 simultaneous circuit-switched calls per port network, except for G3vs and G3si with 180 and G3r with 7,712 (limited by number of call records supported).
2. G3V4 uses TN744 Call Classifier for basic TTR usage as well as call prompting/call classification/MFC. Also, the TN2182 Tone/Clock/Detector is used for multiple tone detection functions. The number of TN748, TN420, or TN744 boards is limited only by the number of available slots. The number of TN2182 boards is limited. There is a single limit on the total number of tone receiver (classifier) ports for the system: TN748/TN420 have 4 ports for TTR use, TN748/TN420 have 2 ports for GPTD use, TN744 has 8 ports for call prompting/call classification/MFC/TTR/GPTD use, and TN2182 has 8 ports for call prompting/call classification/MFC/TTR/GPTD use.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Trunks				
DS1 Circuit Packs	8	8	30	166
Queue Slots for Trunks	32/64	64	198	1,332
PRI Interfaces via PI[1]	NA/4	4	8	NA
PRI Interfaces via PACCON[2]	N/A	8	30	NA
PRI Interfaces via PKTINT	N/A	N/A	NA	166
PRI Temporary Signaling Connections				
TSCs in System	NA/164	164	656	4,256
Call Associated TSCs	NA/100	100	400	4,000
Non-Call Associated TSCs	NA/64	64	256	256
Administered TSCs	NA/64	64	128	128
Ringback Queue Slots	32/64	64	198	1,332
Total PRI Interfaces	NA/4	8	30	166
Trunk Groups Hourly Measurements	25	25	25	75
Trunk Groups in the System	16/32	32	99	666
Trunk Members in a Trunk Group	50	99	99	255
Trunks in System (Including Remote Access)[3]	50/100	100	400	4,000
Measured Trunks in System	50	100	400	4,000

1. Only one Processor Interface (PI) circuit pack is supported in G3vs (CSCC) and G3s (ESCC) configurations, therefore a total of 4 physical links (used for BX.25 or PRI) are available. PRI interface via the PI is not available in Germany. PRI interface via the Packet Controller is used.
2. PRI interface via the Packet Controller is not available on G3vs. Therefore, PRI is not available on G3vs in Germany. Other Countries must use the PI when they have the G3vs configuration.
3. All references to Hospitality Parameter Reduction on the Customer Option form have been removed from the Capacities Tables.

Table A-5. Maximum System Parameters for G3V4 Release 3.0 — *continued*

ITEM	G3vsV4 ABP/PBP	G3siV4	G3siV4 +m	G3rV4
Voice Terminals[1]				
Associated Data Modules (DTDMs)	75	75	800	7,500
BRI Stations[2]	NA	50	1,000	7,000
Digital Stations	400	400	2,400	25,000
Display Stations	400	400	2,400	25,000
Stations	400	400	2,400	25,000
Station Button Capacity (K Units)	102.8	102.8	700.8	5,260
VuStats				
Measured Agents or Login IDs	75	75	400	2,000
Measured Splits	12/24	12/24	99	255
Measured Trunk Groups	16/32	16/32	32	32
Measured VDNs	12/24	12/24	99	512
Reporting Periods				
Intervals	25	25	25	25
Days	1	1	1	1
Display Formats	25	25	25	25
Simultaneous Updating Displays	100	100	100	500

1. The following items detract from the total number of available “Stations” on a given switch:
   - — Analog Music-On-Hold
   - — Attendants
   - — Modem Pool Conversion Resources
   - — TAAS Port
   - — Stations (Digital, Display, BRI, etc.)
   - — Analog Announcements
   - — Analog External Alarm Port
   - — Agent Login IDs
   - — ACD Agents
2. All BRI stations can be display stations (G3vs does not support BRI).

References

B

The following is a list of DEFINITY Communications System Generic 3 documents, including a brief description of each document.

To order copies, refer to the address and phone number on the back of this document's title page. For addition DEFINITY Communications System documents, refer to the *GBCS Publications Catalog,* 555-000-010, available from the AT&T Customer Information Center.

Basic

The following are basic documents for anyone using the DEFINITY Communications System.

DEFINITY Communications System Generic 3 Feature Description, Issue 3 **555-230-204**

Provides comprehensive technical descriptions of system features and parameters. Includes the applications and benefits, feature interactions, administration requirements, hardware and software requirements, and a brief discussion of data communications and private networking configurations.

DEFINITY Communications System Generic 3 Version 4 Release 3 Implementation, Issue 2 **555-230-655**

DEFINITY Communications System Generic 3 V2/V3 Implementation, Issue 1 Addendum and Addendum 2 **555-230-653**

Provides step-by-step procedures for preparing the hardcopy forms that correspond to the screens and are required to implement the features, functions, and services of the system. Includes procedures for completing a communications survey. Includes an initial set of blank forms (555-230-655B, 555-230-653B).

DEFINITY Communications System Generic 3 Version 4 Implementation Blank Forms, Issue 2 **555-230-655B**

DEFINITY Communications System Generic 3 V2/V3 Implementation Blank Forms, Issue 1 **555-230-653B**

Provides additional blank hardcopy forms that correspond to the screens that are required to implement the features, functions, and services of the system.

Copies of these forms are automatically included with the *DEFINITY Communications System Generic 3 Version 4 Implementation, Issue 1*, 555-230-655 or *DEFINITY Communications System Generic 3 V2/V3 Implementation, Issue 1*, 555-230-653. Use this order number to purchase additional forms.

DEFINITY Communications System Generic 3 System Description and Specifications, Issue 4 **555-230-206**

Provides a technical description of the systems and is intended for service personnel, sales personnel, and customers who need a comprehensive overview of the system. Includes descriptions of hardware, software features, technical specifications, environment requirements, maintenance requirements, and illustration of components.

DEFINITY Communications System Generic 3 Version 4 Traffic Reports, Issue 2 **555-230-511**

Provides detailed descriptions of all the measurement, status, and security reports available in the system and is intended for administrators who validate traffic reports and evaluate system performance. This document was titled *System Reports* for earlier systems. Includes corrective actions for potential problems.

DEFINITY Communications System Generic 1 and Generic 3 Installation and Test, Issue 6 **555-230-104**

Provides descriptions of the procedures for installing and testing the system's common equipment and adjuncts. Includes setup procedures for the system management terminal, power and grounding requirements, and testing steps. Includes compete details on system wiring. Provides both domestic and international information.

DEFINITY Communications System Generic 3 Installation (for Single-Carrier Cabinets), Issue 1

555-230-894 UK English
555-230-895 German
555-230-896 French
555-230-897 Spanish
555-230-900 Chinese

Provides procedures and information for hardware installation and initial testing of the DEFINITY Communications System Generic 3, models Generic 3i and Generic 3 single-carrier cabinet switches only. The UK version will be shipped with all single-carrier cabinet systems in the US. Some languages may not be available until a future date.

DEFINITY Communications System Generic 3r Upgrades and Additions, Issue 1 **555-230-109**

Provides procedures for an installation technician to convert an existing DEFINITY Communications System Generic 1, Generic 2, Generic 3 Version 1, Generic 3 Version 2, Generic 3 Version 3, or System 75 R1V3 to Generic 3r Version 4. Included are upgrade considerations, lists of required hardware, and step-by-step upgrade procedures. Also included are procedures to add control carriers, switch node carriers, port carriers, circuit packs, auxiliary cabinets, and other equipment.

DEFINITY Communications System Generic 3Gvs and Generic G3si Upgrades and Additions, Issue 1 **555-230-108**

Provides procedures for an installation technician to convert an existing DEFINITY Communications System Generic 1, Generic 2, Generic 3 Version 1, Generic 3 Version 2, Generic 3 Version 3, or System 75 R1V3 to Generic 3vs and G3si Version 4. Included are upgrade considerations, lists of required hardware, and step-by-step upgrade procedures. Also included are procedures to add control carriers, switch node carriers, port carriers, circuit packs, auxiliary cabinets, and other equipment.

DEFINITY Communications System Generic 3r Maintenance, Issue 5 **555-230-105**

DEFINITY Communications System Generic 3i/s/vs Maintenance, Issue 8 **555-204-105**

Provide detailed descriptions of the procedures for monitoring, testing, and maintaining the systems. Included are maintenance commands, step-by-step trouble-clearing procedures, the procedures for using all tests, and explanations of the system’s error codes.

An Introduction to DEFINITY Communications System Generic 3, Issue 1 **555-230-023**

Provides a detailed overview of the system including descriptions of many of the major features, applications, hardware, system capabilities, and the AT&T support provided with the system. This document reflects Generic 3 Version 2 software, but still contains relevant information.

DEFINITY Communications System Generic 3 Planning and Configuration, Issue 2 **555-230-601**

Provides step-by-step procedures for the account team in determining the customer’s equipment and hardware requirements to configure a system according to the customer specifications. Includes detailed requirements and block diagrams. This document reflects Generic 3 Version 2 software, but still contains relevant information.

GBCS Products Security Handbook, Issue 4 **555-025-600**

Provides information about the risks of telecommunications fraud and measures for addressing those risks and preventing unauthorized use of GBCS products. This document is intended for telecommunications managers, console operators, and security organizations within companies.

DEFINITY Communications System and System 75 and System 85 Terminals and Adjuncts Reference, Issue 7 **555-015-201**

Provides descriptions of the peripheral equipment that can be used with System 75, System 85, and DEFINITY Communications System. This document is intended for customers and AT&T account teams for selecting the correct peripherals to accompany a system.

DEFINITY Communications System Generic 1 and Generic 3 Voice Terminal Operations, Issue 1 **555-230-701**

Provides detailed operating instructions for the system features on each type of voice terminal. Included are definitions of the voice features and user requirements.

DEFINITY Communications System Generic 1, Generic 3, and System 75 Voice Terminal Guide Builder, Issue 1 **555-230-755**

Provides capability to produce laser-printed documentation for specific voice terminals. The software is supported by a comprehensive user's guide and on-line help. This product requires a 386 PC, minimum of 6MB disk space, minimum of 4MB RAM, a printer supported by Microsoft GDI printer drive, and Microsoft Windows 3.1 or higher. A mouse is recommended.

Call Center

The following list of documents are Call-Center specific. Refer also to the basic DEFINITY Communications System documents.

DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide, Issue 4 **585-230-520**

Provides information on how to write, use, and troubleshoot vectors, which are command sequences that process telephone calls in an Automatic Call Distribution (ACD) environment. It is provided in two parts: tutorial and reference.

The tutorial provides step-by-step procedures for writing and implementing basic call vector scripts.

The reference includes detailed descriptions of the call vectoring features, vector management, vector administration, adjunct routing, troubleshooting, and interactions with management information systems (including the Call Management System).

DEFINITY Communications System Generic 3 Basic Call Management System (BCMS) Operations, Issue 4 **555-230-704**

Provides detailed instructions on how to generate reports and manage the system and is intended for telecommunications managers who wish to use BCMS reports and system managers responsible for maintaining the system. If Issue 4 is unavailable, use Issue 3.

Networks

The following list of documents are network-specific. Refer also to the basic DEFINITY Communications System documents.

DEFINITY Communications System Generic 3 Wideband Technical Reference, Issue 1 **555-230-230**

Provides detailed information regarding the Wideband Switching feature for the system and is intended for users and technical support personnel involved with the installation, administration, and operation of this feature. This feature provides high speed end-to-end connectivity between customer endpoints where dedicated facilities are not economical or appropriate. The primary function is to support high speed video-conferencing and data applications.

DEFINITY Communications System Generic 2.2 and Generic 3 Version 2 DS1/CEPT1/ISDN PRI Reference Manual, Issue 1 **555-025-107**

Provides a detailed technical description of digital trunks in the DEFINITY Communications Systems. This includes trunks conforming to the DS1 standard (1.544 Mbps) and the CEPT1 standard and all other methods of signalling, including bit-oriented signalling as well as ISDN-PRI signalling. This document includes background information on these topics, information on how digital trunk capabilities have been designed into the DEFINITY Communications System and information for field personnel and customers on how to provision and administer digital trunk capabilities and features. Provides both domestic and international information.

Application Specific

The following list of documents are application-specific. Refer also to the basic DEFINITY Communications System documents.

DEFINITY Communications System Generic 2 to Generic 3 Version 4 Transition Reference, Issue 1 **555-230-636**

Provides detailed descriptions of the difference between features and administrative forms for systems Generic 2 to Generic 3 Version 4 and is intended for AT&T personnel and customers involved in planning upgrades and migrations from an older system. Includes descriptions of new administrative commands.

DEFINITY Communications System Generic 3 CallVisor ASAI Planning Guide, Issue 4 **555-230-222**

Provides procedures and directions for the account team and customer personnel for effectively planning and implementing the CallVisor Adjunct/Switch Application Interface (ASAI) PBX-Host environment. The CallVisor ASAI is a communications interface that allows adjunct processors to access switch features and to control switch calls. It is implemented using an Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI). Included are hardware and software requirements.

DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference, Issue 4 **555-230-221**

Provides detailed layer 3 protocol information regarding the CallVisor Adjunct/Switch Application Interface (ASAI) for the systems and is intended for the library or driver programmer of an adjunct processor to create the library of commands used by the applications programmers. Describes the ISDN message, facility information elements, and information elements.

DEFINITY Communications System Generic 3 CallVisor ASAI Technical Reference, Issue 4 **555-230-220**

Provides detailed information regarding the CallVisor Adjunct/Switch Application Interface (ASAI) for the systems and is intended for the application designer responsible for building and/or programming custom applications and features.

DEFINITY Communications System CallVisor Installation, Administration, and Maintenance of CallVisor ASAI over the Definity LAN Gateway, Issue 1 **555-230-223**

Provides procedures for installation, administration, and maintenance of the CallVisor Adjunct/Switch Application Interface (ASAI) Ethernet application and is intended for system administrators, telecommunications managers, Management Information System (MIS) managers, LAN managers, and AT&T personnel. The ASAI-Ethernet application provides ASAI functionality using 10Base-T Ethernet rather than BRI as a transport media.

DEFINITY Communications System Generic 3 Automatic Call Distribution (ACD) Agent Instructions, Issue 4 — **555-230-722**

Provides information for use by agents after they have completed ACD training. Includes descriptions of ACD features and the procedures for using them.

DEFINITY Communications System Generic 3 Automatic Call Distribution (ACD) Supervisor Instructions, Issue 4 — **555-230-724**

Provides information for use by supervisors after they have completed ACD training. Includes descriptions of ACD features and the procedures for using them.

DEFINITY Communications System Generic 1 and Generic 3 Console Operation, Issue 2 — **555-230-700**

Provides operating instructions for the attendant console. Included are descriptions of the console control keys and functions, call-handling procedures, basic system troubleshooting information, and routine maintenance procedures.

DEFINITY Communications System Generic 1 and Generic 3 Console Quick Reference, Issue 1

555-230-890 UK English
555-230-891 German
555-230-892 French
555-230-893 Spanish
555-230-920 Chinese

Provides operating instructions for the attendant console. Included are descriptions of the console control keys and functions, call handling, basic system-troubleshooting information, and routine maintenance procedures. Some languages may not be available until a future date.

An Introduction to DEFINITY Communications System Generic 3 Hospitality Services, Issue 1 **555-230-021**

Provides an overview of the features available for use by the lodging and health industries to improve their property management and to provide assistance to their employees and clients. Included are brief definitions of many of the system features, descriptions of the hardware, planning considerations, and list of the system capabilities.

DEFINITY Communications System Generic 1 and Generic 3 User's Guide Hospitality Operations, Issue 2 **555-230-723**

Provides step-by-step procedures for using the features available for use by the lodging and health industries to improve their property management and to provide assistance to their employees and clients. Includes detailed descriptions of reports.

Generic 3 V3 to Generic 3 V4 Transition Reference

C

The following table indicates which features are new to G3V4, which have been enhanced for G3V4 and which have not changed. See the individual feature descriptions in Chapter 3 for a detailed explanation of new features. Feature enhancements are summarized later in this appendix and explained in detail in Chapter 3.

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
AAR/ARS Partitioning			X
AAR/ARS Digit Conversion			X
Abandoned Call Search			X
Abbreviated Dialing		X	
Abbreviated Dialing (Enhanced)		X	
Add/Remove Skills			X
Additional External Alarming[1]			
Administrable Language Displays			X
Administrable Logins			X
Administered Connections			X
Administration Without Hardware			X
Advice of Charge	X		

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
Agent Call Handling		X	
Agent Sizing			X
Alphanumeric Dialing			X
Alternate Facility Restriction Levels			X
Answer Detection by Call Classifier			X
Attendant Auto-Manual Splitting			X
Attendant Call Waiting			X
Attendant Control of Trunk Group Access			X
Attendant Direct Extension Selection With Busy Lamp Field			X
Attendant Direct Trunk Group Selection			X
Attendant Display		X	
Attendant Intrusion (Call Offer)			X
Attendant Override of Diversion Features			X
Attendant Priority Queue		X	
Attendant Recall			X
Attendant Release Loop Operation			X
Attendant Room Status			X
Attendant Serial Calling			X
Audible Message Waiting			X
Audio Information Exchange (AUDIX) Interface			X
Authorization Codes			X
Automatic Alternate Routing (AAR)		X	
Automatic Callback			X
Automatic Call Distribution (ACD)		X	
ACD Auto-Available Split (AAS)			X
Automatic Circuit Assurance			X
Automatic Hold			X
Automatic Incoming Call Display			X
Automatic Route Selection (ARS)		X	

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
Automatic Transmission Measurement System			X
Automatic Wakeup			X
Auto-Start/Don't Split			X
Basic Call Management System (BCMS)		X	
Bridged Call Appearance — Multi-Appearance Voice Terminal		X	
Bridged Call Appearance — Single-Line Voice Terminal		X	
Busy Verification of Terminals and Trunks			X
Call-By-Call Service Selection			X
Call Coverage			X
Call Detail Recording (CDR)		X	
Call Forwarding All Calls		X	
Call Forward Busy/Don't Answer	X		
Call Management System (CMS)		X	
Call Park			X
Call Pickup			X
Call Prompting		X	
Call Vectoring		X	
CallVisor Adjunct/Switch Application Interface (ASAI)		X	
Call Waiting Termination			X
Centralized Attendant Service (CAS)			X
Class of Restriction (COR)			X
Class of Service (COS)			X
CDR Account Code Dialing			X
Code Calling Access			X
Conference — Attendant			X
Conference — Terminal			X
Constellation Voice/Data Terminal Support			X
Consult			X

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
Coverage Callback			X
Coverage Incoming Call Identification (ICI)			X
Customer-Provided Equipment (CPE) Alarm			X
Data Call Setup			X
Data Hot Line			X
Data-Only Off-Premises Extensions			X
Data Privacy			X
Data Restriction			X
DCS Alphanumeric Display for Terminals			X
DCS Attendant Control of Trunk Group Access			X
DCS Attendant Direct Trunk Group Selection			X
DCS Attendant Display			X
DCS Automatic Callback			X
DCS Automatic Circuit Assurance (ACA)			X
DCS Busy Verification of Terminals and Trunks			X
DCS Call Coverage	X		
DCS Call Forwarding All Calls			X
DCS Call Waiting			X
DCS Distinctive Ringing			X
DCS Leave Word Calling			X
DCS Multi-Appearance Conference/Transfer			X
DCS Over ISDN-PRI D-Channel			X
DCS Trunk Group Busy/Warning Indication			X
Default Dialing			X
DEFINITY LAN Gateway	X		
Dial Access to Attendant			X
Dial Plan			X
Digital Multiplexed Interface			X
Direct Department Calling (DDC) and Uniform Call Distribution (UCD)			X

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
Direct Inward Dialing (DID)			X
Direct Inward and Outward Dialing (DIOD) — International			X
Distinctive Ringing			X
Do Not Disturb			X
DS1 Trunk Service			X
E1 Trunk Service			X
EIA Interface			X
Emergency Access to the Attendant			X
Enhanced DCS (EDCS)			X
End-to-End Signaling			X
Expert Agent Selection		X	
Extension Number Portability			X
Extended Trunk Access			X
Facility and Non-Facility Associated Signaling			X
Facility Busy Indication			X
Facility Restriction Levels (FRLs)			X
Facility Test Calls		X	
Flexible Billing	X		
Forced Entry of Account Codes			X
Forced Password Aging			X
Generalized Route Selection			X
Go to Cover			X
Hold			X
Hold—Automatic			X
Hot Line Service			X
Hunting			X
Inbound Call Management			X
Individual Attendant Access			X
Information System Network (ISN) Interface			X

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
Integrated Directory			X
Integrated Services Digital Network — Basic Rate Interface			X
Integrated Services Digital Network — Primary Rate Interface			X
Intercept Treatment			X
Intercom — Automatic			X
Intercom — Dial			X
Internal Automatic Answer			X
Inter-PBX Attendant Calls			X
Intraflow and Interflow			X
ISDN BRI 2-Wire Line	X		
Last Number Dialed			X
Leave Word Calling			X
Line Lockout			X
Look Ahead Interflow			X
Loudspeaker Paging Access			X
Loudspeaker Paging Access — Deluxe			X
Malicious Call Trace			X
Manual Message Waiting			X
Manual Originating Line Service			X
Manual Signaling			X
MERLIN./System 25 — Voice Terminal Support (731xH Series)			X
Misoperations Handling (modified)	X		
Modem Pooling			X
Move Agents from CMS		X	
Multi-Appearance Preselection and Preference			X
Multiple Call Handling		X	
Multiple Listed Directory Numbers			X
Music-on-Hold Access			X

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
Names Registration			X
Network Access — Private			X
Network Access — Public			X
Night Service — Hunt Group			X
Night Service — Night Console Service			X
Night Service — Night Station Service			X
Night Service — Trunk Answer from Any Station			X
Night Service — Trunk Group			X
Off-Premises Station			X
PC Interface			X
PC/PBX Connection			X
Personal Central Office Line (PCOL)			X
Personalized Ringing			X
Power Failure Transfer			X
Priority Calling			X
Privacy — Attendant Lockout			X
Privacy — Manual Exclusion			X
Property Management System Interface			X
Pull Transfer			X
QSIG Global Networking		X	
Queue Status Indications			X
Recall Signaling			X
Recent Change History		X	
Recorded Announcement		X	
Recorded Telephone Dictation Access			X
Redirection On No Answer (RONA)			X
Remote Access		X	
Report Scheduler and System Printer			X
Restriction — Controlled			X
Restriction — Fully Restricted Service			X

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
Restriction — Miscellaneous Terminal			X
Restriction — Miscellaneous Trunk			X
Restriction — Toll			X
Restriction — Voice Terminal — Inward			X
Restriction — Voice Terminal — Manual Terminating Line			X
Restriction — Voice Terminal — Origination			X
Restriction — Voice Terminal — Outward			X
Restriction — Voice Terminal — Public			X
Restriction — Voice Terminal — Termination			X
Ringback Queuing			X
Ringer Cutoff			X
Ringing — Abbreviated and Delayed	X		
Rotary Dialing			X
R2-MFC		X	
Security Violation Notification (SVN)			X
Send All Calls			X
Senderized Operation			X
Service Observing		X	
Single-Digit Dialing and Mixed Station Numbering			X
Straightforward Outward Completion			X
Subnet Trunking			X
Switch Based Bulletin Board		X	
System Measurements			X
System Status Report			X
Temporary Bridged Appearance			X
Tenant Partitioning	X		
Terminal Translation Initiation			X
Terminating Extension Group			X
Through Dialing			X

Table C-1. G3V3 to G3V4 Transition Reference

Feature Name	New V4 Feature	Changed for V4	Not Changed for V4
Time of Day Routing			X
Timed Reminder and Attendant Timers			X
Touch-Tone Dialing			X
Transfer			X
Transfer Outgoing Trunk to Outgoing Trunk			X
Traveling Class Marks (TCMs)			X
Trunk Flash			X
Trunk Group Busy/Warning Indicators to Attendant			X
Trunk Identification By Attendant			X
Trunk-to-Trunk Transfer			X
Uniform Dial Plan (UDP)			X
Unrestricted Uniform Dial Plan (UDP)			X
VDN of Origin Announcements		X	
Visually Impaired Attendant Services (VIAS)			X
Voice Message Retrieval			X
Voice Response Integration			X
Voice Terminal Alerting Options			X
Voice Terminal Display			X
VuStats		X	
Wideband Switching			X
World Class Core BRI	X		
World Class Tone Detection			X
World Class Tone Generation			X

1.For information about this feature see the *DEFINITY Communications System G3i/s/vs Maintenance* manual, 555-204-105, or *DEFINITY Communications System G3r Maintenance* manual, 555-230-105.

The following pages list V4 enhancements to existing features.

Abbreviated Dialing

- Programming of group lists by designated users.
- Increased length for person, group and system lists.
- Alternative list numbering option for Group Number and System Number lists.
- Automatic Dialing buttons with direct access to designated number that is not stored on an Abbreviated Dialing list.

Abbreviated Dialing (Enhanced)

- Option of three or four digit list entry numbers.
- Increased capacities

Agent Call Handling

- Forced Multiple Call Handling.

Attendant Display

- Administration of Call Type button to display type of active call.

Attendant Priority Queue

- Assignment of priority by call type within priority queue categories.

Automatic Alternate Routing

- Increased AAR/ARS dialed digit string maximum length to 28.

Automatic Call Distribution

- Forced Multiple Call Handling is available.

Automatic Route Selection

- Increased AAR/ARS dialed digit string maximum length to 28.

Basic Call Management System (BCMS)

- Some increased capacities.
- Form changes for some reports.

Bridged Call Appearance

- Station can be administered with only bridged appearances.
- Message lamp and certain feature buttons can be administered to apply to a specified extension rather than the extension of the terminal they reside on.

- Option that prohibits bridged terminals from bridging on to a call when the call has Data Privacy or Data Restriction enabled.
- A call can appear at a terminal as both a bridged and a redirected call.
- An analog terminal can have a single bridged appearance of a multiappearance voice terminal primary call appearance.
- New interactions with Conference, Facility Busy Indication, and Transfer.

Call Detail Recording

- Call duration can be reported in hours/minutes/seconds with no truncation to tenths of minutes.
- The feat-flag bit can be administered to reflect whether an outgoing ISDN call was reported as interworked by the network.
- Adds incoming ring interval duration field.

Call Forwarding All Calls

- Call Forwarding Override.
- List Call Forwarding command.

Call Management System (CMS)

- See the CentreVu™ Call Management System (CMS) documentation.

Call Prompting

- Allows administration of Call Prompting timeout.
- For changes in vector commands, see Call Vectoring.

Call Vectoring

- Ability to add and delete vector steps on the switch.
- *Route-to number* with coverage.
- Addition of the i-silent keyword to the *wait-time* command.
- Vector initiated Service Observing.
- Passing ANI to CMS for inclusion in the CMS call record.
- Specifying a priority level with the oldest-call-wait conditional.
- Enhanced comparators (<>, >=, and <=) with the *goto* and *route-to* commands as well as use of “none” as an entry for digits checking, and “active” or “latest” VDN thresholds for indirect VDN references.
- The use of wildcards in digit strings for matching on collected digits and ANI or II-digits.
- Vector Routing Tables.
- Multiple Audio/Music Sources for use with the *wait-time* command.

- Rolling Average Speed of Answer (ASA), Expected Wait Time (EWT), VDN Calls, ANI, and II-Digits Routing.
- Sending DTMF tones to a Voice Response Unit.

CallVisor Adjunct/Switch Application Interface (ASAI)

- Retrieve Internally Measured Data used to provide VuStats information to terminals.
- Send DTMF Signals.
- Flexible Billing.
- Redirect Call.
- ASAI-Associated Integrated Directory Database Service.
- Enhanced Event Reports.
- New transport option, DEFINITY LAN Gateway.

Expert Agent Selection

- Forced Multiple Call Handling.
- Message Waiting Lamp by default tracks messages waiting for EAS agent LoginID but can be administered to track messages for physical terminal.
- With inspect button can display name of the physical terminal where the EAS agent is logged in.

Facility Test Calls

- Logoff Notification to notify system administrator that Facility Test Calls is still enabled.

Move Agent from CMS

- Can change agents' split or skill assignments while the agents are logged in.
- With EAS, one skill can be added, deleted or moved simultaneously for a group of up to 32 agents.

Multiple Call Handling

- Forced Multiple Call Handling, which forces an agent to be interrupted with an additional ACD call from a split or skill.

QSIG Global Networking

- Adds Call Forwarding (Diversion) and Call Transfer supplementary services.

R2-MFC

- Supports incoming ANI.

Recent Change History

- Recent Change History Report lists each time a user logs in or off the system.

Recorded Announcement

- Multiple Integrated Announcement boards can be installed.
- External lineside T1 (DS1) connected announcements can be installed.

Remote Access

- Logoff Notification to notify system administrator that Remote Access is still enabled.
- Status remote-access command displays status of feature and of remote access barrier codes.

Service Observing

- Vector Initiated Service Observing.

Switch Based Bulletin Board

- Reserves space for AT&T high-priority messages and prompts users when high-priority messages have been entered.

VDN of Origin Announcement

- Agent is connected to call after VDN of Origin Announcement finishes playing.
- In Auto Answer mode, agent hears zip tone before and after the message.
- Agent can shorten message by pressing flashing call appearance button while hearing message.

VuStats

- Can display historical data accumulated over an administered number of intervals.
- VuStats button lamp flashes when administered threshold is reached.

Abbreviations

A

AA
Archangel

AAR
Automatic Alternate Routing

AC
Alternating Current

ACA
Automatic Circuit Assurance

ACB
Automatic Callback

ACD
Automatic Call Distribution

ACU
Automatic Call Unit

ACW
After Call Work

AD
Abbreviated Dialing

ADAP
AUDIX Data Acquisition Package

ADM
Asynchronous Data Module

ADU
Asynchronous Data Unit

AE
Access Endpoint

AIM
Asynchronous Interface Module

AIOD
Automatic Identification of Outward Dialing

ALM-ACK
Alarm Acknowledge

AMW
Automatic Message Waiting

AN
Analog

ANI
Automatic Number Identification

AOL
Attendant Offered Load

AP
Applications Processor

APLT
Advanced Private Line Termination

ARS
Automatic Route Selection

ASAI
Adjunct Switch Applications Interface

ASCII
American Standard Code for Information Interchange

ATB
All Trunks Busy

ATD
Attention Dial

ATMS
Automatic Transmission Measurement System

AUDIX
Audio Information Exchange

AUX
Auxiliary

AVD
Alternate Voice/Data

AWOH
Administration Without Hardware

AWT
Average Work Time

B

BCC
Bearer Capability Class

BCMS
Basic Call Management System

BCT
Business Communications Terminal

BHCC
Busy Hour Call Completions

BLF
Busy Lamp Field

BN
Billing Number

BOS
Bit Oriented Signaling

BPN
Billed Party Number

BPS
Bits Per Second

BRI
Basic Rate Interface

BTU
British Thermal Unit

C

CA-TSC
Call-Associated Temporary Signaling Connection

CACR
Cancellation of Authorization Code Request

CAG
Coverage Answer Group

CAMA
Centralized Automatic Message Accounting

CARR-POW
Carrier Port and Power Unit for AC Powered Systems

CAS
Centralized Attendant Service, Call Accounting System

CBC
Call-By-Call and Coupled Bonding Conductor

CC
Country Code

CCIS
Common Channel Interoffice Signaling

CCITT
Consultative Committee for International Telephone and Telegraph

CCMS
Common Channel Message Set

CCS
Centum (Hundred) Call Seconds

CCSA
Common Control Switching Arrangement

CDM
Channel Division Multiplexing

CDOS
Customer-Dialed and Operator Serviced

CDR
Call Detail Recording

CDRP
Call Detail Record Poller

CDRR
Call Detail Recording and Reporting

CDRU
Call Detail Recording Utilities

CEM
Channel Expansion Multiplexing

CEPT1
European Conference of Postal and Telecommunications Rate 1

CI
Clock Input

cm
Centimeter

CM
Connection Manager

CMDR
Centralized Message Detail Recording

CMS
Call Management System

CO
Central Office

COR
Class of Restriction

COS
Class of Service

CP
Circuit Pack

CPE
Customer Premises Equipment

CPN
Called-Party Number

CPN/BN
Calling Party Number/Billing Number

CPTR
Call Progress Tone Receiver

CRC
Cyclical Redundancy Checking

CSA
Canadian Safety Association

CSCN
Center Stage Control Network

CSD
Customer Service Document

CSM
Centralized System Management

CSS
Center Stage Switch

CSSO
Customer Services Support Organization

CSU
Channel Service Unit

CTS
Clear to Send

CWC
Call Work Codes

D

DAC
Dial Access Code or Direct Agent Calling

dB
Decibel

DC
Direct Current

DCE
Data Communications Equipment

DCP
Digital Communications Protocol

DCS
Distributed Communications System

DDC
Direct Department Calling

DDD
Direct Distance Dialed

DID
Direct Inward Dialed

DIOD
Direct Inward and Outward Dialing

DIVA
Data In/Voice Answer

DLC
Data Line Circuit

DLDM
Data Line Data Module

DMI
Digital Multiplexed Interface

DND
Do Not Disturb

DNIS
Dialed Number Identification Service

DOD
Direct Outward Dialing

DOSS
Delivery Operations Support System

DOT
Duplication Option Terminal

DPM
Dial Plan Manager

DPR
Dual Port RAM

DS1
Data Services Level 1

DS1C
Digital Signal Level-1 Converter

DSI
Digital Signal Interface

DSU
Data Service Unit

DTDM
Digital Terminal Data Module

DTE
Data Terminal Equipment

DTGS
Direct Trunk Group Select

DTMF
Dual-Tone Multifrequency

DTS
Disk Tape System

DXS
Direct Extension Selection

E

E&M
Ear and Mouth (receive and transmit)

EAA
Expansion Archangel

EAL
Expansion Archangel Link

EBCDIC
Extended Binary-Coded Decimal Interexchange Code

ECC
Error Correct Code

EFP
Electronic Power Feed

EI
Expansion Interface

EIA
Electronic Industries Association

EMI
Electro-Magnetic Interference

EPN
Expansion Port Network

EPROM
Erasable Programmable Read Only Memory

EPSCS
Enhanced Private Switched Communications Services

ESF
Extended Superframe Format

ETA
Extended Trunk Access

ETN
Electronic Tandem Network

ETSI
European Telecommunications Standards Institute

F

FAC
Feature Access Code

FAS
Facility-Associated Signaling

FAT
Facility Access Trunk

FAX
Facsimile

FCC
Federal Communications Commission

FEAC
Forced Entry of Account Codes

FEP
Front End Processor

FIC
Facility Interface Codes

FNPA
Foreign Numbering-Plan Area

FRL
Facilities Restriction Level

FX
Foreign Exchange

G

G1
Generic1

G3-MA
Generic 3 Management Applications

G3-MT
Generic 3 Management Terminal

G3i
Generic 3, Intel

G3i-G
Generic 3, global

G3r
Generic 3, RISC (Reduced Instruction Set Computer)

GM
Group Manager

GPTR
General-Purpose Tone Receiver

GRS
Generalized Route Selection

H

HNPA
Home Numbering Plan Area Code

Hz
Hertz

I

IAS
Inter-PBX Attendant Service

IC
Inter-Cabinet

ICC
Intercarrier Cable

ICD
Inbound Call Director

ICDOS
International Customer Dialed Operator Serviced

ICHT
Incoming Call-Handling Table

ICI
Incoming Call Identifier

ICM
Inbound Call Management

IDDD
International Direct Distance Dialing

IDF
Intermediate Distribution Frame

IE
Information Element

IMT
Intermachine Trunk

in
Inch

INADS
Initialization and Administration System

INS
ISDN Network Service

INWATS
Inward Wide Area Telephone Service

IO
Information Outlet

ISDN
Integrated Services Digital Network

ISN
Information Systems Network

ISO
International Standards Organization

ISV
Independent Software Vendor

ITP
Installation Test Procedures

ITU
International Telecommunications Union

IXC
Interexchange Carrier Code

K

kHz
Kilohertz

kbps
Kilobits Per Second

kbyte
Kilobyte

kg
Kilogram

L

LAN
Local Area Network

LAP-D
Link Access Procedure on the D-channel

LAPD
Link Access Procedure Data

LATA
Local Access and Transport Area

lb
Pound

LDN
Listed Directory Number

LDS
Long-Distance Service

LEC
Local Exchange Carrier

LED
Light-Emitting Diode

LINL
Local Indirect Neighbor Link

LMM
Low Level Maintenance Monitor

LSU
Local Storage Unit

LWC
Leave Word Calling

M

M-Bus
Memory Bus

MA-UUI
Message Associated User-to-User Signaling

MADU
Modular Asynchronous Data Unit

MAP
Maintenance Action Process

Mbps
Megabits Per Second

Mbyte
Megabytes

MCC
Multi-Carrier Cabinet

MCS
Message Center Service

MDF
Main Distribution Frame

MDM
Modular Data Module

MDR
Message Detail Record

MEM
Memory

MET
Multibutton Electronic Telephone

MFC
Multifrequency Compelled Signaling

MHz
Megahertz

MIM
Management Information Message

MIS
Management Information System

MISCID
Miscellaneous Identification

MMS
Material Management Services

MOS
Message-Oriented Signaling

MPDM
Modular Processor Data Module

MS
Message Server

ms
Millisecond

MS/T
Main Satellite/Tributary

MSA
Message Servicing Adjunct

MSG
Message Service

MSM
Modular System Management

MSS
Mass Storage System

MSSNET
Mass Storage/Network Control

MT
Management Terminal

MTDM
Modular Trunk Data Module

MTP
Maintenance Tape Processor

MTT
Multi-Tasking Terminal

MWL
Message Waiting Lamp

N

NANP
North American Numbering Plan

NAU
Network Access Unit

NCA/TSC
Non-Call Associate/Temporary Signaling Connection

NCOSS
Network Control Operations Support Center

NCSO
National Customer Support Organization

NEC
National Engineering Center

NEMA
National Electrical Manufacturer's Association

NFAS
Non-Facility Associated Signaling

NID
Network Inward Dialing

NM
Network Management

NN
National Number

NPA
Numbering Plan Area

NPE
Network Processing Element

NQC
Number of Queued Calls

NSE
Night Service Extension

NSU
Network Sharing Unit

NXX
Public Network Office Code

O

OA
Operator Assisted

OCM
Outbound Call Management

ONS
On-Premises Station

OPS
Off-Premises Station

OQT
Oldest Queued Time

OSHA
Occupational Safety and Health Act

OSI
Open Systems Interconnect

OSS
Operations Support System

OSSI
Operations Support System Interface

OTQ
Outgoing Trunk Queuing

P

PACCON
Packet Control

PAD
Packet Assembly/Disassembly

PBX
Private Branch Exchange

PC
Personal Computer

PCM
Pulse Code Modulated

PCOL
Personal Central Office Line

PCOLG
Personal Central Office Line Group

PCS
Permanent Switched Calls

PDM
Processor Data Module

PDS
Premises Distribution System

PE
Processing Element

PEC
Price Element Codes

PEI
Processor Element Interchange

PGATE
Packet Gateway

PGN
Partitioned Group Number

PI
Processor Interface

PIB
Processor Interface Board

PIDB
Product Image Database

PKTINT
Packet Interface

PL
Private Line

PLS
Premises Lightwave System

PMS
Property Management System

PN
Port Network

PNA
Private Network Access

POP
Point Of Presence

PPN
Processor Port Network

PRI
Primary Rate Interface

PROCR
Processor

PSC
Premises Service Consultant

PSDN
Packet Switch Public Data Network

PT
Personal Terminal

PTC
Positive Temperature Coefficient

PTT
Postal Telephone and Telegraph

R

R2-MFC
R2 Multifrequency Compelled Signaling

RAM
Random Access Memory

RBS
Robbed-Bit Signaling

RCL
Restricted Call List

RHNPA
Remote Home Numbering Plan Area

RINL
Remote Indirect Neighbor Link

RISC
Reduced Instruction Set Computer

RLT
Release Link Trunk

RMATS
Remote Maintenance, Administration, and Traffic System

RNX
Route Number Index (Private Network Office Code)

ROM
Read-Only Memory

RPN
Routing Plan Number

RS232C
Recommended Standard 232C

RS449
Recommended Standard 449

RSC
Regional Support Center

S

SABM
Set Asynchronous Balance Mode

SAKI
Sanity and Control Interface

SAT
System Access Terminal

SCC
Single Carrier Cabinet

SCD
Switch-Control Driver

SCI
Switch Communications Interface

SCO
System Control Office

SCOTCH
Switch Conferencing for TDM Bus in Concentration Highway

SCSI
Small Computer System Interface

SDDN
Software Defined Data Network

SDI
Switched Digital International

SDLC
Synchronous Data Link Control

SDN
Software Defined Network

SID
Station Identification Number

SIT
Special Information Tones

SMDR
Station Message Detail Recording

SMM
Standby Maintenance Monitor

SN
Switch Node

SNA
Systems Network Architecture

SNC
Switch Node Clock

SNI
Switch Node Interface

SPE
Switch Processing Element

SPID
Service Profile Identifier

SSI
Standard Serial Interface

SSM
Single Site Management

SSV
Station Service

ST3
Stratum 3 Clock Board

STARLAN
Star-Based Local Area Network

SVN
Security Violation Notification

SXS
Step-by-Step

SYSAM
System Access and Administration

T

TAAS
Trunk Answer from Any Station

TAC
Trunk Access Code

TC
Technical Consultant

TCM
Traveling Class Mark

TDM
Time-Division Multiplex(ing)

TDR
Time of Day Routing

TEG
Terminating Extension Group

TEI
Terminal Endpoint Identifier

TOD
Time of Day

TOP
Task Oriented Protocol

TSC
Technical Service Center

TTR
Touch-Tone Receiver

TTT
Terminating Trunk Transmission

TTTN
Tandem Tie Trunk Network

TTY
Teletypewriter

U

UAP
Usage Allocation Plan

UART
Universal Asynchronous Transmitter

UCD
Uniform Call Distribution

UCL
Unrestricted Call List

UDP
Uniform Dial Plan

UL
Underwriter Laboratories

UM
User Manager

UNMA
Unified Network Management Architecture

UNP
Uniform Numbering Plan

UPS
Uninterruptible Power Supply

USOP
User Service Order Profile

UUCP
UNIX-to-UNIX Communications Protocol

UUI
User-to-user information

V

VDN
Vector Directory Number

VIS
Voice Information System

VLSI
Very Large Scale Integration

VM
Voltmeter

VNI
Virtual Nodepoint Identifier

W

WATS
Wide Area Telecommunications Service

WCC
World Class Core

WSA
Waiting Session Accept

Z

ZCS
Zero Code Suppression

Glossary

Numerics

3B2 Message Server
An AT&T software application that combines voice and data messaging services for voice terminal users whose extensions are connected to a G3 switch.

800 service
A service in the USA, which allows incoming calls from a certain area or areas to an assigned number for a flat-rate charge based on usage.

A

AAR
See Automatic Alternate Routing on page GL-5.

ARS
See Automatic Route Selection on page GL-5.

abandoned call
An incoming call, where the caller hangs up before being answered.

access code
A 1-, 2-, or 3-digit dial code used to activate or cancel a feature, or access an outgoing trunk. The star (*) and pound (#) can be used as the first digit of an access code.

access endpoint
Either a nonsignaling channel on a DS1 interface or a nonsignaling port on an analog tie trunk circuit pack that is assigned a unique extension.

access tie trunk
A trunk that connects a main communications system with a tandem communications system in an electronic tandem network (ETN). An access tie trunk can also be used to connect a system or tandem to a serving office or service node. Also called ‘‘access trunk.’’

ACCUNET
A trademarked name for a family of digital services offered by AT&T in the USA.

ACD
See **Automatic Call Distribution**. ACD also refers to the ‘‘Work State’’ indicating that the agent is on an ACD call.

ACD split (or split)
A group of extensions that are staffed by agents trained to handle a certain type of incoming call. Valid split numbers range from 1~through 99. Each number identifies a unique grouping of ACD agent positions. ACD split is also referred to as an *ACD hunt group* or *hunt group*.

ACD work modes
See **work modes**.

active-notification association
A ''link'' that is initiated by the adjunct allowing it to receive Event Reports for a specific switch entity, for example, an outgoing call. This association is initiated by the adjunct via the *Event Notification Request* capability.

active-notification call
A call for which Event Reports are being sent over an active-notification association (communication channel) to the adjunct. Sometimes referred to as a monitored call.

active notification domains
Domains are VDNs and ACD split extensions for which Event Notification has been requested.

adjunct-control association
A relationship initiated by an application via the *Third Party Make Call*, the *Third Party Take Control* or *Domain (Station) Control* capabilities to set up calls and control calls already in progress.

adjunct-controlled calls
Include all the calls that can be controlled using an adjunct-control association. These calls must have been originated via the *Third Party Make Call* or *Domain (Station) Control* capabilities or must have been taken control of via the *Third Party Take Control* or *Domain (Station) Control* capabilities.

adjunct-controlled splits
ACD splits administered to be under adjunct control. Agents logged into such splits must do all telephony and ACD login and/or logout and change work mode functions through the adjunct (except for auto-available adjunct controlled splits, whose agents may not be logged in and/or logged out or have their work modes changed).

adjunct-monitored calls
Include all the adjunct-controlled calls and the active-notification calls. In addition it includes calls which provide event reporting over domain-control associations.

application
An application refers to an adjunct entity that requests and receives ASAI services or capabilities. One or more applications can reside on a single adjunct. However, the switch cannot distinguish among several applications residing on the same adjunct and treats the adjunct, and all resident applications, as a single application. The terms application and adjunct are used interchangeably throughout this document.

after call work (ACW) mode
In this mode, agents are unavailable to receive ACD calls. Agents should enter the ACW mode to perform ACD-related activities such as filling out a form after an ACD call. If agents are in the Manual-In mode and disconnect from an ACD call, they automatically enter the ACW mode. Agents who normally use Auto-In mode can enter the ACW state by depressing the ACW button while on a call.

adjunct
A processor that does one or more tasks for another processor and that is optional in the configuration of the other processor.

adjunct-switch application interface (ASAI)
An AT&T recommendation for interfacing adjuncts and communications systems, based on the CCITT Q.932 specification for layer 3.

administer
To access and change parameters associated with the services or features of a system.

Administered Connection (AC)
Administered Connection is a feature that allows the switch to automatically establish and maintain end-to-end connections between access endpoints (trunks) and/or data endpoints (data modules).

administration terminal
A terminal used to administer and maintain a system. See also **terminal**.

Administration Without Hardware (AWOH)
Provides the ability to administer ports without the need for the associated terminals or other hardware to be physically present.

abandoned call
An incoming call, where the caller hangs up before being answered.

agent (or ACD agent)
An answering position who receives calls that are directed to a split. A member of an ACD hunt group (ACD split).

agents in multiple splits
An agent may be logged into more than one split (three maximum). If, while logged into more than one split, the agent (1) answers an ACD call, (2) is in ACW mode for any split, or (3) makes or receives a direct extension call, the switch will not distribute additional ACD calls to that agent.

agent report
Provides historical traffic information for internally measured agents.

American National Standard Code for Information Interchange
See **ASCII**.

analog
The representation of information by means of continuously variable physical quantities such as amplitude, frequency, and phase.

analog data
Data that is transmitted over a digital facility in analog (pulse code modulation) form. The data must pass through a modem either at both ends or at a modem pool at the distant end.

analog telephone
A telephone that receives acoustic voice signals and sends analog electrical signals along the telephone line. Analog telephones are usually served by a single wire pair (tip and ring). The model-2500 telephone set is a typical example of an analog telephone.

analog-to-digital converter (ADC)
A device that converts an analog signal to digital form. See also **digital-to-analog converter**.

angel
A microprocessor located on each port card in a processor port network (PPN). The angel uses the control-channel message set (CCMS) to manage communications between the port card and the archangel on the controlling switch processing element (SPE). The angel also monitors the status of other microprocessors on a port card and maintains error counters and thresholds. See also **archangel**.

answerback code
An assigned number used to respond to a page from a code-calling or loudspeaker-paging system, or to retrieve a parked call.

appearance
A software process that is associated with an extension and whose purpose is to supervise a call. Also called "call appearance," "line appearance," and "occurrence."

applications processor
A minicomputer used with several user-controlled applications such as traffic analysis and electronic documentation.

architecture
The organizational structure of a system, including hardware and/or software.

ASCII (American National Standard Code for Information Interchange)
The standard code, using a coded character set consisting of 7-bit coded characters (eight bits, including parity check), used for information interchange among data processing systems, data communications systems, and associated equipment. The ASCII set consists of control characters and graphic characters.

asynchronous data transmission
A method of transmitting data in which each character is preceded by a start bit and followed by a stop bit, thus permitting data characters to be transmitted at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard). Also called "asynchronous transmission." See also **synchronous data transmission**.

association
An association is a communication channel between the adjunct and switch for messaging purposes. An active association is one which applies to an existing call on the switch or to an extension on the call.

asynchronous data unit (ADU)
A data communications equipment (DCE) type device that allows direct connection between RS232C equipment and a digital switch.

attendant
A person at a console on a customer's premises who provides personalized service for incoming callers and voice-services users by performing switching and signaling operations. See also **attendant console**.

attendant console
The workstation used by an attendant. The attendant console allows the attendant to originate a call, answer an incoming call, transfer a call to another extension or trunk, put a call on hold, and remove a call from hold. Attendants using the console can also manage and monitor some system operations. Also called "console." See also **attendant**.

Audio Information Exchange (AUDIX)
A fully integrated voice-mail system that can be used with a variety of communications systems to provide call-history data, such as subscriber identification and reason for redirection.

auto-in trunk groups
Those trunk groups where the CO processes all of the digits for the incoming call. Whenever the switch determines that the CO has seized a trunk from an Auto-In trunk group, it automatically (without processing any digits) connects the trunk to the destination. The destination will typically be an ACD split where(emif there are no agents available(emthe call will go into a queue in which the callers wait to be answered in the order in which they arrived.

auto-in work mode
One of four agent work modes. The work mode where an agent indicates, to the system, that the agent is ready to process another call as soon as the current call is completed. Specifically, if an

agent disconnects from an ACD call while in Auto-in Work Mode, then that agent immediately becomes available to receive another ACD call. *See* **Manual-In Work Mode** for a contrast.

Automatic Alternate Routing

Automatic Call Distribution (ACD) split

Calls of a similar type are distributed among agents.

Automatic Route Selection

The software program that controls call routing over an ETN is called Automatic Alternate Routing (AAR). AAR not only determines the route for a call, but, through the Facilities Restriction Level (FRL) function, defines up to eight levels of calling privileges for users of the ETN. Another function of AAR, Subnet Trunking, can convert an on-network number to a public network or international number. This function is useful when all on-network routes are busy or are not provided.

automatic trunk

A trunk that does not require the sending or receiving of addressing information because the destination is predetermined. A request for service on the trunk, called a ''seizure,'' is sufficient to route the call. The normal destination of an automatic trunk is the communications-system attendant group. Also called ''automatic incoming trunk'' and ''automatic tie trunk.''

automatic restoration

A service that restores disrupted connections between access endpoints (nonsignaling trunks) and data endpoints (devices that connect the switch to data terminal and/or communications equipment). This restoration is done within seconds of a service disruption so that critical data applications can remain operational.

auxiliary equipment

Equipment used for optional system features, such as Loudspeaker Paging and Music-on-Hold.

auxiliary trunk

A trunk used to connect auxiliary equipment, such as radio-paging equipment, to a communications system.

aux-work mode

In this mode, agents are unavailable to receive ACD calls. Agents should enter aux-work mode when involved in non-ACD activities such as taking a break, going to lunch, or placing an outgoing call.

When agents log in, they are automatically placed in the Aux-Work mode. They can then use the Auto-In or Manual-In feature to make themselves available to answer the first call.

Also, the last available agent in a split cannot enter the aux-work mode if any ACD calls are remaining in the queue. If the last available agent attempts to enter aux-work mode, the following occurs: (1) Calls in the queue are routed to the agent until the queue is empty (2) If the last available agent has an aux-work button, the light next to the button flashes until all calls in the queue are answered. When the last call is answered, the light next to the button goes on steadily, and the agent then enters aux-work mode.

B

bandwidth

The difference, expressed in Hertz, between the defined highest and lowest frequencies in a frequency range.

barrier code
A security code used with the Remote Access feature to prevent unauthorized access to the system.

baud
In telecommunications applications, a unit of transmission speed equal to the number of signal events per second. See also **bit rate** and **bits per second**.

BCC
The Bearer Capability Class (BCC) identifies the type of a call, for example, voice and different types of data. Determination of BCC is based on the call originator’s characteristics for non-ISDN endpoints and on the Bearer Capability and Low-Layer Compatibility Information Elements of an ISDN endpoint.

Current BCCs are:

- 0: Voice-grade data and voice
- 1: DMI Mode 1, 56 kbps data transmission
- 2: DMI Mode 2, synchronous/asynchronous data transmission up to 19.2 kbps
- 3: DMI Mode 3, 64 kbps circuit/packet data transmission
- 4: DMI Mode 0, 64 kbps synchronous data
- 5: Temporary Signaling Connection
- 6: Wideband Call, 128 to 1984 kbps synchronous data

bit (binary digit)
One unit of information in binary notation having two possible states or values, 0 or 1.

bits per second (bps)
The number of binary units of information that are transmitted or received per second. See also **baud** and **bit rate**.

bit rate
The speed at which bits are transmitted, usually expressed in bits per second. Also called ‘‘data rate.’’ See also **baud** and **bits per second**.

bridge (bridging)
The appearance of a voice terminal’s extension at one or more other voice terminals.

BRI
The ISDN Basic Rate Interface specification.

bridged appearance
A call appearance on a voice terminal that matches a call appearance on another voice terminal for the duration of a call.

buffer
(1) In hardware, a circuit or component that isolates one electrical circuit from another. Typically, a buffer holds data from one circuit or process until another circuit or process is ready to accept the data. (2) In software, an area of memory used for temporary storage.

bus
A multiconductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

business communications terminal (BCT)
An integrated digital data terminal used for business applications. A BCT can function via a digital terminal data module (DTDM) or a processor data module (PDM) as a special-purpose terminal for services provided by an applications processor (AP) or, as a terminal for data entry and retrieval.

BX.25
An AT&T version of the CCITT X.25 protocol for data communications. BX.25 adds a fourth level to the standard X.25 interface. This uppermost level combines levels 4, 5, and 6 of the International Standards Organization (ISO) reference model.

bypass tie trunks
A one-way, outgoing tie trunk from a tandem switch to a main switch in an electronic tandem network (ETN). Bypass tie trunks, provided in limited quantities, are used as a ‘‘last-choice’’ route when all trunks to another tandem switch are busy. Bypass tie trunks are used only if all applicable intertandem trunks are busy.

byte
A sequence of (usually eight) bits processed together.

C

cabinet
Housing for racks, shelves, or carriers that hold electronic equipment.

cable
The physical connection between two pieces of equipment (em for example, cable from a data terminal to a modem (em or between a piece of equipment and a termination field (em for example, circuit pack I/O cables.

cable connector
A cable connector is either a jack (female) or plug (male) on the end of a cable. A cable connector connects wires on a cable to specific leads on telephone or data equipment.

call appearance, attendant console
Six buttons, labeled ‘‘a’’ through ‘‘f,’’ and used to originate, receive, and hold calls. Each button has two lights to show the status of the call appearance.

call appearance, voice terminal
A button labeled with an extension number and used to place outgoing calls, receive incoming calls, or hold calls. Two lights next to the button show the status of the call appearance or the status of the call.

call control capabilities
call control capabilities are all the capabilities (*Third Party Selective Hold, Third Party Reconnect, Third Party Merge*) that can be used in either of the Third Party Call Control ASE (cluster) subsets: Call Control and Domain Control.

call detail recording
A switch feature that utilizes software and hardware to record call data (same as CDRU).

call detail recording utility (CDRU)
Applications software that collects, stores, optionally filters, and outputs call detail records for direct or polled output to peripheral devices.

call management system (CMS)
An application, running on an adjunct processor, that collects information from an Automatic Call Distribution (ACD) unit. CMS enables customers to monitor and manage telemarketing centers by generating reports on the status of agents, splits, trunks, trunk groups, vectors, and vector directory numbers (VDNs), and enables customers to partially administer the ACD feature for a communications system.

call reference value (CRV)
An identifier present in ISDN messages that serves to associate a related sequence of messages. In ASAI, the CRVs distinguish between associations.

call vector
A set of up to 15 vector commands to be performed for an incoming or internal call.

callback call
A call that is automatically returned to a voice terminal user who activated the Automatic Callback or Ringback Queuing feature.

call-waiting ringback tone
A low-pitched tone identical to ringback tone except that the tone decreases in the last 0.2~second (in the United States). A call-waiting ringback tone notifies the attendant that the Attendant Call Waiting feature has been activated and that the called user is aware of the waiting call. Tones in international countries may sound different.

call work code
A number, up to 16 digits, entered by Automatic Call Distribution (ACD) agents to record the occurrence of customer-defined events (such as account codes, social security numbers, or phone numbers) on ACD calls.

carrier
An enclosed shelf containing vertical slots that hold circuit packs.

carried load
The amount of traffic actually served by traffic-sensitive facilities during a given interval.

CCS or hundred call seconds
A unit of traffic measure that is used to determine usage. In order to determine usage for a facility, it is scanned every 100 seconds. If the facility is found busy, then it is assumed to have been busy for the entire scan interval. There are 3600 seconds per hour. The Roman numeral for 100 is the capital letter ‘‘C.’’ The abbreviation for call seconds is CS. Therefore, 100 call seconds is abbreviated as CCS. If a facility is busy for an entire hour, then it is said to have been busy for 36 CCS. *See also* **Erlang**.

capability
A capability is a either a request or indication of an operation. For example, a *Third Party Make Call* is a request for setting-up a call and an *Event Report* is an indication that an event has occurred.

capability groups
Capability groups are sets of capabilities, provisioned through switch administration, that can be requested by an application. Each capability group may contain capabilities from several capability groups. Capability groups are also referred to, in other documentation, as administration groups or Application Service Elements (ASEs). Capability groups denote association types. For example, *Call Control* is a type of association which allows certain functions (the ones in the capability group) to be performed over this type of association.

cause value

A Cause Value is returned in responses to requests or in event reports when a denial occurs or an unexpected condition is encountered. ASAI cause values fall into two ‘‘coding standards’’: Coding Standard 0 includes any cause values that are part of AT&T and CCITT ISDN specifications, and, Coding standard 3 includes any other ASAI cause values. This document uses a notation for cause value where the coding standard for the cause is given first, then a slash, then the cause value. For example, CS0/100 is coding standard 0, cause value 100.

CCITT

CCITT (Comitte Consultatif International Telephonique et Telegraphique) is now called *International Telecommunications Union* (ITU). See this name for information.

center stage switch (CSS)

The central interface between the processor port network (PPN) and expansion port networks (EPNs) in a CSS-connected system.

central office (CO)

The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

central office (CO) codes

The first three digits of a 7-digit public network telephone number in the USA. CO codes are numbered from 200 through 999.

central office (CO) trunk

A telecommunications channel that provides access from the system to the public network through the local CO.

channel

The term channel is nonspecific and must be taken in context. Channel can refer to a circuit-switched call or a communications path for transmitting voice and/or data.

In wideband, a channel refers to all of the time slots necessary to support a call. For example, an H0-channel uses six 64 kbps time slots. This definition of channel is the same whether the time slots necessary to support the call are contiguous or noncontiguous.

Channel can also refer to a DS0 on a T1 or E1 facility not specifically associated with a logical circuit-switched call. In this context, a channel is analogous to a single trunk.

channel negotiation

Channel negotiation is the process by which the channel offered in the Channel Identification Information Element (CIIE) in the SETUP message is ‘‘negotiated’’ to be another channel acceptable to the switch receiving the SETUP message and ultimately to the switch that sent the SETUP. Negotiation will only be attempted if the CIIE is encoded as *Preferred*. Channel negotiation will not be attempted for wideband calls.

circuit

(1) An arrangement of electrical elements through which electric current flows, providing one or more specific functions. (2) A channel or transmission path between two or more points.

circuit pack

A card on which electrical circuits are printed, and integrated circuit (IC) chips and electrical components are installed. A circuit pack is installed in a switch carrier.

Class of Restriction (COR)

A feature that allows up to 64 classes of call-origination and call-termination restrictions for voice terminals, voice terminal groups, data modules, and trunk groups. See also **Class of Service (COS)**.

Class of Service (COS)
A feature that uses a number (0 through 15) to specify if voice terminal users can activate the Automatic Callback, Call Forwarding(emAll Calls, Data Privacy, or Priority Calling features.

common control switching arrangement (CCSA)
A private telecommunications network using dedicated trunks and a shared switching center for interconnecting company locations.

communications system
The software-controlled processor complex that interprets dialing pulses, tones, and/or keyboard characters and makes the proper interconnections both within the system and external to the system. The communications system itself consists of a digital computer, software, storage device, and carriers with special hardware to perform the actual connections. A communications system provides voice and/or data communications services, including access to public and private networks, for telephones and data terminals on a customer's premises. See also **switch**.

confirmation tone
A tone confirming that a feature activation, deactivation, or cancellation has been accepted.

connectivity
The connection of disparate devices within a single system.

console
See **attendant console**.

contiguous
Contiguous, which is a wideband term, refers to adjacent DS0s within one T1 or E1 facility or adjacent TDM or fiber time slots. Note that the first and last TDM bus, DS0, or fiber time slots are not considered contiguous (no wraparound). For an E1 facility with a %D-channel, DS0s 15 and 17 are considered contiguous.

control cabinet
See **control carrier**.

control carrier
A carrier in a multicarrier cabinet that contains the switch processing element (SPE) circuit packs and, unlike a G3r control carrier, port circuit packs. Also called ''control cabinet'' in a single-carrier cabinet. See also **switch processing element**.

controlled station
A station that is being monitored and controlled via a domain-control association.

coverage answer group
A group of up to eight voice terminals that ring simultaneously when a call is redirected to it by Call Coverage. Any one of the group can answer the call.

coverage call
A call that is automatically redirected from the called party's extension number to an alternate answering position when certain coverage criteria are met.

coverage path
The order in which calls are redirected to alternate answering positions.

coverage point
An extension or attendant group, vector directory number (VDN), or Automatic Call Distribution (ACD) split designated as an alternate answering position in a coverage path.

covering user
A person at a coverage point who answers a redirected call.

critical reliability system
A system that has the following duplicated items: control carriers, tone-clock circuit packs, expansion interface (EI) circuit packs, and cabling between port networks (PNs) and center stage switch (CSS) in a CSS-connected system. See also **duplicated common control**, **duplicate processor-only system**, and **duplication**.

D

data channel
A communications path between two points used to transmit digital signals.

data communications equipment (DCE)
The equipment (em usually a modem, data module, or packet assembler/disassembler (em on the network side of a communications link that provides the functions to make the binary serial data from the source or transmitter compatible with the communications channel.

data link
The configuration of physical facilities enabling end terminals to communicate directly with each other.

data module
An interconnection device between a basic rate interface (BRI) or digital communications protocol (DCP) interface of the switch and data terminal equipment (DTE) or data communications equipment (DCE).

data path
The end-to-end connection used for a data-communications link. A data path is the combination of all the elements of an interprocessor communication in a distributed communications system (DCS).

data port
A point of access to a computer that uses trunks or lines for transmitting or receiving data.

data rate
See **bit rate**.

data service unit (DSU)
A device designed to transmit digital data on transmission facilities.

data terminal
An input/output (I/O) device that has either switched or direct access to a host computer or to an applications processor (AP).

data terminal equipment (DTE)
Equipment consisting of the endpoints in a connection over a data circuit. For example, in a connection between a data terminal and a host, the terminal, the host, and their associated modems or data modules make up the DTE. DTE usually consists of the following functional units: control logic, buffer store, and one or more input or output devices or computers. DTE can contain error control, synchronization, and telephone-identification capabilities.

D-channel backup
D-channel backup is used with Non-Facility Associated Signaling (NFAS). With D-channel backup, a primary D-channel provides signaling for an NFAS D-channel group (two or more PRIs facilities). A second (redundant) %D-channel, located on a separate PRI facility of the NFAS D-channel group is designated as backup for the D-channel. The failure of the primary D-channel

causes an automatic transfer of call-control signaling to the backup D-channel. When this happens, the backup becomes the primary D-channel, and when the previous primary is returned to service it becomes the backup D-channel.

DCS
See Distributed Communications System on page GL-13.

delay-dial trunk
A trunk that allows dialing directly into a communications system (em that is, the digits are received as they are dialed.

denying a request
Denying a Request is the same as sending a negative acknowledgement (NAK), and is done by sending an Facility Information Element (FIE) with a *return error* component (a cause value is also provided). It should not be confused with the ‘‘denial’’ event report which applies to calls.

designated voice terminal
The specific voice terminal to which calls, originally directed to a certain extension number, are redirected. Commonly used to mean the ‘‘forwarded-to’’ terminal when Call Forwarding All Calls is active.

dial-repeating tie trunk
A tie trunk that transmits called-party addressing information between two communications systems.

digit conversion
A process used to convert specific dialed numbers into other dialed numbers.

digital communications protocol (DCP)
An AT&T proprietary protocol used to transmit both digitized voice and digitized data over the same communications link. A DCP link is made up of two 64~kbps information (I-) channels and one 8-kbps signaling (S-) channel.

digital data endpoints
In G3iV2, digital data endpoints include devices such as the 510D terminal or the 515-type business communications terminal (BCT).

digital multiplexed interface (DMI)
An interface that provides connectivity between a communications system and a host computer or between two communications systems using digital signal level-1 (DS1) 24th-channel signaling. DMI provides 23 64~kbps data channels and 1 common signaling channel over a twisted-pair connection. DMI is offered through two capabilities: bit-oriented signaling (DMI-BOS) and message-oriented signaling (DMI-MOS).

digital signal level 0 (DS0)
A single 64 kbps voice channel. A DS0 is a single 64 kbps channel in a T1 or E1 facility and consists of eight bits in a T1 or E1 frame every 125 micro-seconds.

digital terminal data module (DTDM)
An integrated or adjunct data module that shares with a digital telephone the same physical port for connection to a communications system. The function of a DTDM is similar to that of a processor data module (PDM) and modular processor data module (MPDM) in that it converts RS232C signals to DCP signals.

digital-to-analog converter
A device that converts data in digital form to the corresponding analog signals. See also **analog-to-digital converter**.

digital transmission
A mode of transmission in which the information to be transmitted is first converted to digital form and then transmitted as a serial stream of pulses.

digital trunk
A circuit in that carries digital voice and/or digital data in a telecommunications channel.

dial-repeating trunks
A PBX tie trunk that is capable of handling PBX station signaling information without attendant assistance.

direct agent
A switch feature accessed only via Adjunct Switch Applications Interface (ASAI) which allows a call to be placed in a split queue but routed only to a specific agent in that split. This allows a call to receive normal ACD call treatment (for example, announcements) and to be measured as an ACD call while ensuring that a particular agent answers.

Direct Extension Selection (DXS)
A feature on an attendant console that allows an attendant direct access to voice terminals by pressing a group select button and a DXS button.

Direct Inward Dialing (DID)
A feature that allows an incoming call from the public network (not FX or WATS) to reach a specific telephone without attendant assistance. DID calls to DID-restricted telephone lines are routed to an attendant or recorded announcement, depending on the option selected.

direct inward dialing (DID) trunk
An incoming trunk used for dialing directly from the public network into a communications system without help from the attendant.

disk drive
An electromechanical device that stores data on and retrieves data from one or more disks.

distributed communications system (DCS)
A network configuration linking two or more communications systems in such a way that selected features appear to operate as if the network were one system.

domain
Available domains are VDNs, ACD splits, and stations. The VDN domain is only used for active-notification associations, the station domain is only used for the domain-control associations. The ACD-split domain is for active-notification associations and domain-control associations.

domain-control association
A *Third Party Domain Control Request* capability initiates a unique ‘‘CRV/link number’’ combination, which is referred to as a domain-control association.

domain-controlled split
A split for which *Third Party Domain Control* request has been accepted. A domain-controlled split provides an event report for logout.

domain-controlled station
A station for which a *Third_Party_Domain_Control* request has been accepted. A domain-controlled station provides event reports for calls that are alerting, connected, or held at the station.

domain-controlled station on a call
A station active on a call which provides event reports over one or two domain-control associations.

duplicated common control
Two processors ensuring continuous operation of a communications system. While one processor is on-line, the other functions as a backup. The backup processor goes on-line periodically or when a problem condition occurs.

duplication
The use of redundant components to improve availability. When a duplicated subsystem fails, its backup redundant system automatically takes over.

duplication option
A system option that duplicates the following:

- Control carrier, which contains the switch processing element (SPE)
- Expansion interface (EI) circuit packs in carriers
- Fiber-optic cabling between port networks (PNs)
- Center-stage switch (CSS) in a CSS-connected system

E

E1
A digital transmission standard that carries traffic at the rate of 2.048 Mbps.

The E1 facility is divided into 32 channels (DS0s) of 64 kbps information numbered from 0~to 31. Channel 0 is reserved for framing and synchronization information. When a D-channel is present, it occupies channel 16.

ear and mouth (E&M) signaling
Trunk supervisory signaling, used between two communications systems, whereby signaling information is transferred through two-state voltage conditions (on the E and M leads) for analog applications and through a single bit for digital applications.

electronic tandem network (ETN)
A tandem tie trunk network that has automatic call routing capabilities based on the number dialed and the most preferred route available at the time the call is placed. Each switch in the network is assigned a unique private network office code (RNX), and each voice terminal is assigned a unique extension number.

Electronics Industries Association (EIA)
A trade association of the electronics industry that establishes electrical and functional standards.

emergency transfer
If a major system failure occurs, the automatic transfer within a communications system of a predefined set of central office (CO) lines to a group of answering telephones with at least one telephone capable of making outgoing calls. The system operates in this mode until the failure is repaired and the system automatically returns to normal operation. Also called ‘‘power-failure transfer.’’

end-to-end signaling
The transmission of touch-tone signals generated by dialing from a voice terminal user to remote computer equipment. A connection must first be established over an outgoing trunk from the calling party to the computer equipment. Then additional digits can be dialed to transmit information to be processed by the computer equipment.

enhanced private-switched communications service (EPSCS)
An analog private telecommunications network based on the No. 5 Crossbar and 1A ESS that provides advanced voice and data telecommunications services to companies with many locations.

Erlang
A unit of traffic intensity, or load, used to express the amount of traffic it takes to keep one facility busy for one hour. One Erlang is equal to 36 CCS. See also **Hundred Call Seconds**.

expansion archangel (EAA)
A network-control microprocessor located on an expansion interface (EI) port circuit pack in an expansion port network (EPN). The EA provides an interface between the EPN and its controlling switch processing element (SPE).

expansion-archangel link (EAL)
A link-access function on the D-channel (LAPD) logical link that exists between a switch processing element (SPE) and an expansion archangel (EA). The EAL carries control messages from the SPE to the EA and to port circuit packs in an expansion port network (EPN).

expansion control cabinet
See **expansion control carrier**.

expansion control carrier
A carrier in a multicarrier cabinet that contains extra port circuit packs and a maintenance interface. Also called ‘‘expansion control cabinet’’ in a single-carrier cabinet.

expansion interface (EI)
A port circuit pack in a port network (PN) that provides the interface between a PN’s time-division multiplex (TDM) bus and packet bus, and a fiber-optic link. The EI carries circuit-switched data, packet-switched data, network control, timing control, and DS1 control. In addition, an EI in an expansion port network (EPN) communicates with the master maintenance circuit pack to provide the EPN’s environmental and alarm status to the switch processing element (SPE).

expansion port network (EPN)
A port network (PN) that is connected to the TDM bus and packet bus of a processor port network (PPN). Control is achieved by indirect connection of the EPN to the PPN via a port-network link (PNL). See also **port network**.

extension-in
Extension-In (ExtIn) is the work state agents go into when they answer (receive) a non-ACD call. If the agent is in Manual-In or Auto-In and receives an extension-in call, it is recorded by CMS as an AUX-In call.

extension-out
Extension-Out (ExtOut) is the work state agents go into when they place (originate) a non-ACD call. If the agent is in Manual-In or Auto-In and places an extension-out call, it is recorded by CMS as an AUX-Out call.

external measurements
Refers to those ACD measurements that are made by the External CMS adjunct.

extension number
A 1- to 5-digit number by which calls are routed through a communications system or, with a Uniform Dial Plan (UDP) or main-satellite dialing plan, through a private network. Extension numbers are primarily used for telephones and data terminals but can also be used with specific features.

external call
A connection between a communications system user and a party on the public network or on another communications system in a private network.

F

facility
A general term used for a telecommunications transmission pathway and associated equipment.

facility associated signaling (FAS)
Signaling in which a D-channel carries the signaling only for those channels on the same physical interface.

feature
A specifically defined function or service provided by the system.

feature button
A labeled button on a telephone or attendant console used to access a specific feature.

fiber optics
A technology using materials that transmit ultrawideband electromagnetic light-frequency ranges for high-capacity carrier systems.

fixed
Fixed is a trunk allocation term. In the fixed allocation scheme, the time slots necessary to support a wideband call are contiguous, and the first time slot is constrained to certain starting points.

flexible
Flexible is a trunk allocation term. The flexible allocation scheme allows the time slots of a wideband call to occupy noncontiguous positions within a single T1 or E1 facility.

floating
Floating is a trunk allocation term. In the floating allocation scheme, the time slots necessary to support a wideband call are contiguous, but the position of the first time slot is not fixed.

foreign exchange (FX)
A central office (CO) other than the one providing local access to the public telephone network.

foreign exchange trunk
A telecommunications channel that directly connects the system to a central office (CO) other than its local CO.

foreign numbering-plan area code (FNPAC)
An area code other than the local area code. The FNPAC must be dialed to call outside the local geographical area.

G

Generic 3 Management Applications (G3-MA)
DEFINITY Communications System Generic 3 Management Applications, a PC-based tool which offers the following capabilities: emulation with DEFINITY switches and voice products, data exchange with voice products, flexible report creation, storage of switch changes to be executed later, global changes of switch forms, auditing of switch translations, automatic updates to CAS for Window Call Accounting software and bulk administration of stations.

Generic 3 Management Terminal (G3-MT)
DEFINITY Communications System Generic 3 Management Terminal, a management terminal used for system administration and maintenance in the switch room or optionally at some distance from the switch.

generalized route selection (GRS)
An enhancement to Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) that performs routing based on call attributes, such as Bearer Capability Classes (BCCs), in addition to the address and facilities restriction level (FRL), thus facilitating a Uniform Dial Plan (UDP) that is independent of the type of call being placed.

glare
The simultaneous seizure of a two-way trunk by two communications systems, resulting in a standoff.

grade of service
The number of call attempts that fail to receive service immediately. Grade of service is also expressed as the quantity of all calls that are blocked or delayed.

ground-start trunk
A trunk on which, for outgoing calls, the system transmits a request for services to a distant switching system by grounding the trunk ring lead. To receive the digits of the called number, that system grounds the trunk tip lead. When the system detects this ground, the digits are sent.

H

handshaking logic
A format used to initiate a data connection between two data module devices.

H0
An ISDN information transfer rate for 384 kbps data defined by CCITT and ANSI standards.

H11
An ISDN information transfer rate for 1536 kbps data defined by CCITT and ANSI standards.

H12
An ISDN information transfer rate for 1920 kbps data defined by CCITT and ANSI standards.

Hertz (Hz)
A unit of frequency equal to one cycle per second.

high reliability system
A system having the following: two control carriers, duplicate expansion interface (EI) circuit packs in the PPN (in G3r with CSS), and duplicate switch node clock circuit packs in the switch node (SN) carriers. See also **duplicated common control**, **duplication**, **duplication option**, and **critical reliability system**.

holding time
The total length of time in minutes and seconds that a facility is used during a call.

home numbering-plan area code
The local area code. The area code does not have to be dialed to call numbers within the local geographical area.

hop
Nondirect communication between two switch communications interfaces (SCIs) whereby the SCI message passes automatically without intermediate processing through one or more intermediate SCIs.

host computer
A computer, connected to a network, that processes data from data-entry devices.

hunt group
A group of extensions that are assigned the Station Hunting feature so that a call to a busy extension will reroute to an idle extension in the group.

I

immediate-start tie trunk
A trunk on which, after making a connection with a distant switching system for an outgoing call, the system waits a nominal 65 ms before sending the digits of the called number. This allows time for the distant system to prepare to receive digits. On an incoming call, the system has less than 65 ms to prepare to receive the digits.

information exchange
The exchange of data between users of two different systems, such as the switch and a host computer, over a local area network (LAN).

information systems network (ISN)
A wide area network (WAN) and local area network (LAN) with an open architecture combining host computers, minicomputers, word processors, storage devices, PCs, high-speed printers, and nonintelligent terminals into a single packet-switching system.

inside call
A call placed from one telephone to another within the local communications system.

Integrated Services Digital Network (ISDN)
A public or private network that provides end-to-end digital communications for all services to which users have access by a limited set of standard multipurpose user-network interfaces defined by the CCITT. Through internationally accepted standard interfaces, ISDN provides digital circuit-switched or packet-switched communications within the network and links to other ISDNs to provide national and international digital communications. See also **Integrated Services Digital Network Basic Rate Interface** and **Integrated Services Digital Network Primary Rate Interface**.

Integrated Services Digital Network Basic Rate Interface (ISDN-BRI)
The interface between a communications system and terminal that includes two 64-kbps B-channels for transmitting voice or data and one 16-kbps D-channel for transmitting associated B-channel call control and out-of-band signaling information — an arrangement called ‘‘2B+1D.’’ ISDN-BRI also includes 48-kbps for transmitting framing and D-channel contention information, for a total interface speed of 192 kbps. ISDN-BRI serves ISDN terminals and digital terminals fitted with ISDN terminal adapters. See also **Integrated Services Digital Network Primary Rate Interface**.

Integrated Services Digital Network Primary Rate Interface (ISDN-PRI)
The interface between multiple communications systems that in North America includes 24 64-kbps channels, corresponding to the North American digital signal level-1 (DS1) standard rate of 1.544 Mbytes per second.

The most common arrangement of channels in ISDN-PRI is 23 64-kbps B-channels for transmitting voice and data and one 64-kbps D-channel for transmitting associated B-channel call control and out-of-band signaling information — an arrangement called ''23B+1D,'' although with nonfacility-associated signaling (NFAS) ISDN-PRI can include 24 B-channels and no D-channel. See also **Integrated Services Digital Network** and **Integrated Services Digital Network Basic Rate Interface**.

intercept tone
An tone that indicates a dialing error or denial of the service requested.

interface
A common boundary between two systems or pieces of equipment.

internal call
A connection between two users within a system.

International Tele-communications Union (ITU)
Formerly known as International Telegraph and Telephone Consultative Committee (CCITT), ITU is an international organization that sets universal standards for data communications, including Integrated Services Digital Network (ISDN). ITU members are from telecommunications companies and organizations around the world. See also **BX.25**.

International Telegraph and Telephone Consultative Committee
See **International Telecommunications Union (ITU)**.

interflow
Allows calls to forward to other splits on the same PBX or a different PBX using the Call Forward All Calls switch feature.

intraflow
Allows calls to be redirected to other splits on the same PBX on a conditional or unconditional basis using call coverage ''busy,'' ''don't answer,'' or ''all'' criteria.

internal measurements
Refers to those BCMS measurements that are made by the system. ACD measurements that are made external to the system (via External CMS) are referred to as external measurements.

in-use lamp
A red light on a multiappearance voice terminal that is illuminated to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

ISDN Gateway (IG)
A feature allowing integration of the switch and a host-based telemarketing application via a link to a gateway adjunct. The gateway adjunct is a 3B-based product that notifies the host-based telemarketing application of call events.

ISDN trunk
A trunk administered for use with Integrated Services Digital Network primary rate interface (ISDN-PRI). Also called ''ISDN facility.''

ISDN-PRI Terminal Adapter
A terminal adapter acts as interface between endpoint applications and an ISDN PRI facility. ISDN-PRI terminal adapters are currently available from other vendors and are primarily designed for video conferencing applications. Accordingly, currently available terminal adapters adapt the two pairs of video codec data (V.35) and dialing (RS-366) ports to an ISDN PRI facility.

L

light-emitting diode (LED)
A semiconductor device that produces light when voltage is applied. LEDs provide a visual indication of the operational status of hardware components, the results of maintenance tests, and the alarm status of circuit packs, and the activation of telephone features.

lightwave transceiver
Hardware that provides an interface to fiber-optic cable from port circuit packs and digital signal level-1 (DS1) converter circuit packs. Lightwave transceivers convert electrical signals to light signals and vice versa.

line
A transmission path between a communications system or central office (CO) switching system and a voice terminal or other terminal.

line port
The hardware that provides the access point to a communications system for each circuit associated with a telephone and/or data terminal.

link
A transmitter-receiver channel that connects two systems.

link-access procedure on the D-channel (LAPD)
A link-layer protocol on the Integrated Services Digital Network basic rate interface (ISDN-BRI) and primary rate interface (ISDN-PRI) data-link layer (level 2). LAPD provides data transfer between two devices, and error and flow control on multiple logical links. LAPD is used for signaling and low-speed packet data (X.25 and mode 3) on the signaling (D-) channel and for mode-3 data communications on a bearer (B-) channel.

local area network (LAN)
A networking arrangement designed for a limited geographical area. Generally, a LAN is limited in range to a maximum of 6.2 miles and provides high-speed carrier service with low error rates. Common configurations include daisy chain, star (including circuit-switched), ring, and bus.

logical link
The communications path between a processor and a basic rate interface (BRI) terminal.

loop-start trunk
A trunk on which, after establishing a connection with a distant switching system for an outgoing call, the system waits for a signal on the loop formed by the trunk leads before sending the digits of the called number.

M

main-satellite-tributary
A private network configuration that can either stand alone or access an electronic tandem network (ETN). A ‘‘main’’ switch provides interconnection, via tie trunks, with one or more subtending switches, called ‘‘satellites’’; all attendant positions for the main/satellite configuration; and access to and from the public network. To a user outside the complex, a main/satellite configuration appears as one switch, with one listed directory number (LDN). A ‘‘tributary’’ switch is connected to the main switch via tie trunks, but which has its own attendant positions and LDN.

maintenance
The activities involved in keeping a telecommunications system in proper working condition: the detection and isolation of software and hardware faults, and automatic and manual recovery from these faults.

management terminal
The terminal that is used by the system administrator to administer the switch. The terminal may also be used to access the BCMS feature.

major alarm
An indication of a failure that has caused critical degradation of service and requires immediate attention. Major alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, logged to the alarm log, and reported to a remote maintenance facility, if applicable.

manual-in work mode
In this mode, agents automatically enter the ACW mode when they disconnect from an ACD call. However, in order to become available to receive another ACD call, they must then manually enter the Auto-In or Manual-In mode. *See* **Auto-In Work Mode** for a contrast.

memory
A device into which information can be copied and held, and from which the information can be obtained at a later time.

message center
An answering service that supplies agents to and stores messages for later retrieval.

message center agent
A member of a message center hunt group who takes and retrieves messages for voice terminal users.

minor alarm
An indication of a failure that could affect customer service. Minor alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, sent to the alarm log, and reported to a remote maintenance facility, if applicable.

modem
A device that converts digital data signals to analog signals for transmission over telephone circuits. The analog signals are converted back to the original digital data signals by another modem at the other end of the circuit.

modem pooling
A capability that provides shared conversion resources (modems and data modules) for cost-effective access to analog facilities by data terminals. When needed, modem pooling inserts a conversion resource into the path of a data call. Modem pooling serves both outgoing and incoming calls.

modular processor data module (MPDM)
A processor data module (PDM) that can be configured to provide several kinds of interfaces (RS232C, RS449, and V.35) to customer-provided data terminal equipment (DTE). See also **processor data module**.

modular trunk data module (MTDM)
A trunk data module (TDM) that can be configured to provide several kinds of interfaces (RS232C, RS449, and V.35) to customer-provided data terminal equipment (DTE).

modulator-demodulator
See **modem**.

multiappearance voice terminal
A terminal equipped with several call appearance buttons for the same extension number, allowing the user to handle more than one call, on that same extension number, at the same time.

multicarrier cabinet
A structure that holds one to five carriers. See also **single-carrier cabinet**.

multifrequency-compelled (MFC), release 2 (R2) signalling
A signal consisting of two frequency components, such that when a signal is transmitted from a switch, another signal acknowledging the transmitted signal is received by the switch. “R2” designates signaling used in the USA and countries outside the USA.

multiplexer
A device used to combine a number of individual channels into a single common bit stream for transmission.

multiplexing
A process whereby a transmission facility is divided into two or more channels, either by splitting the frequency band into a number of narrower bands or by dividing the transmission channel into successive time slots. See also **time-division multiplexing**.

multirate
Multirate refers to the new N x DS0 service (see N x DS0).

N

N x DS0
N x DS0, equivalently referred to as N~x 64~kbps, is an emerging standard for wideband calls separate from H0, H11, and H12 ISDN channels. The emerging N x DS0 ISDN multirate circuit mode bearer service will provide circuit-switched calls with data rate multiples of 64 kbps up to 1536 kbps on a T1 facility or up to 1920 kbps on an E1 facility. In the switch, N x DS0 channels will range up to 1984 kbps using NFAS E1 interfaces.

narrowband
A circuit-switched call at a data rate up to and including 64 kbps. All nonwideband switch calls are considered narrowband.

Non-Facility Associated Signaling (NFAS)
A method that allows multiple T1 and/or E1 facilities to share a single D-channel to form an Integrated Services Digital Network primary rate interface (ISDN PRI). If D-Channel Backup is not used, one facility is configured with a %D-channel, while the other facilities that share the D-channel are configured without D-channels. If D-Channel Backup is used, two facilities are configured to have D-channels (one D-channel on each facility), while the other facilities that share the D-channels are configured without D-channels.

On every facility, all DS0s that are not %D-channels are available as %B-channels. Therefore, a T1 facility without a D-channel has 24~B-channels, and an E1 facility without a %D-channel has 31~B-channels.

network
A series of points, nodes, or stations connected by communications channels.

network-specific facility (NSF)
An information element in an ISDN-PRI message that specifies which public-network service is used. NSF applies only when Call-by-Call Service Selection is used to access a public-network service.

network interface
A common boundary between two systems in an interconnected group of systems.

node
A switching or control point for a network. Nodes are either ‘‘tandem’’ (em they receive signals and pass them on (em or ‘‘terminal’’ (em they originate or terminate a transmission path.

O

offered load
The traffic that would be generated by all the requests for service occurring within a monitored interval, usually one hour.

othersplit
The Work State that indicates the agent is currently active on another split’s call, or in ACW for another split.

P

packet
A group of bits (em including a message element, which is the data, and a control information element (IE), which is the header (em used in packet switching and transmitted as a discrete unit. In each packet, the message element and control IE are arranged in a specified format. See also **packet bus** and **packet switching**.

packet bus
A wide-bandwidth bus that transmits packets.

packet switching
A data-transmission technique whereby user information is segmented and routed in discrete data envelopes called ‘‘packets,’’ each with its own appended control information, for routing, sequencing, and error checking. Packet switching allows a channel to be occupied only during the transmission of a packet; on completion of the transmission, the channel is made available for the transfer of other packets. See also **BX.25** and **packet**.

paging trunk
A telecommunications channel used to access an amplifier for loudspeaker paging.

party/extension active on call
A party is on the call if it is actually connected to the call (in active talk or in held state). An originator of a call is always a party on the call. Alerting parties, busy parties, and tones are not parties on the call.

PCOL
Personal Central Office Line.

primary extension
The main extension associated with the physical station set.

principal
A station that has its primary extension bridged on one or more other stations.

personal computer (PC)
A personally controllable microcomputer.

pickup group
A group of individuals authorized to answer any call directed to an extension number within the group.

port
A data- or voice-transmission access point on a device that is used for communicating with other devices.

port carrier
A carrier in a multicarrier cabinet or a single-carrier cabinet containing port circuit packs, power units, and service circuits. Also called a ‘‘port cabinet’’ in a single carrier cabinet.

port network (PN)
A cabinet containing a TDM bus and packet bus to which the following components are connected: port circuit packs, one or two tone-clock circuit packs, a maintenance circuit pack, service circuit packs, and (optionally) up to four expansion interface (EI) circuit packs in G3. Each PN is controlled either locally or remotely by a switch processing element (SPE). See also **expansion port network** and **processor port network**.

port-network connectivity
The interconnection of port networks (PNs), regardless of whether the configuration uses direct or switched connectivity.

Primary Rate Interface (PRI)
A standard Integrated Services Digital Network (ISDN) frame format that specifies a protocol used on digital circuits between two communications systems.

In North America, PRI runs at 1.544 Mbps and provides 23 64 kbps B-channels (voice or data) and one 64 kbps D-channel (signaling). In most countries outside of North America, PRI runs at 2.048 Mbps and provides 30 64 kbps B-channels (voice or data) and one 64 kbps D-channel (signaling.) The D-channel of the interface contains multiplexed signaling information for the other channels.

PRI endpoint (PE)
The wideband switching capability introduces PRI Endpoints on switch line-side interfaces. A PRI endpoint consists of one or more contiguous B-channels on a line-side T1 or E1 ISDN PRI facility and has an extension number. Endpoint applications have call control capabilities over PRI endpoints.

principal (user)
A person to whom a telephone is assigned and who has message center coverage.

private network
A network used exclusively for the telecommunications needs of a particular customer.

private network office code (RNX)
The first three digits of a 7-digit private network number. These codes are numbered 220 through 999, excluding any codes that have a 0 or 1 as the second digit.

processor carrier
A phrase used for ‘‘control carrier’’ in G3rV2. See also **control carrier**.

processor data module (PDM)
A device that provides an RS232C data communications equipment (DCE) interface for connecting to data terminals, applications processors (APs), and host computers and provides a digital communications protocol (DCP) interface for connection to a communications system. See also **modular processor data module**.

processor port network (PPN)
A port network (PN) controlled by a switch processing element (SPE) that is directly connected to that PN's time-division multiplex (TDM) bus and local area network (LAN) bus. See also **port network**.

processor port network (PPN) control carrier
A carrier containing the maintenance circuit pack, tone/clock circuit pack, and switch processing element (SPE) circuit packs for a processor port network (PPN) and, optionally, port circuit packs.

Property Management System (PMS)
A stand-alone computer used by lodging and health services organizations use for services such as reservations, housekeeping, and billing.

protocol
A set of conventions or rules governing the format and timing of message exchanges to control data movement and correction of errors.

public network
The network that can be openly accessed by all customers for local or long-distance calling.

pulse-code modulation (PCM)
An extension of pulse-amplitude modulation (PAM) in which carrier-signal pulses modulated by an analog signal, such as speech, are quantized and encoded to a digital, usually binary, format.

Q

quadrant
A quadrant is a group of six contiguous DS0s in fixed locations on an ISDN PRI facility. Note that this term comes from T1 terminology (one-fourth of a T1), but there are five quadrants on an E1 ISDN PRI facility (30B + D).

A quadrant is considered available or idle when all six contiguous DS0s are idle. Otherwise, the quadrant is considered contaminated or partially contaminated. This is a dynamic condition; quadrants become idle and contaminated as calls are placed and dropped. Note that a T1 facility containing the primary or backup D-channel (23B + D) has a maximum of three idle quadrants. The fourth quadrant (DS0s 19-24) never has six contiguous idle DS0s because one is always allocated to the D-channel. On an E1 facility, channel 0 is reserved for framing and synchronization, and channel 16 contains the D-channel when present, but five quadrants are potentially available.

queue
An ordered sequence of calls waiting to be processed.

queuing
The process of holding calls in order of their arrival to await connection to an attendant, to an answering group, or to an idle trunk. Calls are automatically connected in first-in, first-out sequence.

R

R2-MFC signaling
MultiFrequency compelled (MFC) signaling is a form of number signaling similar to Dual-Tone MultiFrequency (DTMF) in that tones convey the dialed number. R2-MFC is a version of MFC recommended by CCITT for signaling between a CO and a PBX over analog or digital CO, DID, or tie trunks.

random access memory (RAM)
A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

read-only memory (ROM)
A storage arrangement primarily for information retrieval applications.

recall dial tone
Tones signalling that the system has completed a function (such as holding a call) and is ready to accept dialing.

redirection criteria
The information administered for each voice terminal's coverage path that determines when an incoming call is redirected to coverage.

redirection on no answer
An optional feature that redirects an unanswered ringing ACD call after an administered number of rings. The call is then redirected back to the agent.

remote home numbering-plan area code (RHNPA)
A foreign numbering-plan area code that is treated as a home area code by the Automatic Route Selection (ARS) feature. Calls can be allowed or denied based on the area code and the dialed central office (CO) code rather than just the area code. If the call is allowed, the ARS pattern used for the call is determined by these six digits.

reorder tone
A tone to signal that at least one of the facilities, such as a trunk or a digit transmitter, needed for the call was not available at the time the call was placed.

report scheduler
Software that is used in conjunction with the system printer for the purpose of scheduling the days of the week and time of day that the desired reports are to be printed.

RS232C
A physical interface specified by the EIA. RS232C transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet.

ROSE
Remote Operations Service Element is a CCITT and ISO standard that defines a notation and services that support interactions between the various entities that make up a distributed application.

S

sanity and control interface (SAKI)
A custom, very-large-scale-integration (VLSI) microchip located on each port circuit pack. The SAKI provides address recognition, buffering, and synchronization between the angel and the five control time slots that make up the control channel. The SAKI also scans and collects status information for the angel on its port circuit pack and, when polled, transmits this information to the archangel.

simplex system
A system that has no redundant hardware.

simulated bridged appearance
The same as a **temporary bridged appearance**, allows the station user (usually the principal) the ability to bridge onto a call which had been answered by another party on its behalf.

single-carrier cabinet
A combined cabinet and carrier unit that contains one carrier. See also **multicarrier cabinet**.

single-line voice terminal
A voice terminal served by a single-line tip and ring circuit (models 500, 2500, 7101A, 7103A).

small computer system interface (SCSI)
An ANSI bus standard that provides a high-level command interface between host computers and peripheral devices.

software
A set of computer programs that perform one or more tasks.

split
A condition whereby a caller is temporarily separated from a connection with an attendant. A split condition automatically occurs when the attendant, active on a call, presses the start button.

split number
The split's identity to the switch and BCMS.

split report
Provides historical traffic information for internally measured splits.

split (agent) status report
Provides the real-time status and measurement data for internally measured agents and the split to which they are assigned.

staffed
Indicates an agent position is logged-in. A staffed agent will be functioning in one of four work modes: Auto-In, Manual-In, ACW, or AUX-work.

Station Message Detail Recording (SMDR)
An obsolete term now called "CDR" (see call detail recording), which is a switch feature that utilizes software and hardware to record call data.

standard serial interface (SSI)
A communications protocol developed by AT&T Teletype Corporation for use with the 500 business communications terminals (BCTs) and the 400-series printers.

status lamp
A green light that shows the status of a call appearance or a feature button by the state of the light (lit, flashing, fluttering, broken flutter, or unlit).

stroke counts
A method used by Automatic Call Distribution (ACD) agents to record up to nine customer-defined events per call when the Call Management System (CMS) is active.

switch
Any kind of telephone switching system. See also **communications system**.

switchhook
The buttons located under the receiver on a voice terminal.

switch node (SN) carrier
A carrier containing a single switch node, power units, and, optionally, one or two digital signal level-1 (DS1) converter circuit packs. An SN carrier is located in a center stage switch (CSS).

switch node (SN) clock
The circuit pack in a switch node (SN) carrier that provides clock and maintenance alarm functions and environmental monitors for an SN.

switch node interface (SNI)
The basic building block of a switch node. An SNI circuit pack controls the routing of circuit, packet, and control messages.

switch node link (SNL)
The hardware that provides a bridge between two or more switch nodes. The SNL consists of the two switch node interface (SNI) circuit packs residing on the switch nodes and the hardware connecting the SNIs. This hardware can include lightwave transceivers that convert the SNI's electrical signals to light signals, the copper wire that connects the SNIs to the lightwave transceivers, a full-duplex fiber-optic cable, digital signal level-1 (DS1) converter circuit cards and DS1 facilities if a company does not have rights to lay cable, and appropriate connectors.

switch processing element (SPE)
A complex of circuit packs (em processor, memory, disk controller, and bus-interface cards (em mounted in a processor-port-network (PPN) control carrier. The SPE serves as the control element for that PPN and, optionally, for one or more expansion port networks (EPNs).

synchronous data transmission
A method of sending data in which discrete signal elements are sent at a fixed and continuous rate and specified times.

system administrator
The person who maintains overall customer responsibility for system administration. Generally, all administration functions are performed from the G3 Management Terminal (G3-MT). The switch requires a special login, referred to as the system administrator login, in order to gain access to the system administration capabilities.

system printer
An optional printer that may be used to print scheduled reports via the report scheduler.

system report
Provides historical traffic information for all internally measured splits.

system status report
Provide real-time status information for internally measured splits.

system manager
A person responsible for specifying and administering features and services for a system.

system reload
A process that allows stored data to be written from a tape into the system memory (normally after a power outage).

T

T1
A digital transmission standard that in North America carries traffic at the digital signal level-1 (DS1) rate of 1.544 Mbps. A T1 facility is divided into 24 channels (DS0s) of 64 kbps information numbered from 1 to 24. These 24 channels, with an overall digital rate of 1.536 Mbps, and an 8 kbps framing and synchronization channel make up the 1.544 Mbps transmission. When a D-channel is present, it occupies channel 24.

T1 facilities are also used in Japan and some Middle-Eastern countries.

TAC
Trunk Access Code.

tandem switch
A switch within an electronic tandem network (ETN) that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.

tandem through
The switched connection of an incoming trunk to an outgoing trunk without human intervention.

tandem tie-trunk network
A private network that interconnects several customer switching systems by dial-

TEG
Terminating Extension Group.

terminal
A device that sends and receives data within a system. See also **administration terminal**.

tie trunk
A telecommunications channel that directly connects two private switching systems.

time-division multiplex (TDM) bus
A bus that is time-shared regularly by preallocating short time slots to each transmitter. In a PBX, all port circuits are connected to the TDM bus, permitting any port to send a signal to any other port.

time-division multiplexing (TDM)
Multiplexing that divides a transmission channel into successive time slots. See also **multiplexing**.

time interval
The period of time, either one hour or one-half hour, that BCMS measurements are collected for a report(s).

time slice
See **time interval**.

time slot
A time slot refers to 64 kbps of digital information structured as eight bits every 125 micro-seconds. In the switch, a time slot refers to either a DS0 on a T1 or E1 facility or a 64 kbps unit on the TDM bus or fiber connection between port networks.

time slot sequence integrity
Time slot sequence integrity means that the "N" octets of a wideband call that are transmitted in one T1 or E1 frame arrive at the output in the same order that they were introduced.

to control
To control means that an application can invoke Third Party Call Control capabilities using either an adjunct-control or a domain-control association.

to monitor
To monitor means that an application can receive *Event_Reports* on either an active-notification, adjunct-control, or a domain-control association.

tone ringer
A device with a speaker, used in electronic voice terminals to alert the user.

trunk
A dedicated telecommunications channel between two communications systems or central offices (COs).

trunk allocation
The manner in which trunks are selected to form wideband channels.

trunk data module
A device that provides the interface for connection between off-premises private-line trunk facilities and a G3V2 switch. The trunk data module provides conversion between the RS232C and the Digital Communications Protocol (DCP), and can connect to direct distance dialing (DDD) modems as the DCP member of a modem pool.

trunk group
Telecommunications channels assigned as a group for certain functions that can be used interchangeably between two communications systems or central offices (COs).

U

uniform dial plan
A feature that allows a unique 4- or 5-digit number assignment for each terminal in a multiswitch configuration such as a distributed communications system (DCS) or main-satellite-tributary system.

V

vector directory number (VDN)
An extension that provides access to the Vectoring feature on the switch. Vectoring allows a customer to specify the treatment of incoming calls based on the dialed number.

vector-controlled split

A hunt group or ACD split administered with the ‘‘vector’’ field enabled. Access to such split is only possible by dialing a VDN extension. Vector-Controlled Splits cannot be Active Notification Domains.

voice terminal

A single-line or multiappearance telephone.

W

wide area tele-communications service (WATS)

A service in the USA that allows calls to a certain area or areas for a flat-rate charge based on expected usage.

wideband

A circuit-switched call at a data rate greater than 64 kbps. A circuit-switched call on a single T1 or E1 facility with a bandwidth between 128 and 1536 (T1) or 1984 (E1) kbps in multiples of 64 kbps. H0, H11, H12, and N x DS0 calls are all wideband.

wideband access endpoint

The wideband switching capability extends Access Endpoints to include wideband access endpoints. A wideband access endpoint consists of one or more contiguous DS0s on a line-side T1 or E1 facility and has an extension number. The Administered Connections feature provides call control for calls originating from wideband access endpoints.

wink-start tie trunk

A trunk with which, after making a connection with a distant switching system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

work modes (or ACD work modes)

A work mode is one of four states (Auto-In, Manual-In, ACW, AUX-work) that an ACD agent enters after logging in. Immediately upon logging in, an agent enters the AUX-work mode. To become available to receive ACD calls, the agent enters either the Auto-In or Manual-In work modes. To do work associated with an ACD call, at the conclusion of the call, an agent would enter the ACW mode. If an agent changes work modes while handling a call, the change becomes effective when the agent finishes the call. The system does not recognize the change until the call is completed.

In order to answer an ACD call, the ACD agent must specify a Work Mode. Generally, two methods are available for indicating Work Modes: (1) by pressing the appropriate button on their voice terminal, and (2) by dialing an access code. The four work modes associated with ACD call handling are Auto-In, Manual-In, ACW, and AUX-work. An agent can change work modes while handling a call, but the system will not recognize the change until the call is completed. It is important that the ACD agents always accurately indicate their correct work mode, otherwise the BCMS measurements will not be accurate.

work state

An ACD agent may be a member of up to three different splits. Each ACD agent continuously exhibits a work state for every split that it is a member of. Valid work states are Avail, Unstaffed, AUX-work, ACW, ACD (answering an ACD call), ExtIn, ExtOut, and OtherSpl. An agent’s work state for a particular split may change for a variety of reasons (for example, whenever a call is answered, abandoned, the agent changes work modes, etc.). The BCMS feature monitors the work states and uses this information to provide the BCMS reports.

write operation

The process of putting information onto a storage medium, such as a hard disk.

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