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Introduction

This manual covers all of the SCOPE Fusion Platform 3.1 effects. Some of the effects are included in the Main Studio Package, which comes with all products. Others are components of the Effects I or II packages which you may or may not have acquired. To determine which effects are included with your DSP package, click the Effects-Page-Button on page 1 of this chapter.

Some of the control elements found on the effect panels are standard, and therefore not described here for each individual effect.

Channel Strip - Name

This text displays the mixer channel using the effect. It remains empty under XTC.

Preset List Switch

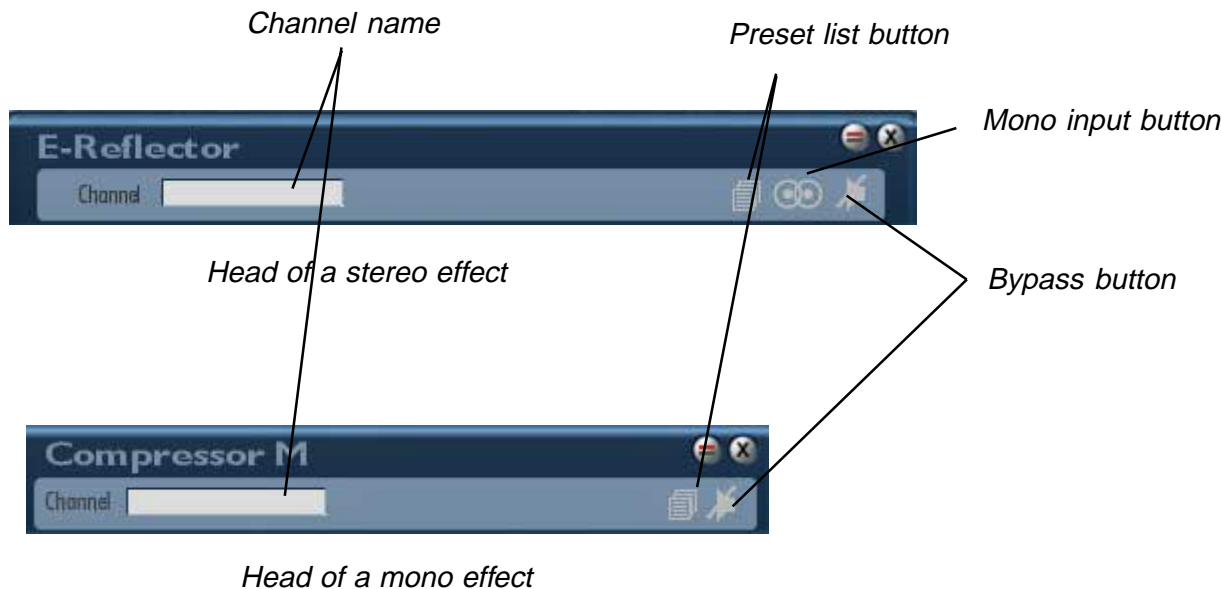
Pulsar features the capability to store and recall presets. This opens the effect's Preset List.

Mono Input Switch

Stereo effects can also be used to process an input signal only through the left channel.

Bypass Switch

Enable the bypass switch when you want to hear only the original signal for comparison purposes.



MasterVerb and MasterVerb Classic

These effects simulate concert hall reverberation. Each of the two versions consists of one section to produce the initial echoes (early reflections) and a second to produce the response (reverb or response). Early reflection, or more precisely, the pattern of early reflections, is essentially responsible for the perception of space—the size of the hall. The response signals produce the overall atmosphere. Filters in the input and response sections control the tonal color of the hall.

The MasterVerb produces a range of authentic early reflection models, while the Classic version provides only a single simulation of early reflections. With regard to DSP usage, the Classic version is more economical than the MasterVerb. The Masterverb's DSP cost is justified by a more realistic impression of space, however. The reverb section is identical in both versions. Choose the version to use with respect to the production requirements. For an accurate sense of spaciousness, use the MasterVerb. To save DSP resources, or to produce only the atmospheric component, use the Classic version.



Input Gain

Controls the level of the signal to be processed. A VU-meter displays the level of the signal as adjusted. A small mark next to the control indicates unity gain. When adjusted to the left of the mark, the signal is attenuated. To the right of the mark, the signal is amplified.

Low Pass Filter

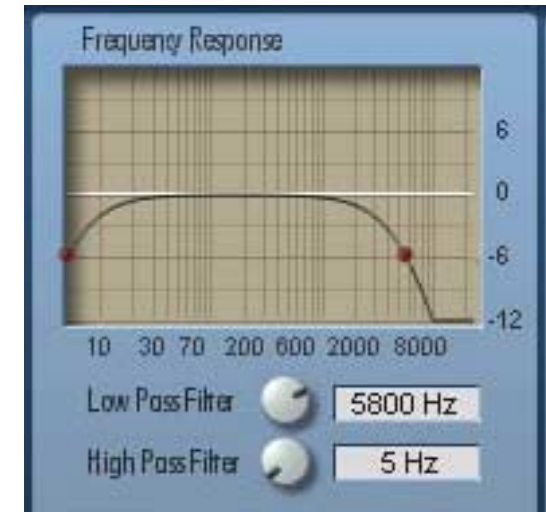
A low pass filter with a slope of 12db/octave follows the gain control in the signal path. There are three ways you can adjust the cutoff frequency: Use the rotary control, enter the value in the text field, or adjust the value directly in the graphic display area using the mouse.

Rooms or halls we classify as "warm" absorb most of the high frequencies—those over 8 kHz (or less). Use the low pass filter to create a warmer response.

High Pass Filter

A high pass filter, also with a slope of 12db/octave, follows the low pass filter in the signal path. As with the high pass filter, there are three ways to adjust the cutoff frequency: Use the rotary control, enter the value in the text field, or adjust the value directly in the graphic display area using the mouse.

The response of many halls lies mainly within the middle frequencies. Together, the low pass and high pass filters form a band pass filter, attenuating both high and low frequencies. Appropriate adjustment of the two filters reproduces the sound of a hall in which the midrange is favored.



Early Reflections (MasterVerb only)

In the MasterVerb, the early reflection response is made up of 16 individual "echoes". Adjust the configuration and characteristics of these as described in the following parameters. The filters described previously influence their tone quality.

Early Refl. (Early Reflections)

This button sets the graphic area to display Early Reflections settings.

ER Type (Early Reflections Type)

From the drop-down list select an early reflection model. The model chosen determines the quality of the spatial impression.

ER Spread (Early Reflections Spread)

Selects the apparent size of the hall to be processed for early reflections.

ER Decay (Early Reflections Decay)

Controls how the early reflections die away over time. With higher values: the greater the delay with respect to the original signal, the quieter the reflections become. For a natural effect, set the decay to its maximum value to shorten the early reflections.

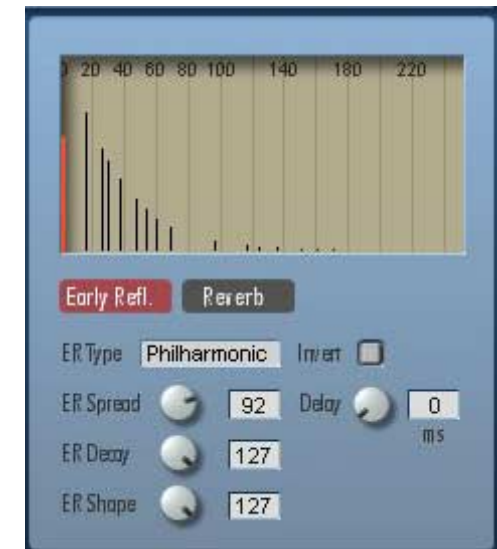
ER Shape (Early Reflections Shape)

The Shape parameter determines the decay "envelope" –the decay curve. An envelope with an exponential shape, as adjusted by setting ER Shape to its maximum value, produces the most natural effect.

Invert

Inverts the decay envelope of the early reflections. In other words, the Early Reflections model remains the same, but each reflection acquires a gain, rather than a diminution, of volume.

This effect is not found in natural acoustical systems, but can be interesting as a special effect.



Reflections (MasterVerb Classic only)

The Reflections section of the MasterVerb Classic produces simulated early reflections at a much lower computational cost than the MasterVerb. The Reflections simulation is achieved by taking part of the response (reverb) signal, splitting it, and delaying it using individual delays for the left and right stereo channels. The density of the Reflections signal results from the settings of the Reverb parameters (q.v.). The filters described previously control the tone quality.

This method of producing early reflections is typical of older digital reverberation techniques, therefore the "Classic" designation in the name.

Reflections

This button sets the graphic area to display Reflections settings.

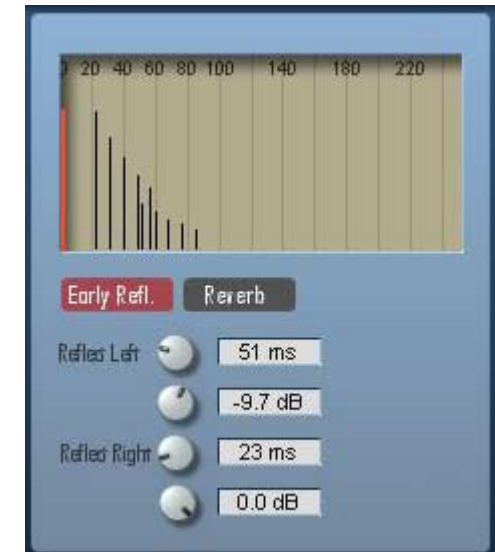
Reflect L (Reflections Left)

Sets the volume and delay for reflections in the left side of the stereo image.

Reflect R (Reflections Right)

Sets the volume and delay for reflections in the right side of the stereo image.

Depending on their adjustment, the reflections may sound very distinctly, as is appropriate for smaller rooms. For large rooms, using the reverb section alone may be sufficient.



Reverb

The following parameters control the behavior of the overall response, and are available in both versions of the reverb. In this section the tone quality is also determined by the settings of the filters mentioned previously.

Reverb

Sets the graphic display to show the reverb parameters.

Rev Delay (Rev Delay)

Adjusts the delay of the response in milliseconds. This is distinct from a Pre-delay as the early reflections or reflections are not affected by this setting.

The Reverberation Delay is used to separate the hall response from the direct signal and the early reflections. This is useful to increase the comprehensibility of vocals or speech. The impression of space is not altered, as the early reflections are not changed.

Diffuse

This setting controls the density of the response during the first milliseconds. For large rooms or halls, this should be set to maximum. When simulating smaller spaces, reducing the level results in added presence to the sound.

In the MasterVerb Classic version, this setting also affects the density of the reflections.

Room Size

Adjusts the apparent room size. This control applies only to the reverb portion of the signal.

In order to avoid interference, the response is muted for brief periods of time when adjusting the room size.

Time

Controls the reverb time (duration). The reverb time is unlimited, and can even be adjusted to infinity if desired.

As in a real-world acoustical environment, long reverb times correspond to large rooms. In small rooms, use shorter times to produce a natural sounding response.

HiDamp Filter (High Damp Filter)

This 6 db/octave filter operates on the reverb response. This filter reduces the high frequencies in the response signal depending on how it is adjusted. There are three ways you can adjust the cutoff frequency: use the rotary control; enter the value directly in the text field; or adjust the value directly in the graphic display area using the mouse.

Rooms or halls absorb high frequencies quite strongly. Therefore, settings between 3 kHz and 6 kHz are typical.



Shape

The Shape control lets you change the envelope of the response signal. Lower values correspond to relatively fast rise and fall times, while larger values result in slower rise and fall times. The effect is analogous to moving a wall, or raising or lowering the ceiling in a concert hall. The apparent size of the space changes correspondingly.

For percussive sounds like drums or other percussion instruments, smaller values for the Shape parameter are best.

Definition

The definition control adds response echoes similar to those produced in a room with smooth walls. This setting controls the "liveness" of the overall effect.

Dry

Controls the volume level of the original signal (without the effect).

Dry Mute

Switches off the original signal (only the effect signal will be heard).

ER (MasterVerb only)

Controls the volume level of the early reflections.

ER Mute (MasterVerb only)

Switches off the Early Reflections.

Reflect (MasterVerb Classic only)

Controls the volume level of the reflection signals.

Reflect Mute (MasterVerb Classic only)

Switches the reflection signals off.

Reverb

Adjusts the proportion of the response signal.

Reverb Mute

Switches the response signal off.

Wet

Controls the volume level of the overall effect signal.

Creating Reverb Presets

Creating good presets with a reverb processor that offers so many parameters requires a good ear and some experience. To help you along, we offer some tips for creating your own presets.

It makes good sense to start by using the supplied presets as a basis from which to create your own. The presets are arranged according to categories. Using suitable test material, listen to the presets while paying attention to the adjustments of the various parameters. Gradually familiarize yourself with the effects of the parameters by adjusting them, one at a time, while listening to the results.

When you become relatively comfortable with the parameters, you can begin to create your own presets. The following section describes a proven methodology, particularly appropriate for the MasterVerb.

First listen to the original signal alone by clicking the mute buttons for Early Reflections and Reverb. After you have listened carefully to the source signal, switch off muting for the Early Reflections signal and select a model corresponding to your basic idea of the room desired. Next, adjust the early reflections until they produce the desired room effect more exactly. Make several comparisons between the original signal and the processed signal as you proceed.

After you have established suitable settings for early reflections, switch in the reverb (response). Adjust the room size of the reverb signal so that the character of the reverb signal matches the character of the early reflections. Next, adjust the reverb time. Here the "less is more" principle applies. This is especially true when you get to the final mix, in which reverb times that are too long can produce disturbing results. Finally, fine-tune the effect with the Diffuse, Shape, and Definition controls.

With a little practice you'll get consistently good results without too much effort. You can also begin by working with the MasterVerb Classic. In this case, start by adjusting the reverb parameters, and adding reflections only if they improve the spatial context you are going for. Reflections in the MasterVerb Classic are not always required.

E-Reflector

This effect simulates the initial echoes (early reflections) of acoustical spaces, rooms or halls. As in the real world, the impression of space results from the combined effect of many individual echoes. Overlaying these signals results in phase cancellations that provide the distinctive character of that particular environment. As well as real spaces, experimental or theoretical examples are also provided. Filters in the input section can be used to change the tone quality of the signals.



Input Gain

Controls the level of the input signal. A level meter serves to help you optimize the level. The small dot indicator next to the Gain control indicates unity gain. If the control is adjusted below the indicator, the signal is attenuated. Above the indicator, the input signal is amplified.

Lowpass Filter

A low pass filter with a slope of 12db/octave follows the gain control. Adjust the frequency of this filter a) with the rotary control, b) by entering the value in the text field, or c) directly with the filter's graphic control interface.

Highpass Filter

After the low pass filter lies a high pass filter with a slope of 12db/octave. Adjust the frequency of this filter a) with the rotary control, b) by entering the value in the text field, or c) directly with the filter's graphic control interface.

Early Reflections

Early reflections are composed of sixteen individual echoes. You can control the nature and shape of these echoes with the following parameters. The filters described previously determine the overall tone quality of early reflections.

ER Type (Early Reflections Type)

From the drop-down list select an Early Reflections model (type). This determines the overall impression of this acoustical space.

ER Spread (Early Reflections Spread)

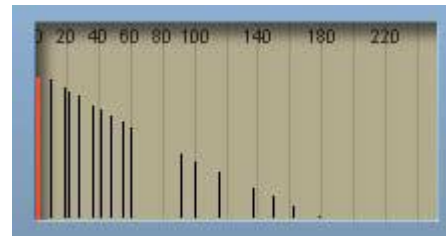
Adjusts the size of the apparent space.

ER Decay (Early Reflections Decay)

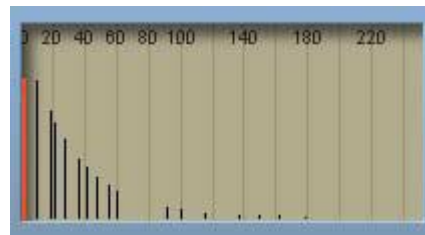
Controls how quickly the early reflections die away (that is, the degree of attenuation of the signal with increasing delay). For natural-sounding effects set the ER Decay to maximum for a quick decay.

ER Shape (Early Reflections Shape)

This parameter also controls the way the early reflections die out - in this case by adjusting the shape of the envelope curve of the signal decay. For natural-sounding effects use an exponential curve by setting ER Shape to its maximum value.



Linear envelope

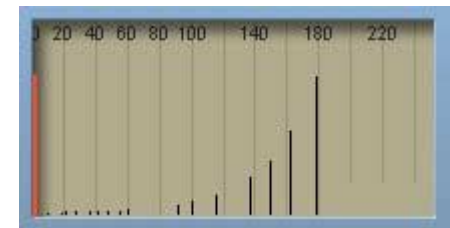


Exponential envelope

Invert

When selected, the gains of the individual reflections are reflected around the center point. That is, the Early Reflections model remains the same, but the envelope is inverted.

This acoustic scenario does not exist in nature. However, as a special effect it is very interesting and useful.



Inverted cluster

Compressor M/S

This effect is provided in both mono and stereo versions. The operation and functionality is identical in each version. A Compressor modifies the dynamics of a sound. The level of louder passages is decreased. This means that the overall level of the sound can be set higher, with the net result that the level of softer sections is increased. The compressor operates by monitoring the level of the input signal. An adjustable threshold level determines the point at which compression begins. Attack and release controls determine how quickly the compressor responds when the threshold is exceeded and when the input signal level falls back below the threshold, respectively. The ratio control determines the intensity of the compression – i.e., the ratio of input level change to output level change. The gain control adjusts the level of the compressed signal. The *Side Chain In* can be used to permit a signal other than the primary input signal to be used for analysis and control. The dynamics of the signal appearing at the Side Chain input will then control the response of the primary audio signal being processed.

Displays

In

Displays the input signal level.

Red (Reduction)

Shows the degree of reduction or attenuation of the signal.

Out

Displays the level of the resulting output signal.



Controls

Side Chain

Switching this on enables the Side Chain input. When it is enabled, the audio signal is controlled dynamically by the signal connected to the side chain input. Look Ahead cannot be used in this mode.

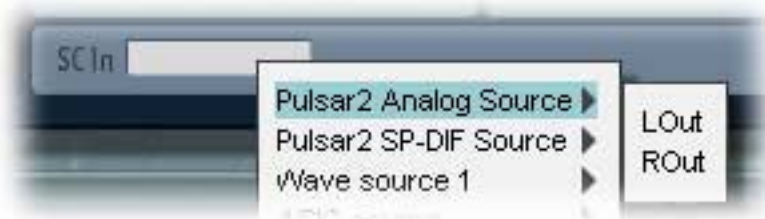
The Side Chain display field can be used to assign a signal. Right-click on the field (Mac: Ctrl-Click) to open a context menu showing all available source signals, then click normally on one of these to select it.

In XTC mode, the Side Chain functions are not accessible, since the Side Chains cannot be utilized by hard disk recording software.

LA (Look Ahead)

With Look Ahead enabled, the input signal being compressed is analyzed more accurately. The compressor uses a little more time to perform its function, but the resulting treatment is more precise. This implies, however, that the overall signal is somewhat delayed, and this must be considered in the mix. Maximum LookAhead times of 4ms and 16ms are available. <Off> disables LookAhead completely.

Keep in mind that LookAhead requires DSP memory. You should always use the shortest LookAhead time that achieves the desired results and disable LookAhead when not needed.



Attack

The Attack time is the compressor's reaction time - the time it takes it to respond to a level spike.

Release

This is the time after the signal falls back under the threshold that compression is no longer active.

Threshold

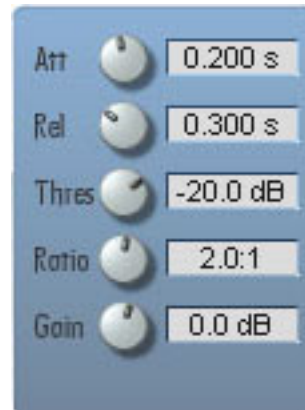
Sets the input signal level above which compression begins.

Ratio

The Ratio adjusts the compression rate for signals that exceed the threshold level. The compression is displayed as a relation value. 1:1 means that there is no compression. 3:1 e.g. means that a an amplification of +3dB of the input signal results only in +1dB at the output.

Gain

A compressor reduces the transient levels of a signal so that the overall signal can be increased later without danger of distortion. This increases the average level of the music, resulting in a fuller sound. Adjust the volume increase with the Gain control.



Ducker M/S

The Ducker is a special-purpose compressor. Both a mono and a stereo version are provided. Separate signals are fed to the main and side-chain inputs and appear mixed at the Ducker output. The level of the side-chain signal remains constant, while the other signal is attenuated according to level variations in the side-chain signal. The familiar Attack, Release and Threshold controls adjust the speed and intensity of the attenuation. One of the classic applications of the Ducker is in radio broadcast, where the level of music is automatically decreased whenever the announcer speaks (music and announcer's voice are fed to the main and side-chain inputs, respectively).

Displays

In

Displays the input signal level.

Red (Reduction)

Shows the degree of reduction or attenuation of the signal.

Out

Displays the level of the resulting output signal.



Because Side Chains cannot be utilized by hard disk recording software, this effect is not available in XTC mode.

Attack

Adjusts the amount of time over which an input signal is faded down when a side-chain signal crosses the threshold.

Release

Adjusts the amount of time over which an input signal is faded back up when the side-chain signal falls back below the threshold.

Threshold

Sets the side-chain signal level at which attenuation of the input signal begins. As the side-chain signal level increases further, the input signal is increasingly attenuated according to the setting of the **Ratio** control.

Ratio

Controls ducking "intensity". Sets the rate of increase of input signal attenuation as the side-chain signal level increases beyond the level specified by the **Threshold** setting. At the maximum setting of inf:1, the input signal is attenuated by exactly the amount by which the side-chain signal level exceeds the threshold. For example, with **Threshold** set to -30 dB, the input signal

will be attenuated by a maximum of 30 dB when the side-chain signal level is at its maximum level of 0 dB, with correspondingly less attenuation at lower side-chain signal levels. Intermediate **Ratio** settings yield smaller maximum attenuation values – at the minimum setting of 1:1, no attenuation occurs and the input signal comes through at its full actual level.

Bypass

The Bypass switch serves here as a "side-chain solo" switch – the side chain signal is heard directly.



Limiter M/S

This effect is related to compression and likewise modifies the dynamics of a sound. It is provided in both a mono and a stereo version. The level of louder passages is decreased. This means that the overall level of the sound can be set higher, with the net result that the level of softer sections is increased. The limiter operates by monitoring the level of the input signal. An adjustable threshold level determines the point at which limiting begins. Attack and release controls determine how quickly the limiter responds when the threshold is exceeded and when the input signal level falls back below the threshold, respectively. The ratio control determines the intensity of the limiting – i.e., the ration of input level change to output level change. The gain control adjusts the level of the limited signal. The *Side Chain In* can be used to permit a signal other than the primary input signal to be used for analysis and control. The dynamics of the signal appearing at the Side Chain input will then control the response of the primary audio signal being processed.

Displays

In

Displays the input signal level.

Red (Reduction)

Shows the degree of reduction, or attenuation of the signal.

Out

Displays the level of the resulting output signal.



Controls

Side Chain

Switching this on enables the Side Chain input. When it is enabled, the audio signal is controlled dynamically by the signal connected to the side chain input. Look Ahead cannot be used in this mode.

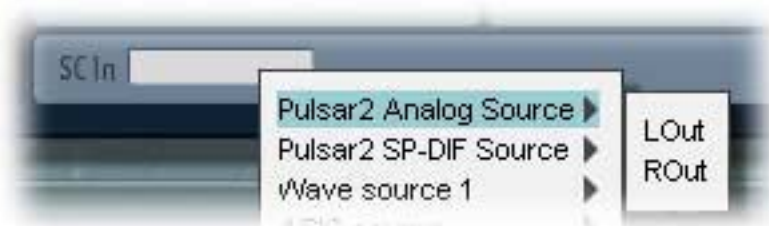
The Side Chain display field can be used to assign a signal. Right-click on the field (Mac: Ctrl-Click) to open a context menu showing all available source signals, then click normally on one of these to select it.

In XTC mode, the Side Chain functions are not accessible, since the Side Chains cannot be utilized by hard disk recording software.

LA (Look Ahead)

With Look Ahead enabled, the input signal being limited is analyzed more accurately. The limiter uses a little more time to perform its function, but the resulting treatment is more precise. This implies, however, that the overall signal is somewhat delayed, and this must be considered in the mix. Maximum LookAhead times of 4ms and 16ms are available. <Off> disables LookAhead completely.

Keep in mind that LookAhead requires DSP memory. You should always use the shortest LookAhead time that achieves the desired results and disable LookAhead when not needed.



Attack

The Attack time (in milliseconds) is the limiter's reaction time - the time it takes it to respond to a level spike.

Release

This is the time (in milliseconds) after the signal falls back under the threshold that limiting is no longer active.

Threshold

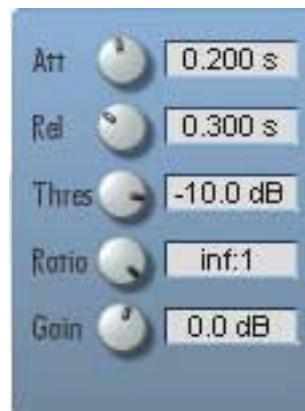
Sets the input signal level above which limiting begins.

Ratio

The Ratio adjusts the compression rate for signals that exceed the threshold level. The compression is displayed as a relation value. 1:1 means that there is no compression. 3:1 e.g. means that a an amplification of +3dB of the input signal results only in +1dB at the output.

Gain

A limiter reduces the transient levels of a signal so that the overall signal can be increased later without danger of distortion. This increases the average level of the music, resulting in a fuller sound. Adjust the volume increase with the Gain control.



Gate M/S

A gate is just what its name says - a gate. When it is open, signals can pass freely. When closed, signals are blocked. A typical gate will open when a specific volume threshold is reached, and remain open for a certain specified time. If the threshold is not reached again during that time, it will close. A gate can serve several purposes, such as muting the signal to avoid background noise (noise gate, instrument separation etc.) or to clean up the trailing out of an instrument sound. The *Side Chain In* can be used to permit a signal other than the primary input signal to be used for analysis and control. The level of the side chain input signal controls when the gate opens and closes.

Displays

In

Displays the input level.

Red (Reduction)

The relative signal reduction is displayed showing the actual working of the Gate.

Out

Displays the output level.



Controls

Side Chain

Switching this on enables the Side Chain input. When it is enabled, the audio signal is controlled dynamically by the signal connected to the side chain input. Look Ahead cannot be used in this mode.

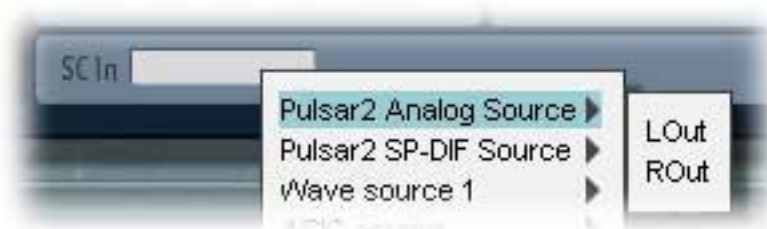
The Side Chain display field can be used to assign a signal. Right-click on the field (Mac: Ctrl-Click) to open a context menu showing all available source signals, then click normally on one of these to select it.

In XTC mode, the Side Chain functions are not accessible, since the Side Chains cannot be utilized by hard disk recording software.

LA (Look Ahead)

With Look Ahead enabled, the input signal being limited is analyzed more accurately. The limiter uses a little more time to perform its function, but the resulting treatment is more precise. This implies, however, that the overall signal is somewhat delayed, and this must be considered in the mix. Maximum LookAhead times of 4ms and 16ms are available. <Off> disables LookAhead completely.

Keep in mind that LookAhead requires DSP memory. You should always use the shortest LookAhead time that achieves the desired results and disable LookAhead when not needed.



Att (Attack)

Amount of time for full opening of the gate once the input signal level has exceeded the turn-on (upper) threshold.

Hold

Minimum amount of time the gate will stay open once it is triggered (i.e., minimum delay between end of attack and start of release).

Rel (Release)

Amount of time which gate takes to fully close once it begins to close (i.e., once the input signal level falls below the turn-off (lower) threshold, and after any remaining hold time has elapsed).

Threshold

Sets the input signal level in dB above which the gate will open. This is the turn-on (upper) threshold, shown in the display as a yellow point. The turn-off (lower) threshold is adjusted in tandem with the turn-on threshold, with an offset determined by the Hyst (hysteresis) control. With the Stereo Gate, opening of the gate is determined by the louder of the two channels.

Hyst (Hysteresis)

Adjusts the difference between the turn-on (upper) and turn-off (lower) thresholds. The turn-off threshold appears as a red point in the display. It can be set up to 10 dB below the turn-on threshold.

Floor

Sets the level to which gate gain drops when the gate closes. The gate can thus be set so that it does not close completely.

Gain

With the Gain control you can increase the Gate output level by up to 18dB.



Expander M/S

The Expander influences the dynamics of a sound by making quiet passages even quieter while leaving other passages alone. Thus the overall dynamic range of the signal increases. You can use this to alter the way an instrument sound decays - for example, to alter a looped drum pattern as it fades out, or to blend undesirable low-level background noise into the signal noise floor. The Expander is provided in both mono and stereo versions. The Threshold sets the level at which the effect begins to process the signal. The Attack and Release controls determine how quickly the effect engages or quits when the threshold is crossed. Ratio controls the degree of expansion - how much the level is changed by the effect. The Ratio value indicates the relation of the original signal to the expanded signal. The *Side Chain In* can be used to permit a signal other than the primary input signal to be used for analysis and control. The dynamics of the signal appearing at the Side Chain input will then control the response of the primary audio signal being processed. Displays

Displays

In

Displays the input level.

Red (Reduction)

The relative signal reduction is displayed showing the actual working of the expander.

Out

Displays the output level.



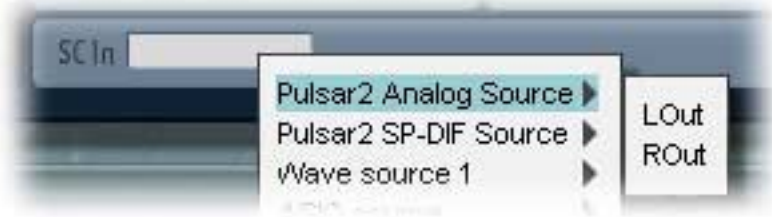
Controls

Side Chain

Switching this on enables the Side Chain input. When it is enabled, the audio signal is controlled dynamically by the signal connected to the side chain input. Look Ahead cannot be used in this mode.

The Side Chain display field can be used to assign a signal. Right-click on the field (Mac: Ctrl-Click) to open a context menu showing all available source signals, then click normally on one of these to select it.

In XTC mode, the Side Chain functions are not accessible, since the Side Chains cannot be utilized by hard disk recording software.



LA (Look Ahead)

With Look Ahead enabled, the input signal being limited is analyzed more accurately. The expander uses a little more time to perform its function, but the resulting treatment is more precise. This implies, however, that the overall signal is somewhat delayed, and this must be considered in the mix. Maximum LookAhead times of 4ms and 16ms are available. <Off> disables LookAhead completely.

Keep in mind that LookAhead requires DSP memory. You should always use the shortest LookAhead time that achieves the desired results and disable LookAhead when not needed.



Attack

The Attack time (in milliseconds) is the expanders's reaction time - the time it takes it to respond when the level falls below the threshold.

Release

This is the time (in milliseconds) after the signal raises over the threshold that expanding is no longer active.

Threshold

Sets the input signal level below which expanding begins.

Ratio

The Ratio adjusts the compression rate for signals that falls below the threshold level. The compression is displayed as a relation value. 1:1 means that there is no expansion. 3:1 e.g. means that a an attenuation of -1dB of the input signal results in -3dB at the output. The maximum Ratio is 10:1.

Gain

The output level of the expander can be adjusted and amplified up to +18dB.

De-Esser

The De-Esser can eliminate excessive sibilance from voice signals. Only the undesirable portion of the signal is affected.

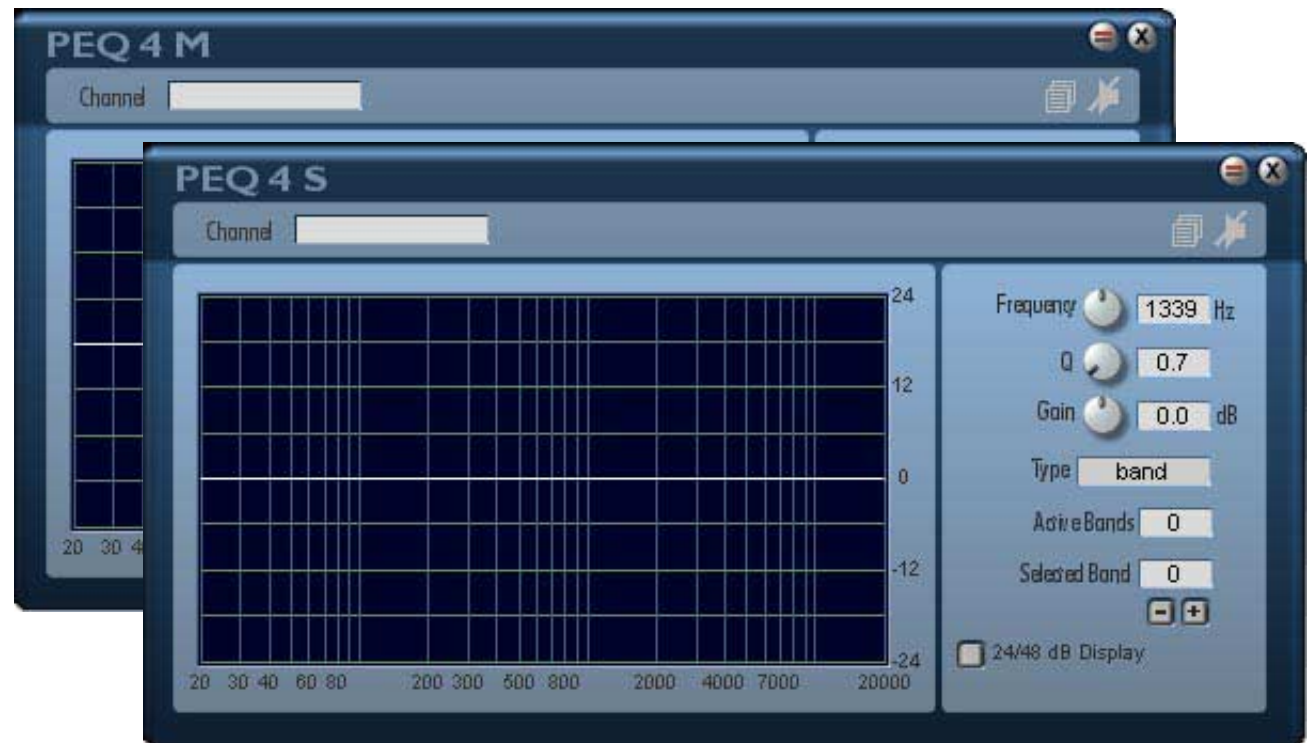


Drive

The De-Esser automatically recognizes sibilance signals and therefore requires only a single control. Drive sets the intensity of the filtering effect. At extreme settings, even normal "S" sounds may be strongly filtered, lending a lisping character to spoken words. If this occurs, decrease the Drive setting slightly.

PEQ 4 M/S

This parametric equalizer offers four bands, each of which can be configured as a different filter type. Depending on the type of filter, each band provides adjustable frequency, Q and gain. The equalizer is provided in both mono and stereo versions.



Graphic Control

Inserting or Deleting a Band

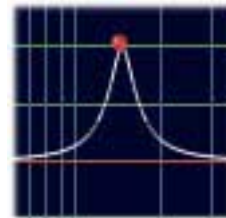
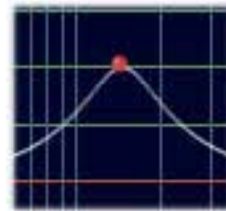
To add a new band, double click in the frequency and level graphic control area. To delete a band, double click on the existing node that represents the desired band.

Adjusting a Band's Frequency and Cut or Boost Level

To adjust a band's frequency or level, click and hold on the red node of the desired band. Drag it to the left or right to change the frequency, or up or down to adjust the cut or boost.

Adjusting the Filter Quality (Q Factor)

To adjust a band's Q factor, right click on the desired node and, while holding the mouse button, drag the mouse cursor up or down. The filter slope will adjust accordingly.



Frequency

Adjusts the frequency of a filter band's cutoff or center frequency.

Q

Sets the quality, or Q factor, for a filter band. This value determines the range of frequencies around a center frequency the filter will act on.

Gain

Adjusts the filter band's cut or boost level in dB.

Type

Use this text fader to select the type of filter to implement for the selected band. The following filter types are available: low pass, high pass, low shelving, high shelving, notch, and band pass.

Note: Not all parameters apply to all filter types. Low and high pass filters have only a frequency parameter. The shelving filters are adjustable for frequency and gain. Frequency and Q are the only parameters available for a notch filter. Only the band pass filter type offers all the adjustable parameters.

Active Bands

Indicates the number of currently active bands, to a maximum number of four.

Selected Band

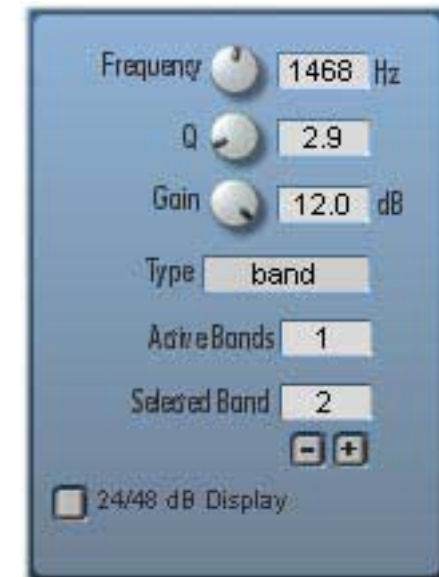
Indicates which band is currently selected. Only the currently selected band can be adjusted using the available controls.

24/48 dB Display

Switches the display resolution between 24 dB and 48 dB. The resolution generally depends on the number of bands in use. For up to two bands, use a resolution of 24 dB. Otherwise use 48 dB. The indicator is lit when 48 dB has been selected.

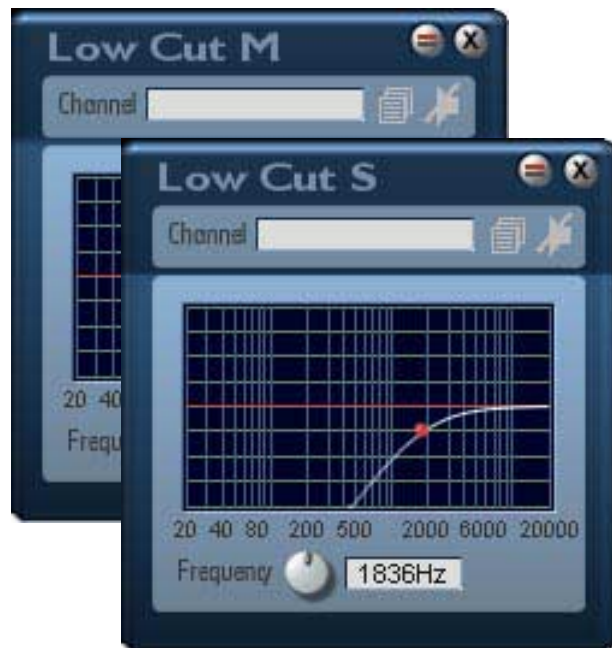
Bypass

Sends the input signal directly to the output, bypassing the effect.



Low Cut M/S

This is a lowpass filter with a 12dB/octave cutoff slope. The filter is provided in both mono and stereo versions.



Frequency

Sets the filter's cutoff frequency.

High Cut M/S

This is a highpass filter with a 12dB/octave cutoff slope. The filter is provided in both mono and stereo versions.



Frequency

Sets the filter's cutoff frequency.

Delay LM/M and LS/S

The Delay or Echo delays the signal in simple or more complex ways producing individual or repeated echos. Repeated echos are created using a feedback. Filters are included in the feedback loop to damp the high or low frequencies of subsequent echos. This effect is provided in versions with large delays (maximum delay time of 5460 ms), or small delays (maximum of 682 ms.). The stereo versions of the effect also offer the option of switching to a cross feedback mode.



ms/BPM Mode

Switches between **ms Mode** (delay time setting in milliseconds) and **BPM Mode** (delay time setting in terms of tempo).



BPM (BPM-Mode)

Specify the tempo you wish to match (available range: 25 .. 300 BPM).

Note (BPM-Mode)

Set delay time in terms of a musical note length relative to the **BPM** (tempo) setting. **P** and **T** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is 1/64T. The longest attainable note value depends upon the tempo – a slower tempo limits the delay to smaller maximum note values. If adjustment of **BPM** causes the maximum delay time (682 ms for Delay M/S, 5460 ms for Delay LM/LS) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Delay (ms-Mode)

Sets delay time directly in milliseconds. The minimum delay setting is 4 ms, the maximum 682 ms (Delay M/S) or 5460 ms (Delay LM/LS).

FB (Feedback)

Sets the strength of the feedback - the portion of the delayed signal that is routed back to the input to create multiple receding echos. The degree of feedback determines the fade out time of the repeated echoes.



Cross FB (Cross Feedback, only on Delay S and Delay LS)

Cross feedback, when enabled, feeds the signal from the left channel to the right channel to be delayed, and the signal from the right channel back to the left. The signal path thus forms a figure eight. When the button is lit, Cross Feedback is enabled.

LDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop. Low-frequency damping removes bass tones from the signal with each repeated cycle through the delay.



HDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop. High-frequency damping "softens" the signal with each repeat, creating a natural sounding fade.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the effect signal.

Delay LCR LS and S

The LCR Delays produce delayed signals on the left and right channels, and also in the center. The delay is adjustable by channel (Left/Right/Center) and a feedback loop is included for repeated echo. Filters are included in the feedback loop to damp the high or low frequencies of subsequent echos. This effect is provided in versions with large delays (maximum delay time of 5460 ms), or small delays (maximum of 682 ms.).



ms/BPM Mode

Switches between **ms Mode** (delay time setting in milliseconds) and **BPM Mode** (delay time setting in terms of tempo).



BPM (BPM-Mode)

Specify the tempo you wish to match (available range: 25 .. 300 BPM).

Note (BPM-Mode)

Set delay time in terms of a musical note length relative to the **BPM** (tempo) setting. **P** and **T** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is 1/64T. The longest attainable note value depends upon the tempo – a slower tempo limits the delay to smaller maximum note values. If adjustment of **BPM** causes the maximum delay time (682 ms for Delay LCR S, 5460 ms for Delay LCR LS) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Delay (ms-Mode)

Sets delay time directly in milliseconds. The minimum delay setting is 4 ms, the maximum 682 ms (Delay LCR S) or 5460 ms (Delay LCR LS).

FB (Feedback)

Sets the strength of the feedback - the portion of the delayed signal that is routed back to the input to create multiple receding echos. The degree of feedback determines the fade out time of the repeated echos.

Spread

This control adjusts the degree to which the stereo channels are separated in the stereo image. The center channel is always in the center, while the left and right signals are distributed proportionately in the left and right channels depending on this setting.

LDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop. Low-frequency damping removes bass tones from the signal with each repeated cycle through the delay.

HDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop. High-frequency damping "softens" the signal with each repeat, creating a natural sounding fade.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the effect signal.

Dual Delay LS and S

This delay provides the left and right channels with their own independent feedback loops. Filters are included in the feedback loop to attenuate the high or low frequencies of subsequent echos. This effect is provided in versions with large delays (maximum delay time of 5460 ms), or small delays (maximum of 682 ms).



ms/BPM Mode

Switches between **ms Mode** (delay time setting in milliseconds) and **BPM Mode** (delay time setting in terms of tempo).

BPM (BPM-Mode)

Specify the tempo you wish to match (available range: 25 .. 300 BPM).

Note (BPM-Mode)

Set delay time in terms of a musical note length relative to the **BPM** (tempo) setting. **P** and **T** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is $1/64T$. The longest attainable note value depends upon the tempo – a slower tempo limits the delay to smaller maximum note values. If adjustment of **BPM** causes the maximum delay time (682 ms for Dual Delay S, 5460 ms for Dual Delay LS) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Delay (ms-Mode)

Sets delay time directly in milliseconds. The minimum delay setting is 4 ms, the maximum 682 ms (Dual Delay S) or 5460 ms (Dual Delay LS).

FB L/R (Feedback)

This controls, for each channel, the amount of the delayed signal which will

be fed back to the input to be delayed again. Simply put, the higher this setting, the more echoes you will get.

LDamp L/R (Low Damp)

Use this control to adjust the amount of low frequency damping in the feedback loop for the respective channel.

HDamp (High Damp)

Use this control to adjust the amount of high frequency damping in the feedback loop for the respective channel. High-frequency damping "softens" the signal with each repeat, creating a natural sounding fade.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the effect signal.

Multitap M/S

The Multitap provides four delays with independently adjustable volume, and, in the stereo version, adjustable pan position as well. The delay is adjustable for each tap, and a feedback loop is inserted after delay 1 to produce recurring echos. Filters are included in the feedback loop to attenuate the high or low frequencies of subsequent echos. The maximum delay time for each tap is 682 ms.

Note (BPM-Mode)

Set delay time in terms of a musical note length relative to the **BPM** (tempo) setting. **P** and **T** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is 1/64T. The longest attainable note value depends upon the tempo – a slower tempo limits the delay to smaller maximum note values. If adjustment of **BPM** causes the maximum delay time (682 ms) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Delay 1-4 (ms-Mode)

Sets delay time directly in milliseconds for each of the four taps. The minimum delay setting is 4 ms, the maximum 682 ms.

FB (Feedback)

Controls the amount of the delayed signal to be fed back to the input to be delayed again. By skillfully setting individual delay times for each tap, interesting rhythmic effects can be produced when feedback is introduced.



ms/BPM Mode

Switches between **ms Mode** (delay time setting in milliseconds) and **BPM Mode** (delay time setting in terms of tempo).



BPM (BPM-Mode)

Specify the tempo you wish to match (available range: 25 .. 300 BPM).

Level 1-4

Adjusts the volume level of each individual tap. Set this to 0 to omit the tap.

Pan 1-4 (only on the Multitap S)

Sets the pan position in the stereo field for each of the four taps

LDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop. Low-frequency damping removes bass tones from the signal with each repeated cycle through the delay.

HDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop. High-frequency damping "softens" the signal with each repeat, creating a natural sounding fade.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the effect signal.

Ducking Delay M/S

The Ducking Delay unites a Ducker and a Delay in a single effect which is especially useful for processing vocals (but can of course be used with instruments as well). The main and side-chain inputs of the Ducker are connected to the delay output and the dry signal, respectively. During vocal passages, the delay effect is partially or entirely faded out, and returns during the pauses between vocal passages. The familiar Attack, Release and Threshold controls adjust the speed and intensity of the attenuation. Both a mono and a stereo version are provided.



Displays

In

Displays the input signal level.

Red (Reduction)

Shows the degree of reduction, or attenuation of the delay signal.

Out

Displays the level of the resulting output signal.



Controls

ms/BPM Mode

Switches between **ms Mode** (delay time setting in milliseconds) and **BPM Mode** (delay time setting in terms of tempo).

BPM (BPM-Mode)

Specify the tempo you wish to match (available range: 25 .. 300 BPM).

Note (BPM-Mode)

Set delay time in terms of a musical note length relative to the **BPM** (tempo) setting. **P** and **T** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is 1/64T. The longest attainable note value depends upon the tempo – a slower tempo limits the delay to smaller maximum note values. If adjustment of **BPM** causes the maximum delay time (682 ms) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Delay (ms-Mode)

Sets delay time directly in milliseconds. The minimum delay setting is 4 ms, the maximum 682 ms.

FB (Feedback)

Sets the strength of the feedback - the portion of the delayed signal that is routed back to the input to create multiple receding echos. The degree of feedback determines the fade out time of the repeated echos.

Cross FB (Cross Feedback, only on Ducking Delay S)

Cross feedback, when enabled, feeds the signal from the left channel to the right channel to be delayed, and the signal from the right channel back to the left. The signal path thus forms a figure eight. When the button is lit, Cross Feedback is enabled.

LDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop. Low-frequency damping removes bass tones from the signal with each repeated cycle through the delay.

HDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop. High-frequency damping "softens" the signal with each repeat, creating a natural sounding fade.

Attack

Adjusts the amount of time over which the delay output signal is faded down when the input signal crosses the threshold.

Release

Adjusts the amount of time over which the delay output signal is faded back up when the input signal level falls back below the threshold.

Threshold

Sets the side-chain signal level at which attenuation of the delay output begins. As the side-chain signal level increases further, the delay output is increasingly attenuated according to the setting of the **Ratio** control.

Ratio

Controls ducking "intensity". Sets the rate of increase of delay output attenuation as the side-chain signal level increases beyond the level specified by the **Threshold** setting. At the maximum setting of inf:1, the delay output is attenuated by exactly the amount by which the side-chain signal level exceeds the threshold. For example, with

Threshold set to -30 dB, the delay output will be attenuated by a maximum of 30 dB when the side-chain signal level is at its maximum level of 0 dB, with correspondingly less attenuation at lower side-chain signal levels. Intermediate **Ratio** settings yield smaller maximum attenuation values – at the minimum setting of 1:1, no attenuation occurs and the delay output signal comes through at its full actual level.

Ducking On/Off

Deactivates the Ducker, so that a normal delay effect is obtained.

Dry

Adjusts the level of the original signal.

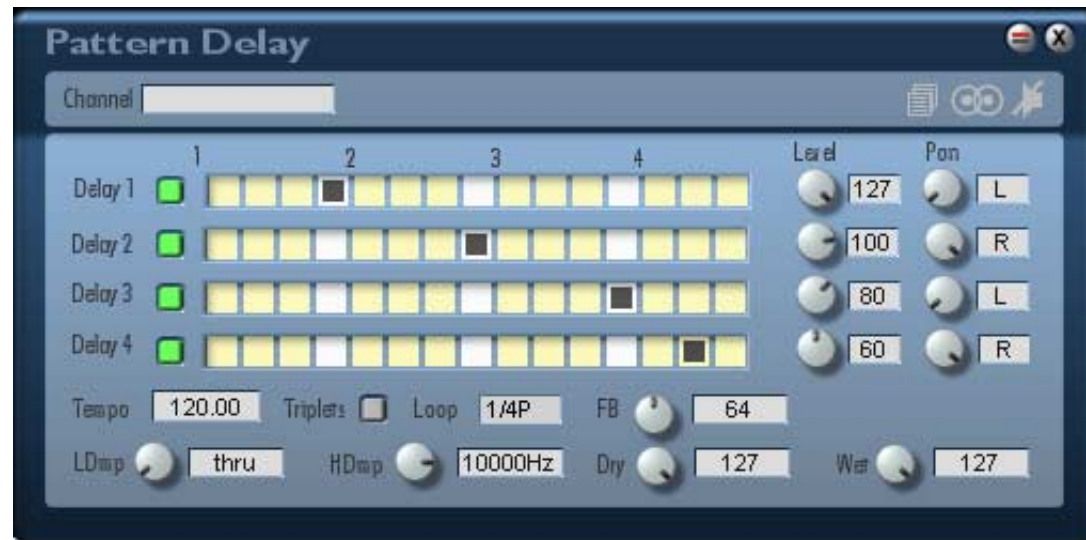
Wet

Adjusts the level of the effect signal.



Pattern Delay

This delay permits easy creation of rhythmic delays and loops. Delay times are set via a pattern editor which displays the 16th-intervals of a bar (or the triplet version of same). The pattern is repeated over the course of the interval specified under Loop. Volume level and pan position can be set per delay. The low-pass and high-pass filters provided are located within the pattern feedback path. The effect functions in stereo.



Loop

Sets the time interval over which the overall pattern is repeated or restarted. This setting is specified in terms of note-length values and can be shorter than the pattern itself if desired. In fact, this control is the delay time setting for a

separate, "hidden" looping delay around which the feedback occurs. The output signal from this delay, combined with the dry input signal, is fed to the inputs of pattern delays 1-4, whose lengths can therefore be longer than the Loop setting.

Delay On/Off 1-4

Each button switches the corresponding delay tap on or off. Switching a delay off removes it from the pattern. A delay is active when the corresponding button is lit.

Delay Length 1-4

The delay time of a tap is specified in terms of note intervals which can be set within the displayed grid. The individual positions in the grid represent either 16ths or triplets, depending upon whether the Triplets option is selected (see below). The white boxes indicate quarter-note positions in either case.

Note that the outputs of delays 1-4 are not included in the feedback loop!

Triplets

Switches the pattern to triplets mode. The individual boxes in the pattern grid now represent triplets instead of "straight" sixteenths. Triplets is activated when the button is lit.

Level 1-4

These controls adjust the volume levels of the individual taps. Set a level control to zero to exclude the corresponding tap.

Pan 1-4

These controls adjust the pan positions of the individual taps.

Tempo

Enter any desired tempo between 25 and 300 BPM.

FB (Feedback)

Adjusts the amount of looping delay output signal which is fed back to the delay inputs, where it is again looped and repeated as a pattern. At typical feedback settings, the repeating echoes decrease in volume with each additional pass through the delay. When this control is set to maximum, signals fed into the delay line will be repeated infinitely.

LDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop. Low-frequency damping removes bass tones from the signal with each repeated cycle through the delay.

HDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop. High-frequency damping "softens" the signal with each repeat, creating a natural sounding fade.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the effect signal.

Decimator M/S

The Decimator lets you play a signal with a different bit resolution and sample rate as the rest of the system. In effect, the signal is internally resampled. Aliasing and quantization noise can be deliberately produced, depending upon the settings used. This module can be used polyphonically as well as monophonically.



Controls

Bit

Controls the number of bits which is used for the Bit quantization.

Bit on/off

Switches Bit quantization on or off. When lit, Bit is enabled.

Sample Rate

Controls the sample rate which is used for the conversion.

Sample Rate on/off

Switches Sample Rate on or off. When lit, Sample Rate is enabled.

Distortion M/S

This effect produces both "soft" or "hard" distortion. Soft distortion is comparable to that which results when analog magnetic tape overload occurs – it comes on slowly with increasing signal level and has a soft sound. Hard distortion is produced via simple clipping. It appears more suddenly and has a harder, edgier sound. This module can be used polyphonically as well as monophonically – thus both before and after the Poly Out module. Since the overall signal level increases when you use the Gain control to raise the distortion level, a *Level* control is provided to attenuate the signal for output.



Gain

Controls the amount of distortion. The level of the overall signal is also increased along with the distortion.

Hard

When Hard is enabled, the signal is digitally clipped (cut off). Hard is enabled when the button is lit.

Soft

This distortion effect emulates that of an overdriven analog tape machine. It is a gentler form of distortion than hard clipping. Soft is enabled when the button is lit.

Level

This control allows you to compensate for the increase in signal volume when distortion is added.

Overdrive M/S

This effect produces a classic overdrive distortion. A *Lowcut* filter is available to limit the distortion to higher frequencies. Since the overall signal level increases when you use the *Gain* control to raise the distortion level, a *Level* control is provided to attenuate the signal for output.



Gain

Controls the amount of distortion. The level of the overall signal is also increased along with the distortion.

LCut (Lowcut)

Use this control to limit the distortion to the higher frequencies.

Level

This control allows you to compensate for the increase in signal volume when distortion is added.

TDrive M/S and TQDrive M/S

This effect simulates the warm, rich sound of the distortion produced by a tube amplifier (hence the "T" in the name). Filters permit further tonal shaping. The TQDrive has one pre-distortion EQ and two post-distortion EQs. Both effects are provided in mono and stereo versions.



Drive

Controls the degree of distortion. The distortion circuit also boosts the signal level. The green and red LEDs next to the text field indicate whether a signal is present and whether digital clipping (overload) is occurring. Clipping can be avoided by adjusting **Level**.

Level

Permits level trimming to avoid digital clipping. If digital clipping occurs (as indicated by the red LEDs), turn **Level** down until only the green LEDs are lit. If reducing **Level** does not eliminate the clipping, reduce **Drive** slightly as well.

LCut (Lowcut)

Use this control to limit the distortion to the higher frequencies.

HDamp (High Damp)

Decreases the level of the high frequencies, producing a warmer, "rounder" distortion sound.

Spread (TDrive S and TQDrive S only)

A short delay to produce a broader stereo image.

Pre and Post-EQs (TQDrive M/S only)

Frequency

Adjusts the filter's center frequency.

Q

Sets the filter's quality, or Q factor. This determines the width of the frequency band which the filter will affect.

Gain

Adjusts the filter band's cut or boost level in dB.



AutoWah M/S

This effect combines the TDrive distortion effect (see preceding section) with a multimode filter whose cutoff frequency is controlled by an envelope follower. The envelope follower follows the level of the original (pre-distortion) signal, since the distortion effect "flattens out" the signal dynamics. When the filter resonance is turned up, the sounds produced resemble the words "wah-wah", hence the name of the effect. A mono and a stereo version of the effect are provided.

Drive

Controls the degree of distortion. The distortion circuit also boosts the signal level. The green and red LEDs next to the text field indicate whether a signal is present and whether digital clipping (overload) is occurring. Clipping can be avoided by adjusting **Level**.

Level

Permits level trimming to avoid digital clipping. If digital clipping occurs (as indicated by the red LEDs), turn **Level** down until only the green LEDs are lit. If reducing **Level** does not eliminate the clipping, reduce **Drive** slightly as well.

LCut (Lowcut)

Use this control to limit the distortion to the higher frequencies.

HDamp (High Damp)

Decreases the level of the high frequencies, producing a warmer, "rounder" distortion sound.



Envelope Follower

Gain

Some signal levels are too low or too high to be processed effectively. Adjust the **Gain** control up or down to optimize the signal level.

Attack

Controls the response time of the envelope follower to rising levels of the audio signal.

Decay

Controls the response time of the envelope follower to falling levels of the audio signal.

Filter

Filter Type

High-pass, low-pass and band-pass modes can be selected. The filter slope is 12 dB/octave in all modes.

Frequency

Sets filter cutoff or center frequency, with the current value displayed in Hz.

Resonance

Sets the amount of filter resonance. Values between 0 and 127 are displayed for reference.

Env Follower

Controls the amount of filter frequency modulation produced by the envelope follower. Negative values produce an inverse modulation.

Chorus S and Harmonic Chorus S

The name "chorus" hints at the sound produced by this effect. It spreads and thickens the sound passed through it, simulating the sound of multiple instruments of the same type playing together – in other words, a chorus. This effect is achieved by means of a short delay line whose delay time is periodically modulated. Mixing of this delayed signal with the original produces the chorus effect. The intensity of the effect depends upon the modulation rate, depth and phase settings as well as the dry/wet (original/delayed) mix. Adjustable feedback and cross feedback controls are also provided. The effect is also useful for creating a stereo sound from a monaural signal.

The Harmonic Chorus splits the signal into two frequency ranges and applies the effect only to frequencies above the split frequency.



Rate

Adjusts the frequency of delay time modulation in the chorus effect.

Depth

Adjusts the amount of delay time modulation in the chorus effect, and thus its intensity.

FB (Feedback)

Controls the level of the feedback signal which produces the comb filter effect, similar to flanging. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Cross FB (Cross Feedback)

When Cross FB is enabled, the left and right feedback signals are cross-routed to the right and left delay line inputs respectively. This creates comb filter effects which differ from those obtained with simple feedback. When the button is lit, Cross FB is enabled.

L/R Phase

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

Split F (Split Frequency, only on Harm. Chorus S)

Sets the frequency at which the original signal is split into two frequency ranges. Only the frequencies above the split frequency will be processed for the chorus effect.

LowL (Low Level, only on Harm. Chorus S)

Adjusts the volume level of the portion of the signal lying below the split frequency. This part of the signal is not processed for the chorus effect.

HighL (High Level, only on Harm. Chorus S)

Adjusts the volume level of the portion of the signal lying above the split frequency. This is the part of the signal that is processed for the chorus effect.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Master Chorus M/S

This sophisticated device offers options for modifying modulation and chorus tone color, in addition to the classic chorus parameters. It covers a wide range of sounds, from especially subtle chorusing to intense chorusing with feedback. Both mono and stereo versions are provided.



PreDel L/R

Adjusts the delay time over the range of 0 to 100 msec for both channels of the integrated stereo delay line which is connected in-line ahead of the chorus effect.

Waveform

Selects either a sine or triangle waveform for modulation of the chorus effect.

Shape

"Warps" the modulation waveform, widening the dips and narrowing the peaks to an adjustable degree, thereby altering the "motion" produced by the modulation.

Rate

Adjusts the frequency of delay time modulation in the chorus effect.

Depth

Adjusts the amount of delay time modulation in the chorus effect, and thus its intensity.

FB (Feedback)

Controls the level of the feedback signal which produces the comb filter effect, similar to flanging. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

L/R Phase (MasterChorus S only)

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

LoDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Hexa Chorus S

The Hexa Chorus taps the signal not just once, but six times, to provide multiple delayed signals for the chorus effect. Each signal can be modulated for a particularly full and rich sound.

The Hexa Chorus S also features an integrated stereo delay with independently adjustable delay times for each channel.



PreDel (Left/Right)

Adjusts the integrated delay for both of the stereo channels. The range is from 0 to 628 ms.

Rate

Adjusts the frequency of delay time modulation in the chorus effect.

Depth

Adjusts the amount of delay time modulation in the chorus effect, and thus its intensity.

Stereo Spread

Controls the distribution of the effect signals between the two stereo channels. At the minimum value, the signals are mixed and sent equally to each channel. At the maximum value, the signals are sent individually to each channel, creating a quasi stereo effect.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the signal after processing by the effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

4Tap Chorus S

This effect features four independent delays with separately adjustable delay time (0 - 682 ms), volume and pan position. A feedback path from the Delay 1 output can produce recurring echos. A filter is included in the feedback loop to attenuate the high frequency content of subsequent echos. Each delay is followed by a chorus effect.



Delay Time Tap 1-4

These controls adjust the delay for each of the four taps individually within a range of 0 - 682 ms.

Level 1-4

Level controls for the individual taps. Set a control to 0 to eliminate the corresponding tap from the effect.

Pan 1-4

Adjusts the position of each tap in the stereo field.

Feedback

Adjusts the amount of Delay 1 output signal being fed back to the input. By skillfully setting individual delay times for each tap, rhythmic effects can be produced when feedback is introduced.

High Damp

Adjusts the amount of high-frequency damping in the feedback loop. High-frequency damping "softens" the signal with each repeat, creating a natural sounding fade.

LFO

Selects between triangle (Tri) and sine as the LFO waveform for delay time modulation in the chorus effect.

Rate

Adjusts the frequency of delay time modulation in the chorus effect.

Depth

Adjusts the amount of delay time modulation in the chorus effect, and thus its intensity.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the signal after processing by the effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Flanger M/S and Harmonic Flanger S

This effect is similar to the chorus. Like the chorus, a flanger is based on a delay line whose delay time is periodically modulated. However, the delay times in a flanger are substantially shorter than those of a chorus. In addition, the flanger utilizes feedback of the delayed signal back to the delay line input. Therefore, it not only thickens the sound but can add noticeable coloration owing to the comb-filter effect which results from the feedback. The intensity of the effect depends upon the modulation rate, depth and phase settings as well as the dry/wet (original/delayed) mix. The effect is provided in both mono and stereo versions and is also useful for creating a stereo sound from a monaural signal.

The Harmonic Flanger splits the signal into two frequency ranges, and applies the effect only to the frequencies above the split frequency.



Rate

Adjusts the frequency of delay time modulation in the flanger effect.

Depth

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

FB (Feedback)

Controls the level of the feedback signal which produces the comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Cross FB (Cross Feedback, only on Flanger S and Harmonic Flanger S)

When Cross FB is enabled, the left and right feedback signals are cross-routed to the right and left delay line inputs respectively. This creates comb filter effects which differ from those obtained with simple feedback. When the button is lit, Cross FB is enabled.

L/R Phase (only on Flanger S and H. Flanger S)

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

Split F (Split Frequency, only on Harm. Flanger S)

Sets the frequency at which the original signal is split. Only the frequencies above the split frequency are processed via the flanger effect.

LowL (Low Level, only on HarmonicFlanger S)

Adjusts the volume level of the portion of the signal lying below the split frequency. This part of the signal is not processed via the flanger effect.

HighL (HighLevel, only on Harm. Chorus S)

Adjusts the volume level of the portion of the signal lying above the split frequency. This part of the signal is processed via the flanger effect.



Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the processed signal.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Master Flanger M/S

This sophisticated device offers options for modifying modulation and flanger tone color, in addition to the classic flanger parameters. It covers a wide range of sounds, from subtle flanging to intense flanging with feedback. Both mono and stereo versions are provided.



PreDel L/R

Adjusts the delay time over the range of 0 to 100 msec for both channels of the integrated stereo delay line which is connected in-line ahead of the flanger effect.

Waveform

Selects either a sine or triangle waveform for modulation of the flanger effect.

Shape

"Warps" the modulation waveform, widening the dips and narrowing the peaks to an adjustable degree, thereby altering the "motion" produced by the modulation.

Rate

Adjusts the frequency of delay time modulation in the flanger effect.

Depth

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

FB (Feedback)

Controls the level of the feedback signal which produces the typical flanger comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

L/R Phase (MasterFlanger S only)

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

LoDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Random Flanger M/S

The sound of this effect is similar to that of the Master Flanger. However, the modulation waveform is a random signal, resulting in an effect which varies continuously in intensity and character. Both mono and stereo versions are provided.



PreDel L/R

Adjusts the delay time over the range of 0 to 100 msec for both channels of the integrated stereo delay line which is connected in-line ahead of the flanger effect.

Waveform

Selects a sine, triangle or stepped waveform for modulation of the flanger effect. The amplitude of the selected waveform varies randomly.

Rate

Adjusts the frequency of delay time modulation in the flanger effect.

Depth

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

FB (Feedback)

Controls the level of the feedback signal which produces the typical flanger comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

L/R Phase (RandomFlanger S only)

Selects between in-phase (0°) and out-of-phase (180°) modulation signals for the left and right channels of the flanger effect. This influences the apparent "breadth" of the stereo image.

LoDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry

Adjusts the level of the original signal.

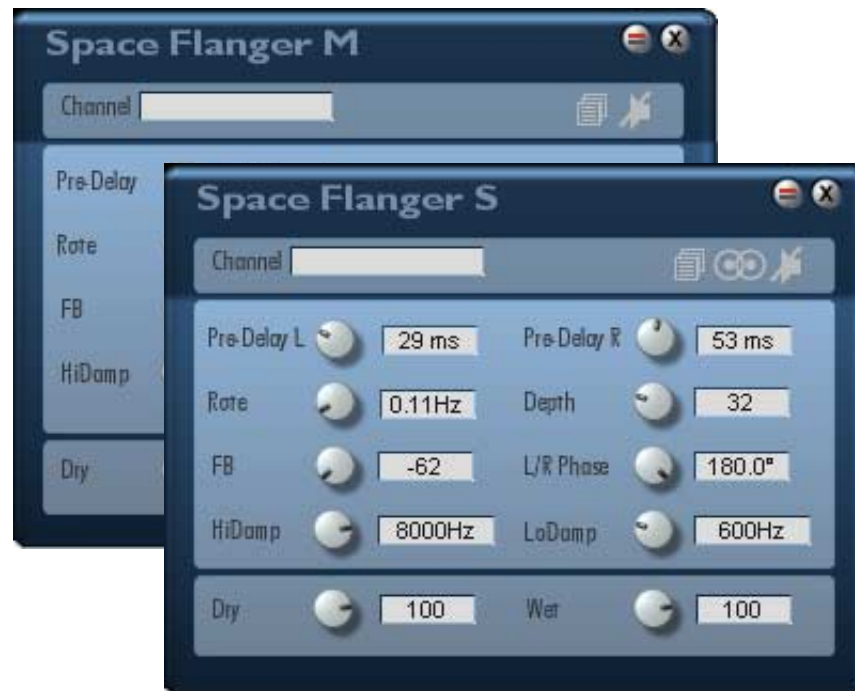
Wet

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Space Flanger M/S

In this flanger, not only the delay position is varied, but the delay length as well. This lends the Space Flanger a very distinctive character. Both mono and stereo versions are provided.



PreDel L/R

Adjusts the delay time over the range of 0 to 100 msec for both channels of the integrated stereo delay line which is connected in-line ahead of the flanger effect.

Waveform

Selects either a sine or triangle waveform for modulation of the flanger effect.

Shape

"Warp" the modulation waveform, widening the dips and narrowing the peaks to an adjustable degree, thereby altering the "motion" produced by the modulation.

Rate

Adjusts the frequency of delay time modulation in the flanger effect.

Depth

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

FB (Feedback)

Controls the level of the feedback signal which produces the typical flanger comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

L/R Phase (Space Flanger S only)

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

LoDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Step Flanger M/S

The Step Flanger applies a sample-and-hold circuit to the modulation waveform. The resulting flanger effects range from a step-wise movement of the comb filter through its spectrum to abrupt large jumps. Both mono and stereo versions are provided.



PreDel L/R

Adjusts the delay time over the range of 0 to 100 msec for both channels of the integrated stereo delay line which is connected in-line ahead of the flanger effect.

Waveform

Selects either a sine or triangle waveform for modulation of the flanger effect.

Shape

"Warps" the modulation waveform, widening the dips and narrowing the peaks to an adjustable degree, thereby altering the "motion" produced by the modulation.

Rate

Adjusts the frequency of delay time modulation in the flanger effect.

Depth

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

Step Rate

Adjusts the rate at which the modulation signal is "chopped" into steps by the sample-and-hold circuit. Specified as a multiple of the modulation frequency. For typical staircase effects, the step rate should be at least twice the modulation rate.

Step Lag

Adjusts the rate at which the modulation signal slews, or makes its transition, from one step value to the next. Larger settings produce more gradual transitions, while a setting of zero yields very hard transitions.

FB (Feedback)

Controls the level of the feedback signal which produces the typical flanger comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

L/R Phase (Step Flanger S only)

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

LoDamp (Low Damp)

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp)

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Phaser M/S

The phaser or phase-shifter is similar to the flanger and chorus, but instead of using a delay line, it functions by introducing frequency-dependent phase shifts into the signal which is passed through it. The key phase-shift frequencies, and therefore the amount of phase shift, are periodically modulated. The phase-shifter effect results when this phase-manipulated signal is mixed with the original signal. The effect is thus similar to the flanger and chorus, but with a sound character of its own. The intensity of the effect depends upon the modulation rate, depth and phase settings as well as the dry/wet (original/phase-shifted) mix. The effect is provided in both mono and stereo versions and is also useful for creating a stereo sound from a monaural signal.



Rate

Controls the frequency of the phase modulation of the Phaser.

Depth

Adjusts the modulation depth - the strength of the phase modulation.

Effects



FB (Feedback)

Controls the level of the feedback signal which intensifies the typical phasing effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the phaser accordingly.

Cross FB (only on Phaser S)

When Cross FB is enabled, the left and right feedback signals are cross-routed to the right and left phase filter inputs respectively. This creates effects which differ from those obtained with simple feedback. When the button is lit, Cross FB is enabled.

L/R Phase (only on Phaser S)

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the processed signal.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

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What is SSB?

SSB is an abbreviation for Single Side-Band and refers to a modulation process by means of which a signal's frequencies can be shifted. The shift is the same for all frequencies – for example, if a signal contains components at 440 Hz, 880 Hz, 1760 Hz and 3520 Hz, an SSB modulator can shift each of these frequencies by +10 Hz to 450 Hz, 890 Hz, 1770 Hz and 3530 Hz. Therefore, SSB is also referred to as frequency shifting or spectrum shifting. Until now, SSB has been available only in expensive modular systems such as the Moog Modular.

Frequency shifting is not to be confused with pitch shifting. In pitch shifting, all frequencies are multiplied by the same factor (or, in other words, transposed) and harmonic relationships are preserved.

With SSB, by contrast, harmonic relationships are as a rule transformed into inharmonic ones, as the above example illustrates: whereas 880 Hz is an octave above 440 Hz, 890 Hz is not an octave above 450 Hz – the shifted frequencies are no longer harmonically related.

SSB Phaser M/S

If SSB is used to shift a signal's frequency spectrum only slightly (by less than 1 Hz), and if the shifted signal is mixed with the original, an effect is produced which is not unlike a phaser, but which differs significantly from a phaser in the following respect: whereas a phaser creates phase-cancellations which move up and down through the frequency spectrum, those produced by an SSB phaser move only in one direction, according to the sign of the spectrum shift (plus or minus). Larger shifts produce spectra and sounds similar to those produced by ring modulation. Both mono and stereo versions of the SSB Phaser are provided.



Frequency Shift

Amount by which all frequencies in the signal spectrum are shifted. Both positive and negative shifts are possible.

Range

Multiplier for the Frequency Shift control – sets its effective range. 1.00000x permits a shift of ± 24000 Hz, while 0.00001x produces a range of ± 0.2400 Hz. By setting Range appropriately, Frequency Shift can be adjusted very finely.

Feedback

Feedback causes the processed signal to be processed over and over again. An already-shifted frequency gets shifted again and again by the same amount. With small frequency shift amounts, this intensifies the phasing effect, while with large shifts, generation of more inharmonic components is the result.

Dry On/Off

Switches the dry (unprocessed) signal on or off. Switch it off when you want to use only the frequency shifter. The dry signal is present when the button is lit.

Link Shifts (SSB Phaser S only)

When this function is activated, the frequency shift can be set simultaneously for both left and right signals.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the SSB effect signal.

SSB Modulator M/S

This effect is the same as the SSB Phaser, but with the additional ability to modulate the frequency shift via an LFO. Waveform, depth and rate of the LFO are adjustable. The LFO signal is summed with the Frequency Shift setting to produce the total shift. Both mono and stereo versions of the effect are provided.



Frequency Shift

Amount by which all frequencies in the signal spectrum are shifted. Both positive and negative shifts are possible.

LFO Depth

Sets the maximum amount of frequency shift produced by the LFO. Both positive and negative shifts are possible.

Range

Multiplier for the Frequency Shift control – sets its effective range. 1.00000x permits a shift of ± 24000 Hz, while 0.00001x produces a range of ± 0.2400 Hz. By setting Range appropriately, Frequency Shift can be adjusted very finely.

Feedback

Feedback causes the processed signal to be processed over and over again. An already-shifted frequency gets shifted again and again by the same amount. With small frequency shift amounts, this intensifies the phasing effect, while with large shifts, generation of more inharmonic components is the result.

LFO Wave

Sets the LFO waveform used for frequency shift modulation. Choices include Sine, Square, Saw Up, Saw Down, Triangle und Random.

Rate

Sets the rate of LFO frequency shift modulation.

Dry On/Off

Switches the dry (unprocessed) signal on or off. Switch it off when you want to use only the frequency shifter. The dry signal is present when the button is lit.

Link Shifts (SSB Modulator S only)

When this function is activated, the frequency shift can be set simultaneously for both left and right signals.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the SSB effect signal.

SSB Delay M/S

This effect combines frequency shifting and delay. The frequency shifting is done in the feedback path of the delay. The signal frequency spectrum thus initially remains unaltered, but is shifted further and further with each succeeding echo. Both mono and stereo versions of the effect are provided.



Frequency Shift

Amount by which all frequencies in the signal spectrum are shifted. Both positive and negative shifts are possible.

Range

Multiplier for the Frequency Shift control – sets its effective range. 1.00000x permits a shift of ± 24000 Hz, while 0.00001x produces a range of ± 0.2400 Hz. By setting Range appropriately, Frequency Shift can be adjusted very finely.

ms/BPM Mode

Switches between **ms Mode** (delay time setting in milliseconds) and **BPM Mode** (delay time setting in terms of tempo).

BPM (BPM-Mode)

Specify the tempo you wish to match (available range: 25 .. 300 BPM).

Note (BPM-Mode)

Set delay time in terms of a musical note length relative to the **BPM** (tempo) setting. **P** and **T** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is $1/64T$. The longest attainable note value depends upon the tempo – a slower tempo limits the delay to smaller maximum note values. If adjustment of **BPM** causes the maximum delay time (682 ms for Delay M/S, 5460 ms for Delay LM/LS) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Delay (ms-Mode)

Sets delay time directly in milliseconds. The minimum delay setting is 4 ms, the maximum 5460 ms.

FB (Feedback)

Sets the strength of the feedback - the portion of the delayed signal that is routed back to the input to create multiple receding echos. The degree of feedback determines the fade out time of the repeated echoes – or, somewhat oversimplified, the number of echoes.

Cross FB (Cross Feedback, SSB Delay S only)

When Cross FB is enabled, the left and right feedback signals are cross-routed to the right and left delay line inputs respectively. This creates more complex frequency shifting effects than those obtained with simple feedback. When the button is lit, Cross FB is enabled.

LDamp (Low Damp)

Adjusts the amount of low-frequency damping applied to a signal per pass through the feedback loop.

HDamp (High Damp)

Adjusts the amount of high-frequency damping applied to a signal per pass through the feedback loop.

Pitch Shifter M/S

Use the Pitch Shifter to change the pitch of a signal without altering its duration. The pitch shifter can be used to produce a second voice at an adjustable, fixed interval from the original signal. With the stereo version, the pitch for each channel can be independently adjusted, as well as the volume and pan position. The mono version includes a delay you can apply to the pitch shifted signal.



Coarse (L / R)

Adjusts the detune value in half-tones. Note that high values here require you to use a correspondingly high Time Range setting for a clean-sounding result.

Fine (L/R)

Adjusts the detune value in very fine increments. The range is +/- 99 cents, where a cent is 1/100th of a half-tone (1/2 tone = 100 cents).

Level L/R (only in Pitch Shifter S)

Controls the volume level of the effect signal.

Pan L/R (only in Pitch Shifter S)

Controls the left/right positioning of the effect signal in the stereo field.

Delay (only in Pitch Shifter M)

Controls the delay of the effect signal. The range is from 0 - 2000 ms.

Feedback (only in Pitch Shifter M)

This controls, for each channel, the amount of the delayed signal which will be fed back to the input to be delayed again. Simply put, the higher this setting, the more echoes you will get.

High Damp

Use this control to adjust the amount of low frequency damping in the feedback loop for the respective channel.

Time Range

This control influences the quality of the effect. The optimal value depends on the nature of the raw material, so you should experiment with it somewhat. In general, the larger the value, the more exact the calculation, although it may take longer.

LFO Modulation

Frequency

You can modulate the pitch with an LFO. This controls the frequency of the LFO, or the rate of the modulation.

Mod Depth

Controls the depth of the pitch modulation.

Dry

Adjusts the level of the original signal.

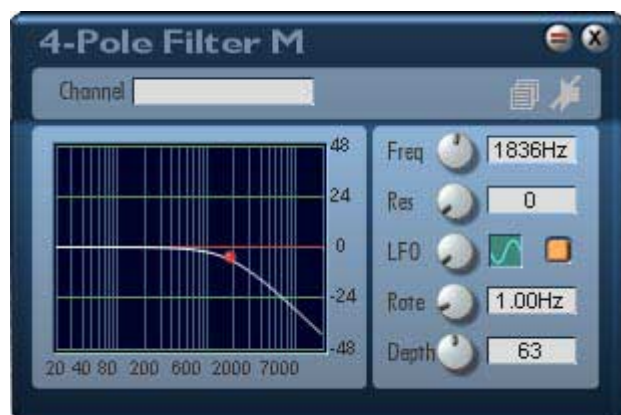
Wet

Adjusts the level of the signal after processing by the effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

4-Pole Filter M/S

This 4-Pole Filter is a lowpass filter with adjustable resonance which can be periodically modulated. Use this filter to create outstanding periodic filter effects. Two versions of the filter are provided, a Mono-Insert and a Stereo-Insert. The functionality of the two is identical. You can adjust the filter quickly and intuitively by direct graphic editing. Adjust the cutoff frequency by moving the mouse horizontally while holding the (left in PC version) mouse button. Adjust the resonance by moving the mouse vertically with the right button pressed ('Ctrl' + mouse button in the Mac version).



Freq

Adjusts the lowpass filter's cutoff frequency.

Res (Resonance)

Strength of the filter resonance.

LFO Wave

Several waveforms are available to be used by the integrated LFO for cutoff frequency modulation. Select the waveform using the arrow keys, or by clicking on the display area and moving the mouse on or off. The following waveforms can be selected: Sine, Rectangle, Saw up, Saw down, Triangle, Rectangle with variable pulsewidth.

LFO On/Off

Use the ON button to enable the LFO for cutoff frequency modulation.

Depth

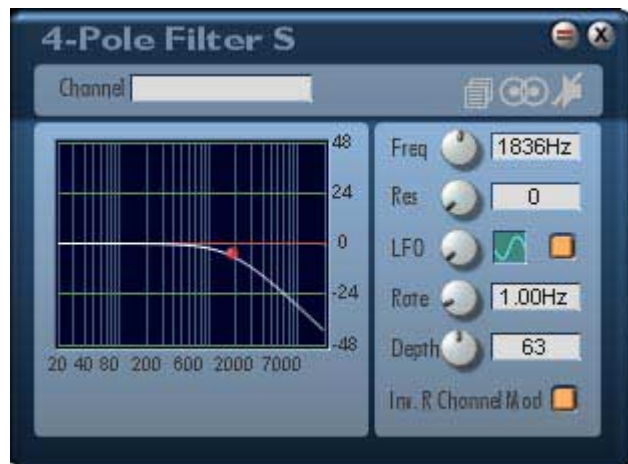
Strength of the filter frequency modulation by the LFO.

Rate

Adjusts the LFO frequency.

Inv. R Channel Mod (invert right channel modulation, only on 4-Pole Filter S)

Inverts the modulation phase of one channel, which leads to some very interesting effects.



MidiPole M/S

The MidiPole is a low-pass filter with 24 dB/octave rolloff, adjustable resonance and an integrated LFO for cutoff modulation which can be synchronized to a MIDI clock and resynchronized to a specifiable starting phase via MIDI note triggering. Both mono and stereo versions of the effect are provided.



Frequency

Adjusts the filter cutoff frequency.

Res (Resonance)

Adjusts the amount of filter resonance.

LFO Wave

Sets the LFO waveform used for cutoff frequency modulation. Choices include Sine, Square, Saw Up, Saw Down, Triangle und Random.

LFO On/Off

Switches LFO modulation on or off. Modulation is active when the button is lit.

Retrigger

Enables synchronization or restarting of the LFO waveform via MIDI note-on events. Retriggering is enabled when the button is lit.

Note

Sets the MIDI note number which can be used for LFO retriggering when the Retrigger function is switched on.

MIDI

Sets the MIDI channel upon which LFO retriggering via note events can be done when the Retrigger function is switched on. (MIDI clocking functions independently of the MIDI channel setting.)

Internal Clock

Switches between internal und external (MIDI) clocking. Internal clocking is selected when the button is lit.

BPM

For setting tempo (internal clocking) or displaying the tempo of an external MIDI clock source. In the field at left, whole-BPM values are entered or displayed. In the field at right, values of 1/100th of a BPM are entered or displayed.

Repeat

Indicates the note length to which a single period of the LFO corresponds.

Depth

Adjusts the amount of filter cutoff frequency modulation produced by the LFO.

Phase

Sets the phase, or waveform position, to which the LFO is reset when it is retriggered by the occurrence of the MIDI note specified by the Note setting (the Retrigger function must be active).

Inv. R Channel Mod (invert right channel modulation, MidiPole S only)

Inverts the LFO modulation of the right-channel filter cutoff with respect to that of the left channel – the cutoffs of the two filters are modulated out-of-phase with one another, rather than in tandem.

Resonator M/S

This effect implements a comb filter which is optionally modulated by its own LFO. The filter is adjustable for *frequency*, *damping* and *resonance*. The LFO has selectable *waveforms*, and adjustable *depth* and *rate* parameters. This effect is provided in both mono and stereo versions.



Freq (Frequency)

Sets the basic frequency for the filter. This frequency will be modulated by the LFO.

Res(Resonance)

Determines the strength of the comb filter effect. The greater the resonance, the more pronounced the effect.



Damp

Controls the resulting overall tone color of the filter. Higher values produce darker, or softer results.

Gain

The Gain control boosts the filtered signal by up to +12 dB.

LFO Wave

Selects the LFO waveform to be used to modulate the filter frequency. The following waveforms are available: Sine, Square, Saw Up, Saw Down, and Random.

Depth

Sets the strength of the modulation of the adjusted filter frequency by the LFO.

Rate

Controls the frequency of the LFO (modulation rate).

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the signal after processing by the resonator effect.

Ring Modulator M/S

The Ring Modulator multiplies an audio input signal by an internal sine wave and outputs the result. If you modulate the sine wave using an LFO, all kinds of sonic effects evolve, from subtle spectral changes to more obvious electronic or bell-like effects. The Ring Modulator is supplied in both mono and stereo versions.



InGain

Adjusts the level (preamplification) of the input signal. It can be boosted by up to 12 dB.

Sine Freq

Controls the basic frequency of the sine wave oscillator. The range is from 1 Hz, to 4000 Hz.



LFO Wave

Selects the waveform the LFO will use to modulate the sine wave. The following waveforms are available: Sine, Square, Saw Up, Saw Down, and Random.

Depth

Sets the modulation depth. This controls the amount of deviation from the basic frequency the sine wave oscillator will produce.

Rate

Controls the rate at which the LFO modulates the sine wave oscillator.

Dry

Adjusts the level of the original signal.

Wet

Adjusts the level of the processed signal.

Tremolo M/S

The Tremolo effect modulates the amplitude of an input signal periodically according to the waveshape selected in the *Wavefield* at a rate set under *Rate*. The *Depth* parameter controls the degree of modulation. This effect is provided in both mono and stereo versions.



Depth

Controls the depth, or strength of the modulation of the input signal's amplitude.

Rate

Controls the rate, or speed, of the amplitude modulation.

Wave

Selects the waveform the LFO will use to modulate the amplitude. The following waveforms are available: Sine, Square, Saw Up, Saw Down, and Random.



Inv. R Channel Mod (invert right channel modulation, only on Tremolo S)

Shifts the phase of the right channel of the modulation signal relative to the left by 180 degrees. The modulation of the right channel follows the left.

MidiTremolo M/S

A tremolo effect which can be synchronized to MIDI clock. Both mono and stereo versions of the effect are provided.



LFO Wave

Selects the waveform the LFO will use to modulate the amplitude. The following waveforms are available: Sine, Square, Saw Up, Saw Down, and Random.

LFO On/Off

Switches LFO modulation on or off. Modulation is active when the button is lit.

Retrigger

Enables synchronization or restarting of the LFO waveform via MIDI note-on events. Retriggering is enabled when the button is lit.

Note

Sets the MIDI note number which can be used for LFO retriggering when the Retrigger function is switched on.

MIDI

Sets the MIDI channel upon which LFO retriggering via note events can be done when the Retrigger function is switched on. (MIDI clocking functions independently of the MIDI channel setting.)

Internal Clock

Switches between internal und external (MIDI) clocking. Internal clocking is selected when the button is lit.

BPM

For setting tempo (internal clocking) or displaying the tempo of an external MIDI clock source. In the field at left, whole-BPM values are entered or displayed. In the field at right, values of 1/100th of a BPM are entered or displayed.

Repeat

Indicates the note length to which a single period of the LFO corresponds.

Depth

Adjusts the amount of amplitude modulation produced by the LFO.

Phase

Sets the phase, or waveform position, to which the LFO is reset when it is retriggered by the occurrence of the MIDI note specified by the Note setting (the Retrigger function must be active).

Inv. R Channel Mod (invert right channel modulation, MidiTremolo S only)

Inverts the LFO modulation of right-channel amplitude with respect to that of the left channel – the two channels are amplitude-modulated out-of-phase with one another, rather than in tandem.

Autopan

When used with stereo signals, the Autopan effect exchanges the left and right channels periodically. With a mono signal, the effect operates as a classic panning effect - the signal sweeps periodically from the left channel to the right and back. The modulation is controlled by an LFO with settings for *Waveform*, *Depth*, and *Rate*.



Depth

Controls the depth, or strength of the pan modulation.

Rate

Controls the rate, or speed, of the pan modulation.

Wave

Selects the waveform the LFO will use for pan modulation. The following waveforms are available: Sine, Square, Saw Up, Saw Down, and Random.

Stereo Pan

The Stereo Pan effect permits control over the stereo width of a stereo signal. Since both left and right channels can be positioned anywhere in the full pan range, it's possible, for example, to reverse the stereo image.



Left

Adjusts the position of the left input channel signal in the stereo output field.

Right

Adjusts the position of the right input channel signal in the stereo output field.

Mono (use left input)

To generate a stereo signal from a mono input signal, connect a signal to the left input and enable this option. The input signal is then fed proportionately to both the left and right channels. This mode is enabled when the indicator is lit.

StereoExpander

The StereoExpander permits the "breadth" of a stereo image to be increased or decreased. The processing is mono-compatible and frequency-independent.

Controls

Amount

Adjusts the intensity of the effect. Positive values produce a broadening of the stereo image, negative values cause it to become narrower.



Soft Clip M/S

The SoftClip module permits the loudness of a signal to be boosted without introducing digital overload distortion (hard clipping). Even with signals which are already at the maximum level, an apparent loudness increase can be produced. In addition, the sound takes on a warmer, more "analog" character. Both mono and stereo versions of the effect are provided.



Controls

Drive

Adjusts the intensity of the effect. An orange LED lights when the effect begins working.

Level

If you wish to alter merely the sound of a signal and not its loudness, you can use this control to counteract the loudness increase produced by the effect.



Dither/Shaper M/S

The Dither/Shaper is a mastering tool used to compensate for quantization errors that result from converting a digital audio signal from one resolution to another, lower resolution. For example, in Pulsar, audio signals are processed at a 32-bit resolution internally. When transferring the signal using one of the various drivers (Wave, Asio, EASI etc.) for 16 or 24 bit recording, the lower, or least significant, bits are ignored. Since the dynamic range is reduced by 8 or 16 bits, quantization noise increases. Our ears perceive this as a subtle, but unpleasant digital noise. The Dither/Shaper replaces this unpleasant noise with noise that is more psychoacoustically agreeable (more pleasant). It also shifts the noise into a frequency range in which human hearing is less sensitive.



Before using the Dither/Shaper, it is important to note that its use is only recommended if you have extremely quiet passages in your music. Generally, if your signal levels are well controlled, you should work without the Dither/Shaper.

Bit Resolution

This sets the new, or current bit resolution you are working with,

Dither

Selects the desired dither function or turns dither off.

Use your ears to determine the appropriate algorithm. Select the function that is the most inconspicuous.

Shaper

Selects the desired Shaper algorithm, or turns the Shaper function off. The Shaper is responsible for shifting the noise into the less sensitive human hearing ranges.

As with the dither function, you should use your ears to determine the best Shaper algorithm. Select the one that is most inconspicuous.

DC Filter M/S

Removes any existing DC component from a signal. Both mono and stereo versions of the filter are provided.

Controls

Gain

The DC Filter operates with 6 dB headroom in order to avoid distortion. Typically, these extra dB can be recovered following DC removal via the amplifier stage which follows. However, if the peak-indicator LED begins lighting, the Gain control should be backed off slightly.

Signal Peak

The signal LED (green) indicates all signals above the level of -96dB. The Peak LED indicates the presence of distortion in the amplifier stage following the DC filter.



MultiFX M/S

These modules permit series-chaining of up to six effects in any combination. Since they feature MIDI interfaces, the loaded effects can be MIDI-automated. A MultiFX module can be loaded into a mixer effect insert slot, where it serves to expand the capacity of that slot. Use of a MultiFX module in XTC mode offers a further advantage: only a single communication connection between DSP card and HDR software is required for all of the series-connected effects loaded into that module. This minimizes the card-software communication traffic, as well as the latency times associated with it. Both a mono and a stereo version are provided.

Controls

Insert Slots

Drag any desired effect into any of these slots to load it into that slot. Double-clicking on the label in the slot opens the control surface of the effect. To remove an effect, click on the slot to select it and hit the Delete (PC) or Num (Mac) key on your computer keyboard.



Active

This button enables or disables the effect for its respective insert slot. An effect is enabled when the indicator is lit. When an insert slot is disabled, the signal is passed through to the next slot and no DSP resources are consumed for that effect.

MIDI Active

Switches MIDI reception on or off.

In XTC mode, the MIDI functions are not accessible, since reception of MIDI data by effects modules is not possible in this mode.

MIDI Channel

Sets the MIDI channel for MIDI reception.

Effects loaded into the Insert Rack can be controlled by MIDI controllers. The assignment of MIDI controllers is described in the user manual. In XTC mode, the MIDI functions are not accessible, since reception of MIDI data by effects modules is not possible in this mode.