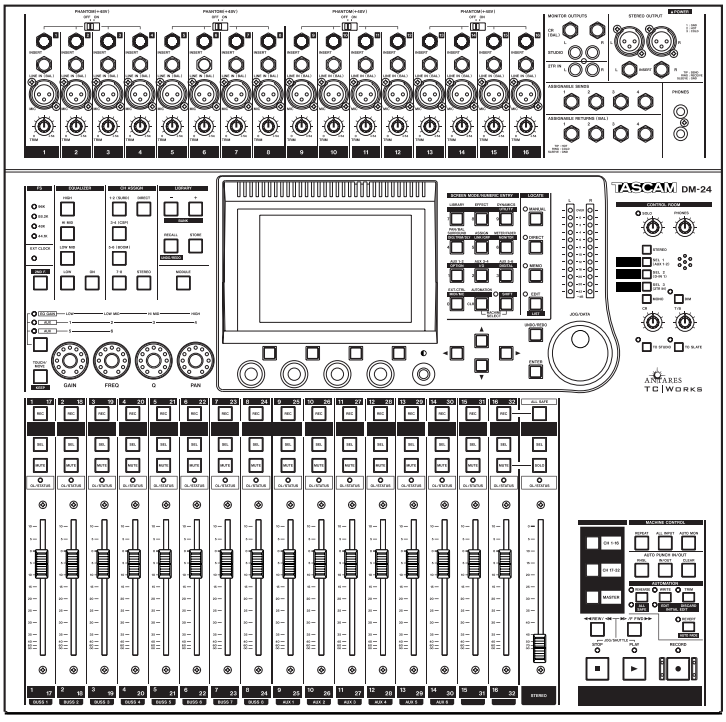
TASCAM

TEAC Professional Division

DM-24

Digital Mixing Console



EFFECTS MANUAL

Table of Contents

The DM-24 contains a number of high-quality effects that you can use within your project, either while recording, or on mixdown.

The effects available include:

- Microphone modelling (single-channel)
- Speaker modelling (two channels)
- Chorus (two channels)
- Delay (two channels)
- Distortion (single-channel)
- Guitar compression (single-channel)
- Soft compression (two channel)
- Phaser (two channels)
- Pitch shifter (two channels)
- Flanger (two channels)
- De-esser (two channels)
- Exciter (two channels)
- Reverb (two channels)

Out of these, the two effects are available at any one time in 44.1k or 48k sampling frequency mode. In high sampling frequency modes (88.2k or 96k), only one effect is available.

NOTE

The following combinations cannot be used: reverb + reverb, reverb + speaker modeler.

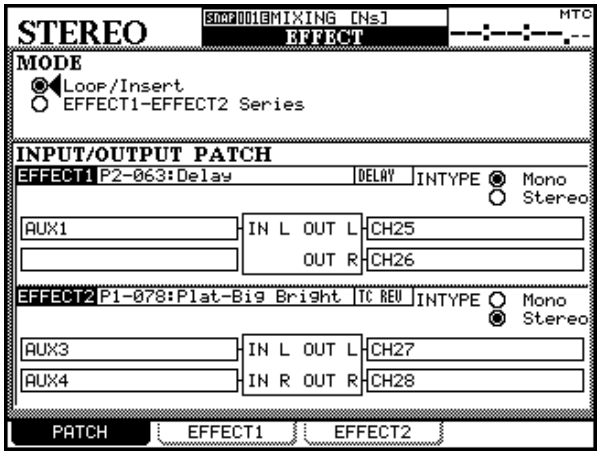
In high sampling frequency mode, the reverb, microphone modeler and speaker modeler are unavailable.

Patching and setting up effects

All effect settings are managed using the **EFFECT** key. This allows the assignment of sends and returns, as well as the selection and parameter setting for the effects.

There are two primary options, to use the effects independently in loop or insert mode, or to use the two effects in series, with the output of effect 1 feeding the input of effect 2 (similar to some multi-effect units).

The first of these modes is known as the Loop/Insert mode, and the second as the EFFECT1 EFFECT2 Series mode.



Press the **EFFECT** key followed by soft key 1 (PATCH) to bring up the patch screen as shown here.

NOTE

*When using the DM-24 in high sampling frequency mode, only one effect is available, and only one effect (*EFFECT 1*) is shown on this screen.*

Use the cursor keys to select the mode (either Loop/Insert or EFFECT1 EFFECT2 Series), and the **ENTER** key to confirm the choice.

The lower part of the screen contains a representation of the two internal effect inputs and outputs. An effect may have two inputs (L and R) and two outputs (L and R). See “Mono and stereo inputs” on page 4 for more information.

However, this does not mean that there are two separate effect processors in each effect. It is possible to use the two inputs of the effect processor “creatively” (that is, have two completely separate feeds for the left and the right inputs of the effect), but this is not recommended.

We strongly suggest that only pairs of inputs (e.g. stereo inserts, odd-even pair buss and aux inserts and odd/even pairs of aux sends) are selected as stereo inputs for the effects.

Use this screen to select the input sources for the internal effects.

The choices available are:

Effect source	Display shows
Aux sends 1 through 6	AUXx
Buss 1 through 8 insert	BUSS1 INS SEND
Aux 1 through 6 insert	AUXx INS SEND
Stereo L, R insert	ST-L PRESEND, STR-R PRE SEND
Assignable insert 1 through 4	ASGN INSx SEND

Use the cursor keys, dial and **ENTER** key to set the value for each input.

WARNING

Although it is theoretically possible to select both an aux send and an aux insert as input sources for an effect, a few seconds' thought will show that this will result in a feedback loop, resulting in possible damage to equipment (and ears!). You should therefore avoid making this type of setting.

A popup message appears to show that the assignment has been made.

NOTE

The same source cannot be selected twice to feed two different effect inputs (except for the aux sends). A popup message appears to warn of attempted duplicate assignments.

Any send/return assignments made to the effects will override any assignments made to external send/return insert loops.

The effect output destination cannot be selected here—the destination of the effect outputs is determined by the choice of the input source, and in the case of the aux sends by the settings made in the I/O screens.

Mono and stereo inputs

The DM-24 internal effects are either single-channel or two-channel, as listed at the start of this section.

At the top of the input/output patch section for each effect, there is a field called INTYPE (input type).

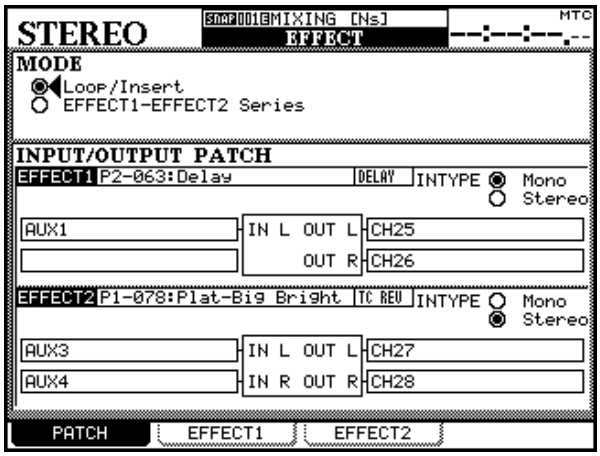
In the case of single-channel effects, the only option available is Mono.

In the case of dual-channel effects, there is a pair of radio buttons: Stereo and Mono. Select one of these as appropriate, depending on whether one mono source, or a stereo pair of sources (e.g. a pair of aux sends) will be used to feed the effect.

Once again, we recommend that only pairs of inputs (e.g. stereo inserts, odd-even pair buss and aux inserts and odd/even pairs of aux sends) are selected as stereo inputs for the effects.

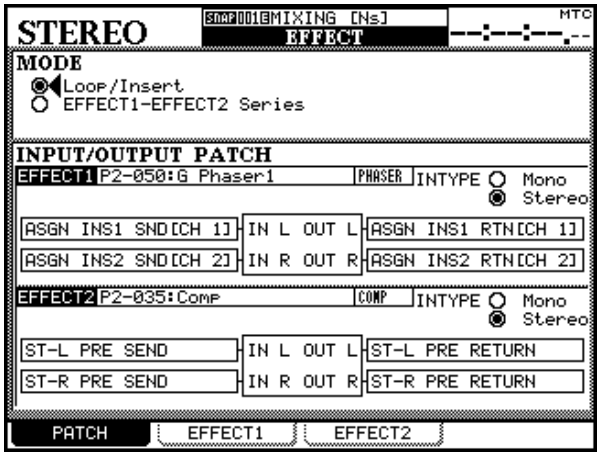
The number of outputs available for an effect depends on a number of factors: the type of effect currently selected, the mono/stereo input type currently selected, and the destination of the effect (for instance, if effect 1 is patched in series with a single-channel effect used in effect 2, only one channel is output from effect 1).

Example 1 (Loop/insert setting with 1=mono input and 2=stereo input In this example, the delay line is fed by a mono signal source (for example a microphone) and the output is spread between the left and right outputs.



The stereo inputs to the plate reverb maintain the image of the stereo source (for example, if a pair of overhead mics has been set up to record a drum kit).

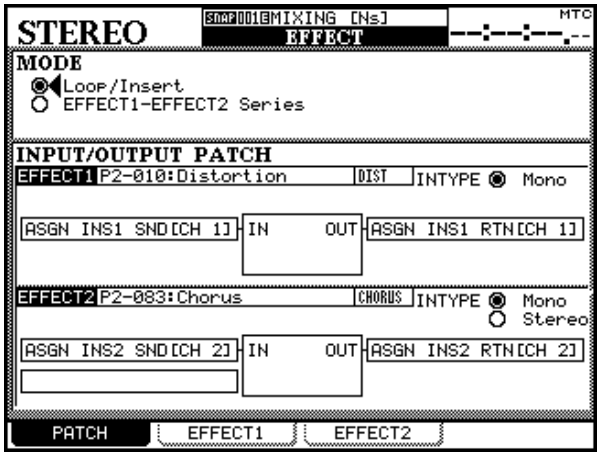
Example 2 (Loop/insert setting with 1 & 2 both = stereo input) In this example, both effects are used in insert mode. Busses 1 and 2 use a effect 1 as a stereo phaser (this can be turned on or off as needed for a creative effect).



The stereo compressor assigned to effect 2 is inserted in the stereo output buss in order to limit the dynamic range of the stereo outputs.

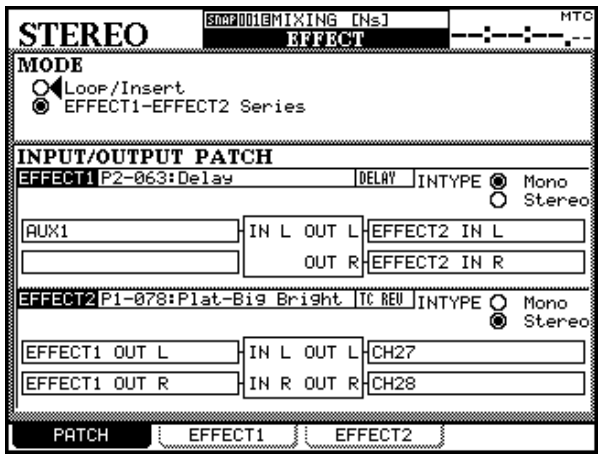
Example 3: (Loop/insert setting with 1 & 2 both = mono input) Here again, both effect 1 and effect 2 are used as inserts, but they both have mono inputs.

Effect 1 (a distortion setting) is being used with a distortion effect, in order to achieve a distorted vocal sound.



Effect 2 (a chorus) is inserted into an input channel (for example, a fretless bass) in order to thicken the sound. In this example, because these effects are being used by only one channel each, there is no need to tie up the aux sends and returns, which can then be used for other purposes.

Example 4: 1/2 series (1=mono input, 2=stereo input) In this example, the two effects are put in series, with effect 1 (echo) taking a mono mic signal and echoing it to the left and right channels.



These echoes are then passed to the reverb, where they are processed “in-place” to provide an interesting stereo effect (note that reversing these two effects would produce echoed reverb—probably less desirable).

Effect send sources

Whether the effect (or in series mode, both effects together) is used as a loop or an insert depends on the source selected for the effect inputs.

Aux 1 through 6 When these are selected as effect input sources, the effect is placed in a loop.

The effect output with this setting is assigned to a channel using the I/O screens (see “Signal sources” on page 36 of the main manual).

If a channel has already been assigned to take its input from an internal effect, this channel is shown

on the OUT section of the screen when the loop assignment is made and is shown as Chxx (xx = 1 through 32). If no channel has been assigned, the display shows ---. If more than one channel has been assigned as a return, the display shows ****.

Aux 1 through Aux 6 insert When these are selected as effect input sources (AUXx INS SEND), the effect becomes an insert-type effect.

This insert is made post aux send fader.

The outputs of the effect are automatically assigned to the appropriate aux insert returns and shown as AUX INS RETURN.

Buss 1 through buss 8 insert When these are selected as effect input sources (BUSSx INS SEND), the effect becomes an insert-type effect.

This insert is made post buss level fader.

The outputs of the effect are automatically assigned to the appropriate buss insert returns and shown as BUSS INS RETURN.

Stereo L, R insert When these are selected as effect input sources (ST-L PRE SEND and ST-R PRE SEND), the effect becomes an insert-type effect.

This insert is made pre stereo master fader.

The outputs are automatically assigned to the stereo insert returns are shown as ST-L PRE RETURN and ST-R PRE RETURN.

Assignable inserts 1 through 4 When these are selected as effect input sources (ASGN INSx SEND), the effect becomes an insert-type effect.

For these to be effective, the assignable send/returns must be set to be inserts, not send/return loops (see “Assignable sends and returns” on page 43 of the main manual). If they have been set to send/return loops, a popup message appears informing you of the fact.

Note that when an assignment is made to these inserts, the corresponding physical 1/4” jacks are no longer available (these settings override the physical jack insert assignments).

The outputs from the effects are sent to the assignable insert return. This is shown in the output assignment section of the effect as ASGN INSx RTN CH y if a channel assignment has been made, or ASGN INSx RTN --- if no assignment has been made.

Effect 1-2 series

When the two effects units are selected to act in series, with effect 1 feeding effect 2, although both effect 1 and effect 2 are shown on the screen, only the inputs to effect 1 may be set.

The output(s) from effect 1 are automatically routed to the input(s) of effect 2.

If the source of effect 1 is an aux send, the effect 2 output is assigned to a channel (set using the I/O screen).

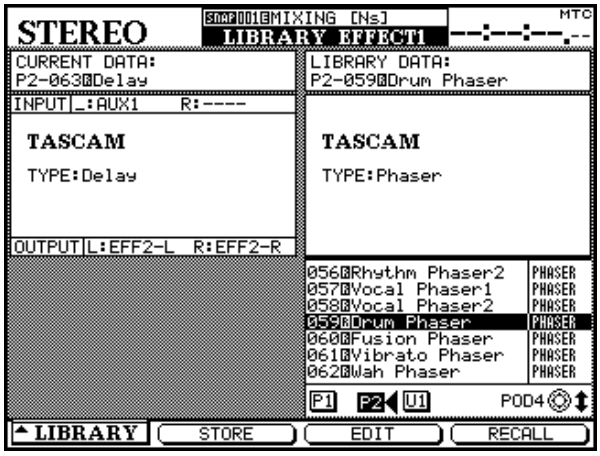
If the source of effect 1 is an insert, the output of effect 2 defaults to the insert return as shown in the output assignment section of effect 2.

The outputs from effect 1 are shown as EFFECT2 IN L and EFFECT2 IN R, and the inputs to effect 2 are shown as EFFECT1 OUT L and EFFECT1 OUT R (if effect 2 is a dual-channel effect).

If channel 2 is set to output only a single channel, the single output from effect 1 is labeled as EFFECT2 IN L and if there is a single input to effect 2, this is labeled EFFECT1 OUT L.

Setting up the effects units

To use one of the internal effect units, press the **EFFECT** key and then soft key 2 or soft key 3 (EFFECT 1 or EFFECT 2).



Next, use the EFF... LIB-> key (soft key 4) to enter the effects library.

Use the cursor keys and **ENTER** key to select either one of the two effect preset banks or the user effect bank.

There are two preset library banks:

- The first bank holds the TC Works reverb preset settings, as well as blank templates for the Antares microphone and speaker modelers (“Preset reverb settings” on page 22).
- The second holds preset TASCAM effect settings (“Preset effect settings” on page 27).

Use the library screen to scroll through the list of different library entries in the selected bank.

See “Library functions” on page 99 of the main manual for full details regarding library functions.

NOTE

Although the preset library banks are named 1 and 2, an effect from library bank 1 may be recalled for use with effect 2, etc., as well as the other way round. Remember, though, that when using effects in series, effect 1 always feeds effect 2. If the effect settings are recalled to the wrong effect, the sound may not be quite what you expect (echoed reverb is rather different from reverbed echo, for example).

When you recall a library entry from the library, a popup message appears confirming the selection.

When the **EFFECT** key is pressed, the effect screen showing the values and parameters appropriate for that particular type of effect is displayed.

Once an entry has been recalled, there is no way of changing its type through the on-screen parameters. You must reload an entry of another type from the library in order to change the effect type.

NOTE

The points at which the effects are returned are set in the I/O screens and are selectable in the same way as for mic/line inputs, etc.

Changing parameters

The parameters of the entry are changed using the cursor keys and PODs, dial and **ENTER** key, in the same way as other parameters on the DM-24.

These parameter settings take place immediately (that is, the effect of the change can be heard immediately after the parameter has been changed).

See the appropriate sections of this manual for details of how the parameters change depending on the effect type selected.

Essentially, there are two different types of effect: the in-line type of processor, typically used in an insert mode, and the send/return type, typically used in a loop mode (aux send to channel return).

There are no hard and fast rules as to how these should be used, though. If you wish to use the guitar amplifier simulator to add an unusual sound to a string quartet, you are of course free to do so!

Note how all effect screens have a pair of input meters and a pair of output meters at the top left of the screen so that the level can be properly adjusted.

The microphone modeler has (in the top row of PODs) an input and an output level control.

The speaker modeler has (in the top row of PODs) an input control.

The reverb and the other (TASCAM) effects all have (in the top row of PODs) an input and output level control, as well as a mix (wet/dry) control.

NOTE

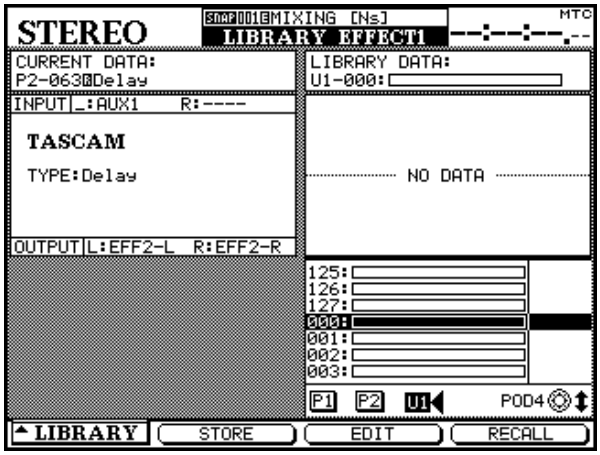
On account of unavoidable processing delays, it is recommended that the mix control be always set to 100% (fully wet), as the original and processed sound may be a few samples out of phase with each other, resulting in audio artifacts if the two signals are mixed.

Storing your settings

When you have set up the parameters of an effect, you can store it for further use in the user effects library.

This saves you having to make the same settings every time for a commonly-used microphone model, for example.

While in the effect parameters screen, press soft key 4 (the EFF...LIB key) to bring up the library screen. This allows you to scroll through the list of settings and either save to an unused library entry in the user effect bank, or to overwrite an existing setting stored in the library.



Again, consult the main manual for details of how to name and manage library entries.

Because of the nature of the DM-24's routing, a little thought may be needed when assignments are made on the effect patch page, as the DM-24 allows the same signal to be routed to more than one channel simultaneously.

Although this kind of versatility is often desirable, it is important to make sure that this kind of assignment does not happen accidentally, causing unexpected and unwanted results.

This section provides some tips and pointers on how to best set up the DM-24 in order to avoid any such possible problems.

Default snapshot settings

The default mix snapshot returns the outputs from effect 1 and effect 2 to channels 25/26 and 27/28 respectively.

In the same snapshot, assignable returns are assigned to channels 29 through 32.

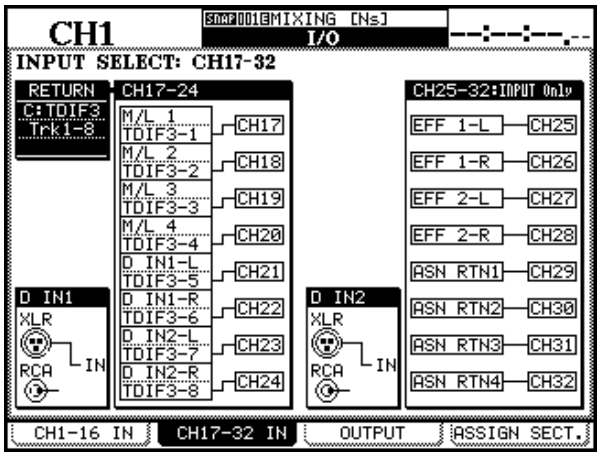
These settings are designed for the use of the internal sends with the aux sends and returns, and external effects with the hardware insert loops (assignable sends and returns). If you are making changes to use the internal effects or assignable sends and returns as inserts or assignable inserts, a little work must be done, using the assignment screens.

Using the internal effects as inserts (i)

In this example, effect 1 will be used as an insert on buss 2.

To do this, the effects returns must first be removed (de-assigned) from channels 25 and 26.

With the **SHIFT** indicator lit, press the **I/O** key until the screen allowing assignment of channels 17 through 32 appears (or use soft key 2 to access the screen).

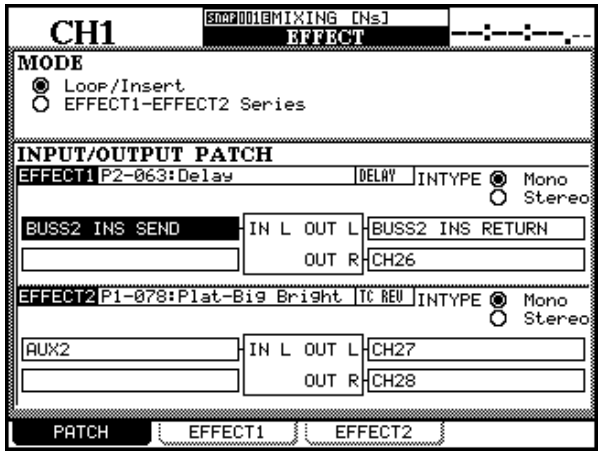


You should set these inputs to some "harmless" setting which will not conflict with any other setting that is already in use. A useful source here might be one of the digital inputs (if you are not already using them).

NOTE

It is possible to assign the same source to more than one channel. There are obvious dangers associated with such an action, so we do not recommend that you do this.

Next, return to the effect patch page (press the **EFFECT** key until the patch screen appears):



Select BUSS2 INS SEND as the input source for effect 1. When you do this, the output for effect 1 will automatically change (BUSS INS RETURN).

If you had not removed the effect returns from channels 25 and 26 before assigning the buss insert send, the effect return would have been routed to these channels as well as to the buss insert return (as in the screen shot above).

Using the internal effects as inserts (ii)

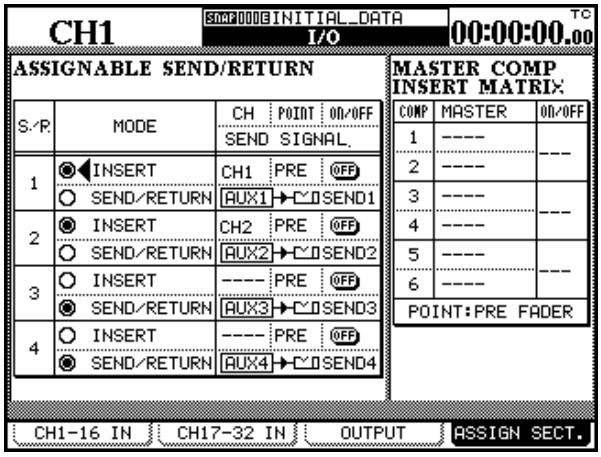
In this section, we look at how you change the default settings to use effect 2 as a stereo input processor using assignable send/return inserts 1 and 2. The insert will be assigned with channels 1 and 2.

Again, use the **I/O** key (the **SHIFT** indicator must be lit) to access the 17 through 32 screen (soft key 2), allowing the de-assignment of the assignable returns 1 and 2 from channels 29 and 30.

Again, pick a “safe” or harmless option of an input that you are not using.

Next, the assignable sends and returns have to be changed from their send/return loop setting to an insert setting.

Press soft key 4 to access the assignable output screen:



Assignable send/returns 1 and 2 should be set to the insert mode.

The insert channels (that is, the channels on which the inserts will work) should be set to channels 1 and 2 (of course, you are free to change this if you want to use other channels with this effect.

In the **EFFECT** patch page, select Stereo as the input type for effect 2.

Change the input source to ASGN INS 1 SEND (CH 1) for the left input and ASGN INS 2 SEND (CH 2) for the right input.

NOTE

It is important that these operations are carried out in the order here.

If you try to route these assignable inserts to effect 2 (on the effect patch screen) without changing the assignable send/return mode first, a popup message appears telling you that the assignable insert is in the send/return mode.

If you then try to correct this by changing the send/return mode to the insert mode, another message appears, informing you that return 1 is currently assigned to channel 29.

NOTE

All names of microphone manufacturers and microphone model designations appearing in this manual and on the DM-24 are used solely to identify the microphones analyzed in the development of the digital models and do not in any way imply any association with or endorsement by any of the named manufacturers.

This effect allows you to model the characteristics of a particular model of microphone and apply it to the microphone you are actually using.

In addition to reproducing the sonic characteristics of the modeled microphones, this effect also allows for the reproduction of certain options on the modeled microphone (for example, low cut filters, etc.).

Typically, you will want to "re-record" already recorded tracks with another microphone model at the mixdown stage, as this allows you to experiment with settings.

When you use the modeler at the mixdown stage, though, it is important that you have clear and detailed notes of the microphone conditions which were used to make the original recording.

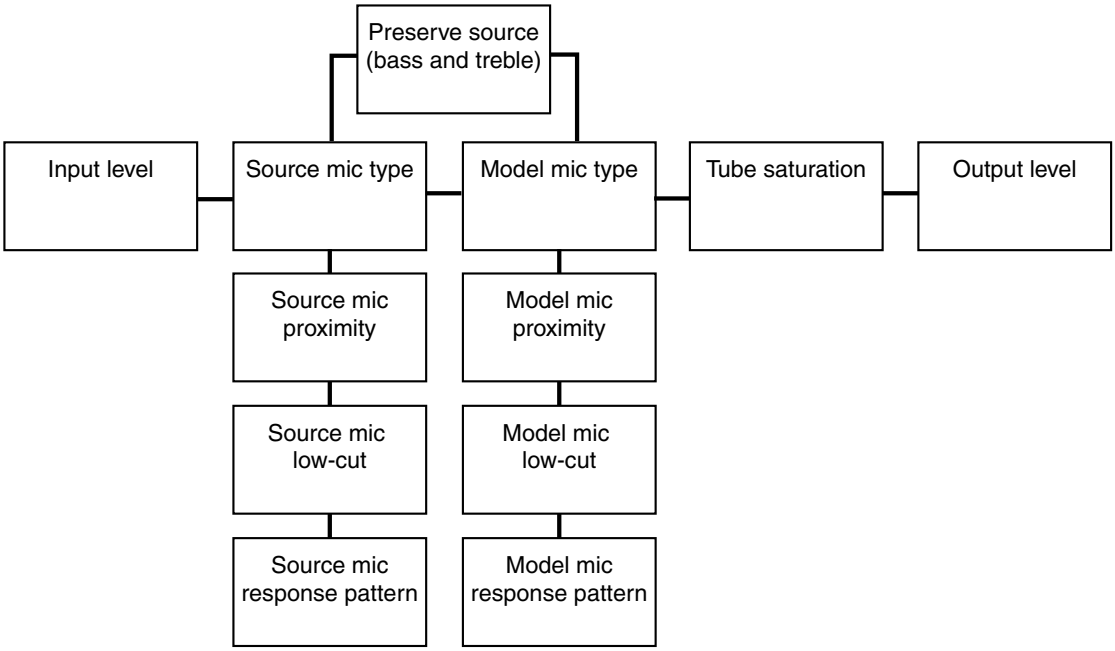
Among other useful information which should be noted when the recording is made:

- Type of microphone
- Distance of source from microphone
- Any filter settings made on the microphone
- The response pattern used when recording

Of course, it is also possible to record directly using one physical microphone and modelling another, but in this case, it is more difficult to experiment, and to make changes afterwards.

Note that when we talk about the microphone modeler, we use the term *source* microphone to describe the actual physical device and the description of it in the modeler, and *model* to describe the target, virtual microphone.

The diagram below gives an approximate idea of how the parameters available interact with each other (signal flow may be taken as being from left to right).



Limitations

Be aware, though, that while the microphone modelling will produce excellent effects, it is not capable of producing something from nothing. In other words, a poor recording made with a cheaper source microphone will not be magically transformed into a good recording, if an expensive microphone model is selected—it will still sound like a poor recording, but made with an expensive model.

Nor can the microphone modeler magically restore missing parts of the signal which are missing because of the limitations of the source microphone. If a cheap microphone with limited bass response is used to record, using an expensive model with the microphone modeler will not put the missing bass back into the recording.

Excessive frequency boosting can occur if processes intervening between the microphone and the modeler produce noise. This noise will be excessively boosted, especially if the filtering on the microphone and the recording process has accentuated this.

Polar response patterns can be simulated, but cannot automatically change the pattern of the source microphone. For example, if a recording has been made using a microphone with a cardioid response pattern, setting the model's pattern to omnidirectional will not automatically turn the source microphone into an omnidirectional microphone (and add the room ambience that would be present if the microphone actually was an omnidirectional one).

Likewise, if a source microphone has a particular off-axis response, this individuality will be retained even if a different model is selected.

NOTE

The microphone modeler can only be used with the L input and output of either effect 1 or effect 2. It is not possible to use the microphone modeler to process two channels at the same time using one effect.

The microphone modeler is not available in high sampling frequency mode.

Selecting the microphone modeler

Recall the preset library entry 1-100 in order to load the microphone modeler.

See "Setting up the effects units" on page 7 for further details.

Overall settings

These settings apply to the overall effect (not to the source or model microphones individually).

Input gain This (INPUT) allows you to set the relative gain for the input source (top row, POD 2).

Start at 0dB, but you may want to increase the level slightly to increase the amount of saturation available to the processor. The signal may be cut by a value up to –30 dB and boosted by up to 12 dB.

NOTE

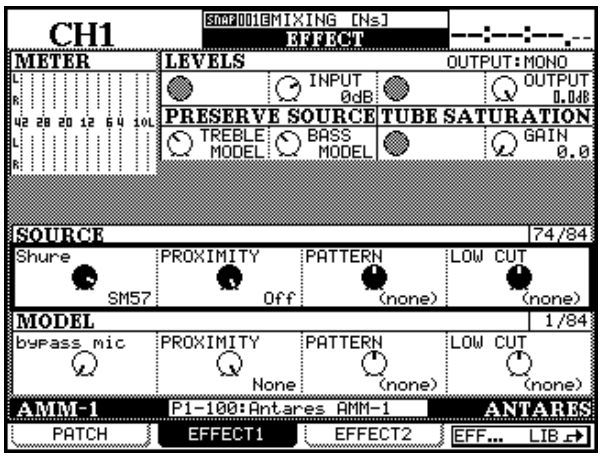
Increasing this input level to obtain the highest possible non-clipping meter level does not result in the improvement in dynamic range that would result if this operation was to take place on an all-analog system.

Output level This (OUTPUT) allows the overall output gain from the modeler to be adjusted from 0 dB to –12 dB.

Bypass This allows the whole of the microphone modeler to be bypassed for A-B comparisons. It is not the same as selecting the bypass microphone model ("The bypass microphone model" on page 13), which is a "neutral" microphone model for either source or output (but it is the same as selecting it for source and output).

Selecting the source microphone

Move the cursor to the Source Microphone, selecting the model using POD 1.



The manufacturer name is given at the top left of the box, and the model at the bottom right.

There may be two listings for a particular source microphone model, one of them ending with a -W. This means that this is the model of microphone with a supplied windscreen (thereby affecting the acoustic characteristics of the microphone).

There may also be a (m1) or (m2) following the microphone name. These refer to different examples of the same kind of microphone. Pick the one which is most appropriate for your particular microphone.

If you do not have a microphone listed in the list of source microphones provided:

- Use a different microphone which is listed, if you have one.
- Select a similar model of microphone from the same manufacturer; that is, one with similar characteristics to the one in use.
- Select another microphone of the same type (for example, another large condenser microphone, etc.).
- Select Bypass (that is, no microphone) as the source.

Note that if you do select a microphone of a different type to the actual microphone, though you will probably obtain acceptable results, the resulting sound will not be 100% accurate.

The bypass microphone model

The bypass microphone model is equivalent to no microphone being used. This may be useful in the case of electric instruments which have been direct-injected (that is, with no microphone involved) and where the model microphone is to be used to provide a distinctive sound for these instruments.

Although this may not produce an absolutely realistic model of the model microphone, it will almost certainly produce an interesting sound.

Source microphone settings

In addition to the type of microphone used as the source microphone, the modeler needs to know a few more things before it can achieve the best results:

Proximity This is the average distance of the sound source from the microphone when the recording is made. The distance is measured in inches (1 inch = 2.54cm). If this is not set, then the "proximity effect" (an artificial boost in bass frequencies at close range) may not be properly compensated, and the sound will be unnatural. Note that microphones with an omnidirectional response do not exhibit this proximity effect, and any settings made here with an omni source microphone will have no effect.

Use pod 2 on row 3 for this setting.

NOTE

As the source moves away from the microphone, an amount of ambient room tone is added to the recording. The microphone modeler cannot add the room tone, but a little reverb added to the signal may help here.

Pattern The pattern of the source mic, if selectable, should be echoed in this setting. If the source mic is fixed-pattern, no selection is possible here, and the display shows None here. Use POD 3 on row 3.

Low-cut filter Many microphones have a bass cut filter. If this filter has been set on the real physical source microphone, this setting should be made on the source microphone of the modeler as well.

This is done using POD 4 on row 3.

The actual name of this filter varies according to the actual name on the physical microphone, and will not exist at all if the mic does not actually have such a filter fitted (the display shows None).

NOTE

The modeler assumes that the source was recorded on-axis. Since there is no way to tell the modeler about the actual position of the source relative to the microphone, the modeler cannot compensate for frequency differences, etc. caused by off-axis placement of the source.

Selecting a model for output

In the same way as you selected a microphone as the source mic, move the cursor to the Model Microphone field (POD 1, bottom row), and select the model of the microphone to be modeled.

As with the source microphone, a -w indicates that a windscreen has been added to the model. There may also be variants of the base model, as described for the source microphone.

If the Bypass "microphone" is selected here, and a source microphone is selected, the effect will be that of the source microphone's characteristics. If Bypass is selected both for the source and the model, the final result of the modeler will be the input source, with the addition of any tube saturation added by the modeler (see below).

Model microphone parameters

As with the source microphone, there are a number of different additional parameters available which can be set.

Proximity As with the source microphone, the model can also have a proximity value set (in inches again). Use POD 2, bottom row.

When used with the model, it will affect the final character of the sound, as if the source was the specified distance from the model microphone when the recording was made.

Note that this setting cannot add "room tone" to a recording, even though the further away a real microphone is from a sound source, the more room tone is added to the final recording.

NOTE

Since omnidirectional microphones do not exhibit the proximity effect, if the model microphone is omnidirectional or has its pattern set to omnidirectional, this setting will have no effect.

Low-cut filter If the modeled microphone is fitted with a low-cut filter, this is also available on the model (if there is no filter available on the actual microphone being modeled, the model does not have a filter available, and shows None).

Note that this filter is not a straight low-cut filter—it is a representation of the actual filter incorporated on the physical microphone being modeled.

NOTE

Although it is not a hard and fast rule, it is a good idea to include the low-cut filter on the model if the filter has been used on the source microphone.

Response pattern As with the source microphone, the model can also take different response patterns (if the actual physical microphone being modeled is capable of this kind of flexibility—otherwise None is displayed for this option). Pod 3, bottom row is used here.

Remember that the modeler cannot spontaneously recreate missing data, so if a recording has been made with the source off-axis, this setting cannot be used to add the frequencies that were lost by the off-axis recording.

Preserve source settings

These settings allow you to make a hybrid microphone, dividing the microphones (both source and model) into their treble and bass components.

In this way, the two halves of the microphones can be "sandwiched" together to produce unusual creative effects.

Usually, however, you will want to keep the desirable characteristics of the source microphone (for example, a bass response) and eliminate the undesirable side (say, a poor treble response).

First, make all the appropriate source microphone settings. Bypassing the system is not a good idea here, as it will not have any useful effects.

Use the Preserve Source controls (PODs 1 and 2, second row) to select the portion of the source microphone that you want to keep (either the treble portion on the bass portion).

Keeping the original shows PRESERVE, and sending the signal through the processor shows PROCESS.

When a portion of the source microphone is preserved in this way, it overrides the corresponding portion of the model microphone.

Obviously preserving the source for both the bass and treble portions of the source is not terribly useful (though the proximity settings for both the source and the model remain effective).

Tube saturation

One of the more attractive aspects of older studio equipment is tube (valve) saturation. The microphone modeler provides you with a way to simulate this on the output side of the modeler.

Pick a value of the GAIN which suits your ears. The maximum value which may be set here is +10dB (0.1 dB steps). The signal to be recorded must therefore be at a level which is at least greater than –10dB for this to have any effect.

Use POD 4, second row to set the amount of gain.

However, you should take care that the input level is not increased to the point where digital distortion occurs.

You may need to “juggle” the values of the input level and the drive gain to achieve the most satisfactory results for this parameter.

Microphone models

The microphone models listed here are available for the DM-24 microphone modeler.

In the display, typically parameters and names are shown as given here, but the spacing of words on screen may sometimes differ from those given in the table.

The microphone response patterns are shown in uppercase, as follows: CARDIOID (cardioid), OMNI (omni-directional), HYPERCARDIOID (hyper-cardioid), FIGURE 8 (figure-of-8), WIDE CARDIOID (wide cardioid), w A98SPM (w A98SPM) and MS (MONO SIM) (MS (mono sim)).

Number	Microphone Maker	Microphone name	Low cut	Pattern
0	bypass mic	-	none	none
1	AKG	C 1000S	none	none
2		C 12A	none / -7 dB/oct / -12dB/oct	cardioid / omni
3		C 3000	off / on	cardioid / hypercardioid
4		C 4000 B	0 Hz / 100Hz	cardioid / hypercardioid / omni
5		C414	0 Hz / 75Hz / 150Hz	cardioid
6		C 414B-ULS (mod1)	0 Hz / 75Hz / 150Hz	cardioid / hypercardioid /figure 8 / omni
7		C 414B-UHS (mod2)	0Hz / 75Hz / 150Hz	cardioid / omni
8		C 414B-UHS Gold	0Hz / 75Hz / 150Hz	cardioid / hypercardioid /figure 8 / omni
9		C 414B-ULS Gold (w)	0Hz / 75Hz / 150Hz	cardioid / hypercardioid /figure 8 / omni
10		C 460 B, CK 61-ULS	0Hz / 50Hz / 70Hz / 150Hz	none
11		D 122 (1)	none	none
12		D 122 (2)	none	none
13		D 790	none	none
14	Alesis	AM61	off / on	none
15	Audio Technica	3525	off / on	none
16		4033	off / on	none
17		4047 sv	off / on	none
18		4050	off / on	cardioid / fugire8 / omni
19		4055	none	none
20		4060	none	none
21		853Rx	none	none
22		ATM11	none	none
23		ATM31	none	none
24	Audix	D4	none	none
25		OM2	none	none
26		OM3-xb	none	none
27		OM5	none	none
28	Beyer	CK-703	off / on	none
29		M-500 LE Classic	none	none
30		MC-834	LIN / 80Hz / 160Hz	none
31	Brauner	VM1	none	cardioid / hypercardioid / wide cardioid / figure 8 / omni
32	B & K	4007	none	none

Number	Microphone Maker	Microphone name	Low cut	Pattern
33	CAD	95Ni	none	none
34		C400S	none	none
35		Equitek E100	off / on	none
36		Equitek E200	off / on	cardioid / figiue8 / omni
37		Equitek E350	off / on	cardioid / figure8 / omni
38		VSM1 (mod 1)	off / on	none
39	Coles	4038	none	none
40	Earthworks	TC-30K	none	none
41		Z30X	none	none
42	ElectroVoice	N D 357	none	none
43		PL20	off / on	none
44	Gefell	UMT 800	off / on	cardioid / hypercardioid / wide cardioid / figure 8 / omni
45	Groove	Tubes MD-1	none	none
46	Lawson	L47	none	none
47	Manley	Reference Gold	none	cardioid / figure 8 /omni
48	Neumann	KM 184	none	none
49		KM 184(w)	none	none
50		M 149	20Hz / 40Hz / 80Hz / 160Hz	cardioid / hypercardioid/wide cardioid / figure 8 / omni
51		TLM 103	none	none
52		TLM 193	none	none
53		U 47	none	cardioid / omni
54		U 87 GOLD	off / on	cardioid / figure 8 /omni
55		U 87	off / on	cardioid / omni
56	Oktava	MC 012	none	cardioid / hypercardioid / omni
57		MK-319	off / on	none
58	RCA	BK-5A	M (music) / V1 (voice) / V2 (voice)	none
59	Rode	NT1	none	none
60		NT2	off / on	cardioid / omni
61		NT2(w)	off / on	cardioid / omni
62		NTV	none	none
63	Royer	R-121	none	none
64	Sennheiser	E 609	none	none
65		E 835S	none	none
66		MD 421	M (music) / 3 / 2 / 1 / S (speech)	none
67		MD 441	M (music) / 3 / 2 / 1 / S (speech)	none

Number	Microphone Maker	Microphone name	Low cut	Pattern
68	Shure	Beta 52	none	none
69		Beta 57A	none	none
70		Beta 87A	none	none
71		Beta 98D-S	none	none
72		KSM32	LC 0 / LC 1 / LC 2	none
73		SM57	none	none
74		SM58	none	none
75		SM7A	LC off Mid off / LC off Mid on / LC on Mid off / LC on Mid on	none
76		SM81	LC 0 / LC 1 / LC 2	none
77		SM98A	off / on	w A98SPM
78		VP88 (mono sim)	off / on	MS (mono sim)
79	Sony	C37P	M / M1 / V1 / V2	none
80		C48	M (music) / V (voice)	cardioid / figure 8 / omni
81		C800G	none	cardioid / omni
82		C800G(w)	none	cardioid / omni
83	Telefunken	TELE U47	none	cardioid / omni

Updating microphone models

The modeler provides up to 100 models of microphone. More may be made available in the future through the TASCAM Web site.

Consult your dealer for availability.

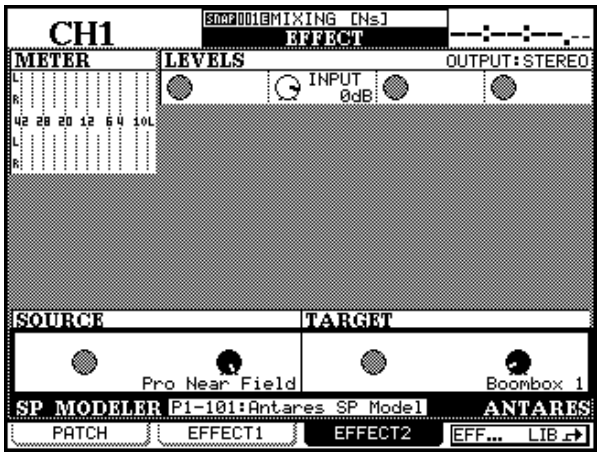
In the same way that microphones can be modeled, the DM-24 allows for the modeling of speakers.

Once again, it is important to remember that it is not possible to instantly transform a pair of low-end near-field monitors into a pair of expensive, top-of-the-line monster monitors (even modern technology has its limits), but it can be useful for simulating some of the speaker types on which your final project will be played, and for which you may not have space in your control room (or where it may be inconvenient to reproduce the sound—for example, not many people will wish to purchase a SUV merely for the acoustical properties of the interior!).

The technique for using this is similar to the microphone modeler, but not so complex.

Basically, you define a set of *source speakers* (the real speakers that you are listening to) and a set of *target speakers* (the ones that you wish to model).

This speaker modeler can be inserted anywhere in the signal chain, but obviously it is more useful if it is selected as an insert in the main stereo outputs.



NOTE

Due to technical limitations, if the speaker modeler is selected as one effect, the reverb cannot be selected as the second effect.

It is also not available in high sampling frequency mode.

Selecting the speaker modeler

Recall the preset library entry 1-101 in order to load the speaker modeler.

See “Setting up the effects units” on page 7 for further details.

General parameters

There are three general settings which are all set using the top row of PODs.

INTYPE stands for input type. There are four options here: Stereo, L mono, R mono and LR mono. The first three explain themselves, but the last refers to a mono signal comprised of the L and R output signals added together.

INPUT the input level can be adjusted (in 1 dB steps) from –30 dB to +6 dB.

BYPASS the whole of the speaker modeler effect can be turned on and off with this parameter.

Source speaker types

The source speakers which may be selected are generic types of speaker, not individual models.

The different selections available (pod 2, bottom row) are:

- Bypass speaker (as if there was no output speaker connected to the DM-24)
- Cheap Near Field (for “cheap”, read “low-cost, but acceptable performance”, but that is too long to fit on the display!)
- Large Studio (dedicated studio monitors)
- Mid Field Studio
- Near Field (better quality than the “cheap” model)
- Pro Near Field (more expensive than the other models here)

Choose the setting which you feel comes closest to your set of speakers.

You can test the source model by selecting the Bypass type for the target speaker and changing between the different models, making A-B comparisons with the whole effect bypassed in order to achieve the closest match.

Target speaker types

The speakers modeled here are generic, rather than reproducing a particular make or model of speaker.

They represent a wide range of speaker types on which your material may eventually be played.

- Bypass speaker (no model for the output speaker)
- Boombox 1 (one type of “boombox”)
- Boombox 2 (another variation on the boombox theme)
- Car Sedan (an average car sound system)
- Car SUV (the kind of sound you might expect from an SUV sound system)
- Compact Stereo (domestic stereo system, but small speakers)
- Computer Speaker (useful for multimedia sound mixes)
- Large Home Studio (good quality domestic/semi-pro speakers)
- Mid Sound Reinforcement (not necessarily top-of-the-range, but good-quality sound reinforcement)
- Small Home Studio (smaller speakers intended for the musician/home recordist)
- TV (typical TV speaker sound)

Use POD 4 on the bottom row to select the speaker type.

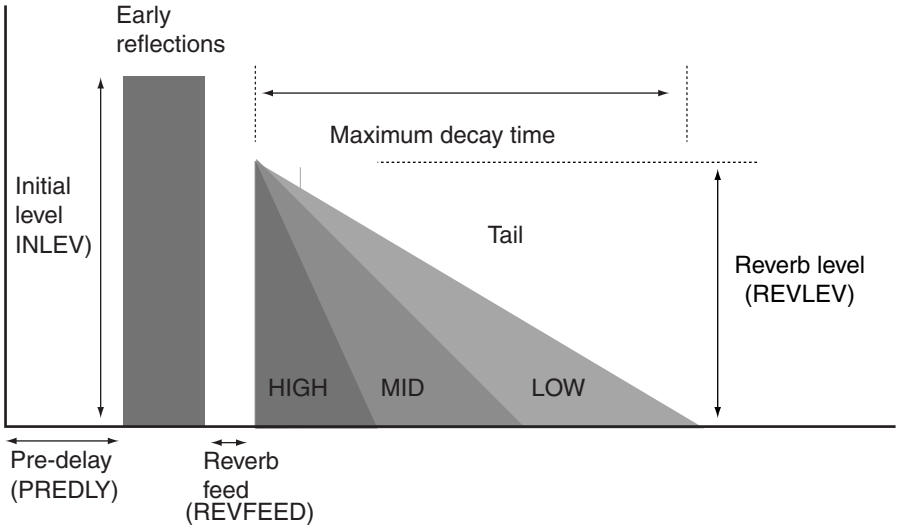
A few limitations

Once again, it is necessary to emphasize that you cannot turn a poor-quality set of speakers into an expensive pair of monitors.

However, what you can do is to reproduce the tonal characteristics of a certain type of speaker and environment, allowing you to “field-test” your project for a particular purpose without even changing your speaker system.

The TC Works reverberation built into the DM-24 is a sophisticated reverb system, allowing you to simulate many different kinds of acoustic environment.

Most common (and a few less common) parameters can be edited, allowing fine control of the whole sound.



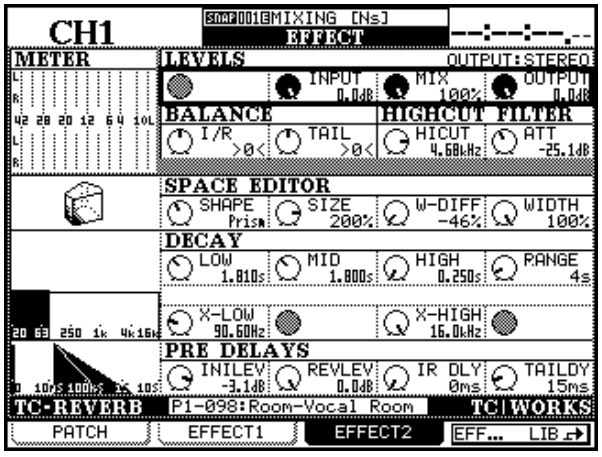
NOTE

Due to technical limitations, if reverb is selected as one effect, the reverb or speaker modeler cannot be selected as the second effect.

It is also not available in high sampling frequency mode.

General parameters

There are three general settings which are all set using the top row of PODs.



INTYPE stands for input type. There are four options here: Stereo, L mono, R mono and LR mono. The first three explain themselves, but the last refers to a mono signal comprised of the L and R output signals added together.

INPUT the input level can be adjusted from Off, and then (in 5 dB steps) from –140 dB to –60 dB, (in 1 dB steps) from –60 dB to –20 dB and (in 0.1 dB steps) from –20 dB to 0 dB.

OUTPUT The output level can be adjusted from Off, and then (in 5 dB steps) from –140 dB to –60 dB, (in 1 dB steps) from –60 dB to –20 dB and (in 0.1 dB steps) from –20 dB to 0 dB.

MIX Adjust the wet/dry mix from 0% (dry) to 100% only reverb) in 101 1% steps.

Balance controls

There are two balance controls on the second POD row (BALANCE).

I/R This stands for “Initial Reflections” (sometimes called Early Reflections”). POD 1 controls the left/right balance of these reflections from 50 through 0 to 50.

TAIL This stands for “reverb tail”—the final decay of the reverb sound. POD 2 controls the left/right balance of these sounds from 50 through 0 to 50.

High-cut filter

This filter can be used to cut off the upper frequencies of the reverb signal.

Use PODs 3 and 4 on row 2 of the screen.

HICUT Sets the frequency at which the filter operates. Use POD 3 to select a value. The lower limit is 20 Hz, and the higher limit is 16 kHz.

ATT Short for "attenuation"—the amount by which the high-cut filter cuts the upper frequencies. Use POD 4 to set this value from –40.0 dB to 0 dB in 0.1 dB steps.

Space editor

These four parameters allow you to set the basic reverb type (row 3, labeled SPACE EDITOR).

SHAPE allows you to set the basic shape of the simulated room in which the sound is being reflected to produce the reverberation effect.

Turn POD 1 to select from the following list (a small representation of the room shape appears on the left of the screen as you make these changes):

- HALL—a hall-shaped room (basically a cube)
- H.SHOE—(horseshoe) a room where one wall is flat, and the other walls curve round.
- PRISM—a prism-shaped space with two parallel walls focusing down to a wedge (similar to many auditoria).
- FAN—even more wedge-shaped than the prism setting.
- CLUB—a T-shaped space, with a recessed stage area.
- SMALL—a smaller, more intimate version of the cube.

SIZE The size of the space. Units are arbitrary, and may be set from 0.04 to 4.0 in the following steps: 0.04, 0.05, 0.06, 0.08, 0.10, 0.13, 0.16, 0.20, 0.25, 0.32, 0.40, 0.50, 0.63, 0.80, 1.0, 1.3, 1.6, 2.0, 2.5, 3.2, 4.0.

W-DIFF Wall diffusion—the "liveness" of the room space, and the amount that the sound is scattered. Set a value from –50% to +50% I 1% steps.

WIDTH Not, strictly speaking, the width of the simulated room, but the stereo width of the reverb signal (which is affected by the width of the room). Set from 0% (mono point source) to 100% (full width) in 1% steps.

Decay characteristics

The decay can be set for three bands independently, allowing, for example, the treble portion of the sound can continue to reverberate after the bass and mid sounds have decayed, giving a bright quality to the reverb.

The crossover points for the three bands can be set independently.

Use the four pods in row 4 and the two pods in row 5 (labeled DECAY) to set the band times, as well as a "scale" which allows fine-tuning of the times without having to turn the PODs an excessive number of times.

LOW, MID, HIGH Each of the three bands can be set independently, in a range from 0.25 s to 9.99 s and from 10.0 s to 64 s (1024 steps in total).

RANGE The overall range for these three bands (and therefore the number of times the PODs must be turned to set a value) an be set to one of three values: 4 s, 16 s and 64 s.

X-over The two crossover frequencies to divide the sound spectrum into three bands can be set. Each of these frequencies can be set between values of 20 Hz and 16 kHz. Use PODs 1 and 3 on row 5 to set these values.

Pre-delay settings

The bottom row of the screen allows different delay settings to be made.

See the diagram above for details of exactly what these settings change.

INLEV This is the initial level of the early reflections.

Use POD 1 to set this value from Off, through –140 dB to 0 dB.

REVLEV This is the level at which the decay "tail" portion of the reverb starts.

Use POD 2 to set this value from Off, through –140 dB to 0 dB.

PREDLY This is the pre-delay portion of the reverb. It describes the time from the initial sound to the first of the initial reflections.

Use POD 3 to set this value from 0 ms to 160 ms in 1 ms steps.

REVFEED This is the time separating the feed from the initial reflections to the "tail" part of the reverb.

Use POD 4 to set this value from 0 ms to 100 ms in 1 ms steps.

Conclusion

Although the array of parameters and options for this reverb may seem a little baffling at first, compared with some other units, a little experimentation will soon make it clear what the different parameters actually control in terms of the sound produced.

The preset library entries provide useful starting points for your own experiments, allowing you to either simulate real reverb situations, or to invent imaginary spaces with their own, distinctive, reverberation characteristics.

Preset reverb settings

These settings are all stored in the first preset effects library.

The English names give an idea of the kind of sound that can be obtained from these settings.

Ambience settings basically give a feeling of life, without a definite reverb.

Box settings are smaller or and typically have a rather "live" sound.

Chamber settings provide a sound a little like a room type of reverb.

FX settings provide a special effect, which may not sound totally natural, but may have a useful place in your project.

Tunnel settings provide the image of a long narrow live space.

Hall settings give the sound of larger enclosed spaces. There is a range of hall settings provided, with different acoustical characteristics, including "church" and "cathedral" settings.

Drum settings are specifically tailored for use with drum instruments. Of course, they can be used with other instruments and sources, but they may not be so effective as when they are used with drum sources.

Perc settings are suitable (but not exclusively) for use with percussion instruments and percussive sounds.

Plate Settings reproduce the sound of a vintage plate reverb unit.

Room settings provide the effect of a smaller, tighter space than a hall.jcASCAM effects

Number	Name	LCD indication
000	Ambience - Bright 1	Ambi-Bright 1
001	Ambience - Bright 2	Ambi-Bright 2
002	Ambience - Bright 3	Ambi-Bright 3
003	Ambience - Dark	Ambi-Dark

Number	Name	LCD indication
004	Ambience - Midnight	Ambi-Midnight
005	Ambience - Mornin' Vocal	Ambi-MorninVocal
006	Ambience - Soft 1	Ambi-Soft 1
007	Ambience - Soft 2	Ambi-Soft 2
008	Ambience - Space	Ambi-Space
009	Box - Bright	Box-Bright
010	Box - Dark	Box-Dark
011	Chamber - Large, Dark	Chmb-Large,Dark
012	Chamber - Small	Chmb-Small
013	Chamber - Small, Dark	Chmb-Small,Dark
014	Chamber - Very Small	Chmb-Very Small
015	FX - Big Barrel Space	FX-BigBarrelSpce
016	FX - Big Pre Delay Slap	FX-BigPreDlySlap
017	FX - Bright Cymbals	FX-BrightCymbals
018	FX - Drum Boom Slap	FX-DrumBoom Slap
019	FX - Dry After Taste	FX-DryAfterTaste
020	FX - Icy Shower	FX-Icy Shower
021	FX - Lost in Space	FX-Lost in Space
022	FX - Neighbor (Hallway)	FX-NeighborHallw
023	FX - Neighbor 2 (Floor)	FX-NeighborFloor
024	FX - Not so Dry After Taste	FX-NotsoDryAfter
025	FX - Short Non-Lin Like	FX-Short Non-Lin
026	FX - Slap Back	FX-Slap Back
027	FX - Steel Works	FX-Steel Works
028	FX - Steel Works 2	FX-Steel Works 2
029	FX - Subtle Slapback	FX-SubtleSlapbac
030	FX - Take Off	FX-Take Off
031	FX - Tight Bounce Around	FX-Tight Bounce
032	FX - Ultra Bright	FX-Ultra Bright
033	FX - Under The Surface	FX-Under Surface
034	FX - Wet After Taste	FX-WetAfterTaste
035	FX - Wet After Taste w/Rain	FX-W.A.T w/Rain
036	FX - Wood Floor	FX-Wood Floor
037	Tunnel - Bright	Tunn-Bright
038	Tunnel - Dark	Tunn-Dark
039	Tunnel - Tube	Tunn-Tube
040	Hall - Big Bright	Hall-Big Bright
041	Hall - Big Clear	Hall-Big Clear
042	Hall - Big Predelayed	Hall-BigPredelay
043	Hall - Big Warm	Hall-Big Warm

Number	Name	LCD indication
044	Hall - Cathedral 12s	Hall-Cathdral12s
045	Hall - Cathedral 7s	Hall-Cathedral7s
046	Hall - Church	Hall-Church
047	Hall - Dome	Hall-Dome
048	Hall - Huge Clear	Hall-Huge Clear
049	Hall - Huge Warm	Hall-Huge Warm
050	Hall - Last Row Stadium Con	Hall-LastRowStdm
051	Hall - Lush Ballad	Hall-Lush Ballad
052	Hall - Medium Bright	Hall-Med.Bright
053	Hall - Medium Clear	Hall-MediumClear
054	Hall - Medium Warm	Hall-Medium Warm
055	Hall - Outside the Stadium	Hall-OutsideStdm
056	Hall - Small Bright	Hall-SmallBright
057	Hall - Small Clear	Hall-Small Clear
058	Hall - Small Warm	Hall-Small Warm
059	Hall - Stage	Hall-Stage
060	Hall - Warm Vocal Hall	Hall-Warm Vocal
061	Drum - Boom Room	Drum-Boom Room
062	Drum - Drum Booth	Drum-Drum Booth
063	Drum - Huge Low Tubular	Drum-HugeLowTubu
064	Drum - Low Tubular	Drum-Low Tubular
065	Drum - Snare Hall	Drum-Snare Hall
066	Drum - Snare Room	Drum-Snare Room
067	Drum - Subtle Kick Boom	Drum-SubtleKick
068	Perc - Big Bright	Perc-Big Bright
069	Perc - Big Clear	Perc-Big Clear
070	Perc - Big Warm	Perc-Big Warm
071	Perc - Medium Bright	Perc-Med.Bright
072	Perc - Medium Clear	Perc-MediumClear
073	Perc - Medium Warm	Perc-Medium Warm
074	Perc - Small Bright	Perc-SmallBright
075	Perc - Small Clear	Perc-Small Clear
076	Perc - Small Room	Perc-Small Room
077	Perc - Small Warm	Perc-Small Warm
078	Plate - Big Bright	Plat-Big Bright
079	Plate - Big Clear	Plat-Big Clear
080	Plate - Big Warm	Plat-Big Warm
081	Plate - Tight	Plat-Tight
082	Room - Bathroom	Room-Bathroom
083	Room - CD Master	Room-CD Master

Number	Name	LCD indication
084	Room - Dark & Mellow 5 sec	Room-Dark&Mellow
085	Room - Dry House	Room-Dry House
086	Room - Empty Garage	Room-EmptyGarage
087	Room - Empty Room	Room-EmptyRoom
088	Room - Empty Room, Small	Room-EmptyRoom S
089	Room - Large Garage	Room-LargeGarage
090	Room - Percussion Room	Room-Perc Room
091	Room - Small	Room-Small
092	Room - Small Damped Room	Room-S Dmp Room
093	Room - Small Yet Big	Room-SmallYetBig
094	Room - Small Yet Big w/Pre	Room-S.Y.B w/Pre
095	Room - Stage	Room-Stage
096	Room - Vocal Booth	Room-Vocal Booth
097	Room - Vocal Dry	Room-Vocal Dry
098	Room - Vocal Room	Room-Vocal Room
099	Room - Vocal Room 2	Room-Vocal Room2
These are not reverb settings, but this is the way in which the microphone and speaker modelers are selected		
100	Antares AMM-1	Antares AMM-1
101	Antares SP modeler	Antares SP Model

The general effects provided within the DM-24 as described in "Internal Effects on the DM-24" on page 3 may be used either as in-line inserted effects, or as part of an effect loop, using aux send and returns.

There are no hard and fast rules as to how these effects can be used, but on the whole, any effect where there is a wet/dry level is suitable for use within effect loops, and the others are suitable for use as insert processors.

Common parameters

There are three parameters on the top row of the screen for all of these effects, using PODs 2, 3 and 4.

INPUT stands for input level. The input level of the signal fed to the effect is adjusted using this control.

MIX is the wet/dry balance of the output. When set to 0%, the output signal is composed totally of the original signal, and at 100%, it is completely the effect signal.

NOTE

There is a slight unavoidable processing delay on some effects. If you are using the effect as an insert, you should probably keep this setting at 100%.

OUTPUT The output level from the effect can be adjusted from Off, and then (in 5 dB steps): from –140 dB to –60 dB, (in 1 dB steps) from –60 dB to –20 dB and (in 0.1 dB steps) from –20 dB to 0 dB.

Effect parameters

The different parameters for use within these effects are as follows:

Chorus	Rate	Depth	Pre- delay	Feedback	Wet Mix Level	Dry level	
	0.1 Hz to 10 Hz (91 steps)	0% to 100% (101 steps)	0.05 ms to 500 ms (101 steps)	0% to 90% (91 steps)	–40 dB to –0 B (41 steps)	–40 dB to +20 dB (61 steps)	
De-esser	Threshold level	Knee shape	Center Frequency	Output level	Bypass		
	–40 dB to –1 dB (40 steps)	0.50 to 1.00 (11 steps)	1.0 kHz to 10 kHz (91 steps)	–40 dB to _20 dB (61 steps)	On/off		
Delay	Delay Time	Feedback Time	Feedback Level	Feedback Type	Wet Mix Level	Output level	
	0.05 ms to 650 ms (651 steps)	0.05 ms to 650 ms (651 steps)	0% to 90% (91 steps)	Stereo/ Ping-pong/ Multi-tap	–40 dB to –0 B (41 steps)	–40 dB to +20 dB (61 steps)	
Distortion	Drive Ratio	Drive Boost	EQ Pattern	Output level			
	0 to 42 (43 steps)	x 1 to x32 (32 steps)	Overdrive 1/2, Distortion 1/2, Amp 1/2	–40 dB to +20 dB (61 steps)			
Exciter	Sense	Frequency	Output level	Bypass			
	0 to 42 (43 steps)	1.0 kHz to 10 kHz (91 steps)	–40 dB to +20 dB (61 steps)	On/off			
Flanger	Rate	Depth	Resonance	Delay	Wet Mix Level	Dry level	
	0.1 Hz to 10 Hz (91 steps)	0% to 100% (101 steps)	0.00 to 1.00 (91 steps)	0.05 ms to 500 ms (101 steps)	–40 dB to –0 B (41 steps)	–40 dB to +20 dB (61 steps)	
Guitar Compressor	Ratio	Attack	Output level	Bypass			
	0 to 42 (43 steps)	0.1 ms to 5.0 ms (50 steps)	–40 dB to +20 dB (61 steps)	On/off			
Phaser	Steps	LFO Rate	LFO Depth	Resonance	Output level	Bypass	
	1 to 16 (16 steps)	0.1 Hz to 10 Hz (99 steps)	0% to 100% (101 steps)	0% to 100% (101 steps)	–18 dB to +12 dB (31 steps)	On/off	
Pitch	Semitone shift	Pitch Fine	Pre- Delay	Feedback	Wet Mix Level	Dry level	
	–12 to +12 (25 steps)	–50 to +50 (101 steps)	0.05 ms to 500 ms (91 steps)	0% to 90% (91 steps)	–40 dB to –0 B (41 steps)	–40 dB to +20 dB (61 steps)	

	Threshold	Ratio	Attack Time	Release Time	Knee Shape	Output level	Bypass
Soft Compressor	–40 dB to –1 dB (40 steps)	1:1.00 to 1:∞	0.05 s to 5.0 s (100 steps)	50.0 ms to 500 ms (451 steps)	1.0x to 0.5x (21 steps)	–18 dB to +12 dB (31 steps)	On/off

Some of these characteristics are difficult to explain in words, and quite frankly, the only way in which you can find out exactly what they do is to experiment with the settings, if you are unfamiliar with them.

However, the bulk of these settings should be familiar to anyone who has used any multi-effects processor in the past.

A few notes may be in order here:

- All effects here are dual-channel except for the distortion and the guitar compressor, which are single-channel effects.
- The "EQ patterns" in the distortion corresponds to the approximate equivalent patterns produced by a number of popular guitar and bass amplifier/speaker combinations. Experiment with these to obtain the guitar sound you want (or any other instrument you care to put through this processor)
- The different patterns on the delay correspond to Stereo, Ping-pong and Multi-tap settings.
- The knee shape on the soft compressor affects the sharpness of the compressor effect.

Preset effect settings

Please bear in mind that the descriptions here are very subjective. When a sound is recommended "for use with bass", for example, this is not a rule—simply a recommendation.

Feel free to play with the different sounds and experiment, using these preset sounds as the basis for your own effects.

Effect type	Preset No.	Title	LCD indication	Comments
Guitar Compressor				
	0	Guitar Comp.	Guitar Comp.	Basic compressor
	1	Classic Comp.	Classic Comp.	A classic compressor sound
	2	Sustain	Sustain	Compressor setting for guitar sustain
	3	Fat Comp.	Fat Comp.	A rather deeper, "fatter" type of compression.
	4	Deep Comp.	Deep Comp.	Deep compressor sound.
	5	Rhythm Comp.	Rhythm Comp.	A cutting compression setting for percussion.
	6	Fast Attack	Fast Attack	A fast attack setting.
	7	Slow Attack	Slow Attack	A rather slower attack setting.
	8	Slap Comp.	Slap Comp.	Suitable for slap bass
	9	Percussive	Percussive	A clean sound, suitable for percussive guitar work, etc.
Distortion				
	10	Distortion	Distortion	A basic distortion sound.
	11	Over Drive	Over Drive	A basic overdrive sound.
	12	Blues 1	Blues 1	Suitable for a "front pickup" blues guitar style.
	13	Blues 2	Blues 2	A rather stronger sound than the previous Blues 1 sound.
	14	Vocal Dist	Vocal Dist	Useful if you need distorted vocals.
	15	Rock 1	Rock 1	Suitable for 70s rock music.
	16	Rock 2	Rock 2	Another kind of rock-type distortion.
	17	Rhythm 1	Rhythm 1	A sweet-sounding distortion for backing work.

Effect type	Preset No.	Title	LCD indication	Comments
	18	Rhythm 2	Rhythm 2	A lighter backing style distortion.
	19	Bass Dist	Bass Dist	Use this distortion with bass.
	20	Fusion 1	Fusion 1	Use this with solo instruments to fill out the sound.
	21	Fusion 2	Fusion 2	Distortion used for a smooth, sweet fusion style.
	22	British	British	A fat classic “single-coil” distortion.
	23	Fuzzy	Fuzzy	A rather heavy fuzz distortion.
	24	Guts	Guts	A “single-coil” overdrive sound.
	25	Sweet	Sweet	A rather sweet, “rear pickup”-type solo sound.
	26	Mellow	Mellow	Mellow distortion. Try with the front pickup.
	27	Cheap	Cheap	A cheap and cheerful distortion sound.
	28	Lead	Lead	A lead solo distortion sound.
	29	Bottom	Bottom	Somewhat bass-heavy driving sound.
	30	Strong	Strong	A powerful driving sound.
	31	Trebly	Trebly	Driving sound with a lot of treble.
	32	Solo	Solo	"Humbucker” solo sound.
	33	Crunch	Crunch	“Crunch”
	34	Fat Drive	Fat Drive	A thick, fat sound
Compressor				
	35	Comp	Comp	Basic compressor sound
	36	Fast Attack	Fast Attack	A compressor with a fast attack
	37	Slow Attack	Slow Attack	A compressor with a slow attack.
	38	Short Release	Short Release	Quick-release compressor.
	39	Long Release	Long Release	Slow release compressor.
	40	Vocal Comp 1	Vocal Comp 1	Use this compressor with vocals.
	41	Vocal Comp 2	Vocal Comp 2	Maybe a little more natural-sounding than Vocal 1.
	42	Inst	Inst	This setting is good with a rhythm box or drum machine.
Exciter				
	43	Exciter	Exciter	Helps the definition of musical sounds.
	44	Edge	Edge	The treble is attenuated in this setting.
	45	Vocal EX	Vocal EX	Suitable for use with vocals.
	46	Rhythm G	Rhythm G	Use this setting with rhythm guitars.
	47	Bass EX	Bass EX	Use with bass guitars and instruments.
De-esser				
	48	De-esser	De-esser	Use this to reduce sibilance.
Phaser				
	49	Phaser	Phaser	Basic phase sound.
	50	G Phaser 1	G Phaser 1	Use this phase with guitars
	51	G Phaser 2	G Phaser 2	Use this with backing guitars.
	52	G Phaser 3	G Phaser 3	Use the resonance in this sound with guitars.
	53	Bass Phaser 1	Bass Phaser 1	Use with fast passages from bass guitar.

Effect type	Preset No.	Title	LCD indication	Comments
	54	Bass Phaser 2	Bass Phaser 2	Slower bass phaser.
	55	Rhythm Phaser 1	Rhythm Phaser 1	A phase for cutting rhythm.
	56	Rhythm Phaser 2	Rhythm Phaser 2	Fast rhythm phaser.
	57	Vocal Phaser 1	Vocal Phaser 1	A rather “loose” phase sound.
	58	Vocal Phaser 2	Vocal Phaser 2	“Inspirational” vocal phasing.
	59	Drum Phaser	Drum Phaser	Use this with drums to create space.
	60	Fusion Phaser	Fusion Phaser	Sounds good with fusion styles.
	61	Vibrato Phaser	Vibrato Phaser	Phase used as vibrato.
	62	Wah Phaser	Wah Phaser	Phaser used like a wah-wah pedal.
Delay				
Stereo	63	Delay	Delay	A basic delay setting.
	64	Long Echo	Long Echo	Long echo setting.
	65	Stereo Echo	Stereo Echo	A long stereo echo sound.
	66	Bath	Bath	Singing in the bath?
	67	Doubling	Doubling	A doubling echo setting.
	68	One Time	One Time	One-shot echo.
	69	Rhythm Echo	Rhythm Echo	A good echo sound to use with drums.
Ping-Pong	70	Oasis	Oasis	A lazy, casual echo sound.
	71	Short Echo	Short Echo	Short repeat echo.
	72	Loose	Loose	Slightly looser” echo setting.
	73	Vocal Echo 1	Vocal Echo 1	A “karaoke”-type echo.
	74	Vocal Echo 2	Vocal Echo 2	Use this setting with vocals for a short repeat.
Multi-tap	75	Cross Feedback	Cross Feedback	Left and right echoes cross over.
	76	Cool	Cool	Almost a vibrato setting.
	77	100bpm 1	100bpm 1	Use this at 100bpm.
	78	100bpm 2	100bpm 2	
	79	120bpm 1	120bpm 1	Use this at 120bpm.
	80	120bpm 2	120bpm 2	
	81	150bpm 1	150bpm 1	Use this at 150bpm.
	82	150bpm 2	150bpm 2	
Chorus				
	83	Chorus	Chorus	The basic chorus sound.
	84	Backing Chorus	Backing Chorus	An “arpeggio” type of chorus setting.
	85	Fast Chorus	Fast Chorus	A fast chorus setting.
	86	Slow Chorus	Slow Chorus	A slower, lazier chorus.
	87	Soft Chorus	Soft Chorus	Soft and gentle.
	88	Deep Chorus	Deep Chorus	A deep chorus sound.
	89	Ensemble 1	Ensemble 1	A thick, “multiple” chorus sound.
	90	Ensemble 2	Ensemble 2	A chorus sound with a strong tremolo.
	91	Ensemble 3	Ensemble 3	Another kind of ensemble sound.

Effect type	Preset No.	Title	LCD indication	Comments
	92	Clean Chorus 1	Clean Chorus 1	A light chorus sound.
	93	Clean Chorus 2	Clean Chorus 2	Use this clean sound with vocals.
	94	Clean Chorus 3	Clean Chorus 3	A vibrato-type chorus effect.
	95	Chorus Flange 1	Chorus Flange 1	A feedback chorus setting, almost like a flanger.
	96	Chorus Flange 2	Chorus Flange 2	A flanger-like setting for use with bass.
	97	Chorus Flange 3	Chorus Flange 3	Strong modulation setting.
Pitch				
	98	Pitch shifter	Pitch shifter	Octave doubler.
	99	Ensemble 1	Ensemble 1	A repeat setting to give an ensemble effect.
	100	Ensemble 2	Ensemble 2	A short repeat provides a “coming and going” effect.
	101	Ensemble 3	Ensemble 3	Useful when used with chorus.
	102	3th Harmony 1	3th Harmony 1	Thirds-type harmony.
	103	3th Harmony 2	3th Harmony 2	Lower thirds harmony.
	104	Octave 1	Octave 1	Octave up pitch shift.
	105	Octave 2	Octave 2	Octave down pitch shift.
	106	5th Harmony 1	5th Harmony 1	Fifth up harmony.
	107	5th Harmony 2	5th Harmony 2	Fifth down harmony.
	108	Pitch Chorus 1	Pitch Chorus 1	Detune and echo gives a chorus effect.
	109	Pitch Chorus 2	Pitch Chorus 2	Strong pitch change effect provides a chorus-like feel.
	110	12 Strings	12 Strings	12-string guitar emulation.
	111	Glow up	Glow up	Pitch shift and feedback for an interesting effect.
	112	Mystery	Mystery	A sound of mystery.
Flanger				
	113	Flanger	Flanger	A “sparkling” flanger setting.
	114	G Flanger 1	G Flanger 1	Use this flanger setting with guitars.
	115	G Flanger 2	G Flanger 2	A fast flange setting.
	116	G Flanger 3	G Flanger 3	A looser flange setting.
	117	Bass Flanger 1	Bass Flanger 1	Use this flanger with bass instruments.
	118	Bass Flanger 2	Bass Flanger 2	Another sound for use with bass instruments.
	119	Vocal Flanger	Vocal Flanger	This can be used to add life to vocals.
	120	Funny	Funny	Creatures from outer space?
	121	Jet Flanger 1	Jet Flanger 1	Resonance to simulate a jet takeoff.
	122	Jet Flanger 2	Jet Flanger 2	A spacious “jet” sound.
	123	Sweet Flanger	Sweet Flanger	A smoother, sweet flange setting.
	124	Flanger Echo	Flanger Echo	Repeat and flange together.
	125	Tremolo Flange	Tremolo Flange	Flanger used as a tremolo.
	126	Deep Flanger	Deep Flanger	A deep flanger setting.
	127	Metallic Tone	Metallic Tone	A flanger setting giving a metallic tone.

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