Effects

Aux Effects: Chorus/Flanger, Delay and Reverb

The Aux Effect Editor

Adding the effect component of an Aux effect to the signal of another effect

Effect Source – Effect presets in Multi mode

Stereo Chorus / Flanger

Delay

LCR Delay

Dual Delay

Multitap

Reverb

Insert Effects

Loading insert effects

List of Insert Effects

Common Effects Controls

Compressor

Limiter

Gate

Expander

Parametric EQ / Stereo EQ

Graphic Control

Graphic EQ

Decimator

Distortion

Overdrive

Graphic Control of the Equalizer

AutoWah

Amplifier

Pre EQ

PostEQ

Chorus Effects

Ensemble

Harmonic Chorus

Master Chorus

Triple Chorus / Hexa Chorus

Flanger Effects

Harmonic Flanger

Master Flanger

Random Flanger

Space Flanger

Step Flanger

Master Phaser

What is SSB?

SSB Phaser

2 Voice Pitch Shifter

Stereo Pitch Shifter

Feedback Pitch Shifter

Resonator

Ringmodulator

Tremolo

Autopan

Stereo Expander

Tube Processor

Softclip

Dynamics

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fidelity at work.

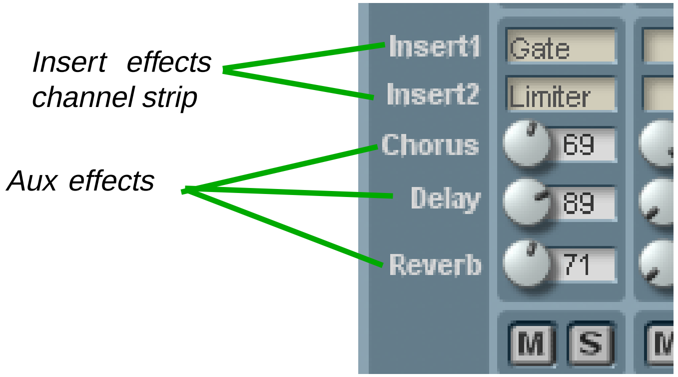
NOAH - Tactive Instrument Modeller

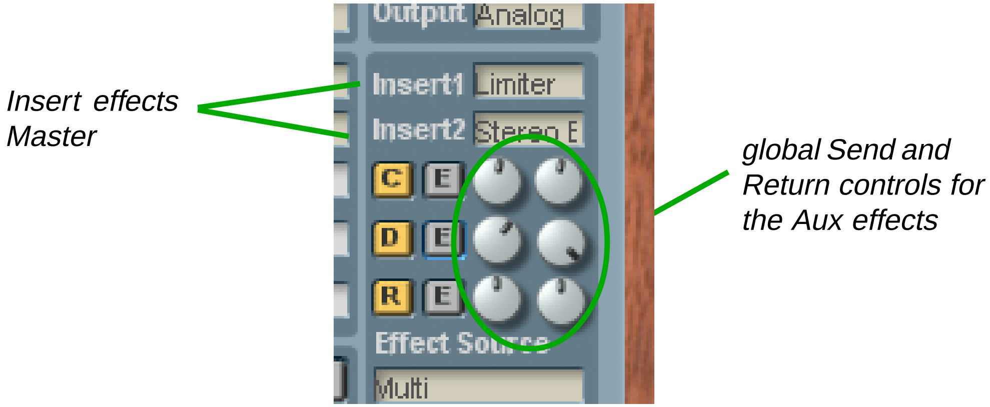
Effects

With Noah you can add two kinds of effects, Aux and Insert:

A) The Chorus/Flanger, Delay and Reverb effects can be routed over the mixer's Aux path to be used as Aux effects (also referred to as Send effects).

B) The Mixer's channel strips are equipped with Insert slots for two effects you can select from the extensive palette of supplied Insert effects. Because each of the channel strips is supplied with insert slots and led to the master channel, insert effects (maximum two) can also be applied to external signals.





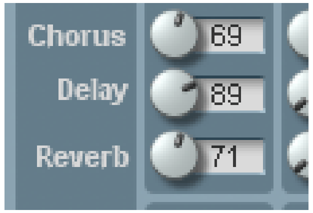
Aux Effects: Chorus/Flanger, Delay and Reverb

These effects are implemented as Auxilliary Effects; that is, they are available to all mixer channels (Instrument slots A – D, Analog In, USB In) through each channel's Aux Send lines. The settings of the effect parameters are common to all signal sources, however, as only one instance of the respective effect is loaded for all channels.

The Aux Effects are always loaded into their own address range on the DSPs and are therefore available independent of other loaded instruments.

Apart from the specific parameters of individual effects, which you adjust on the control panel of the effect itself, the following Mixer settings apply to Aux Effects.

Mixer channel strip:



Chorus: Controls the strength of the signal sent to the chorus effect for each channel.

Delay: Controls the strength of the signal sent to the delay effect for each channel.

Reverb: Controls the strength of the signal sent to the reverb effect for each channel.

Mixer master functions:

C: Switches the chorus on or off.

D: Switches the delay on or off.

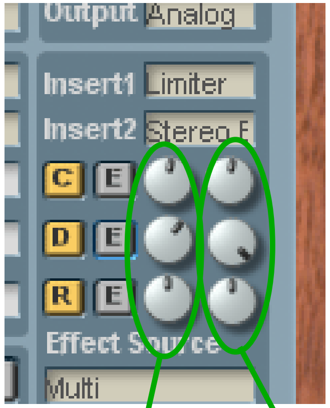
R: Switches the reverb on or off.

E: Opens the editor for the respective chorus, delay or reverb effect.

Send: Controls the effect send level for the respective chorus, delay, or reverb effect.

This refers to the global send level. The send levels routed to the aux effects from the individual instruments are adjusted via the *Chorus*, *Delay* and *Reverb* controls of the individual channel strips.

These controls are also present in the Aux editor.



Send controls *Return controls*

Return: Controls the effect return level for the respective chorus, delay, or reverb effect – thus, the level with which the effect appears in the overall mix.

These controls are also present in the Aux editor.

Effect Source: In *Multi* mode, you may at times wish to have the insert effect settings of a *Single* preset from one of the instruments in the *Multi* apply, instead of those of the *Multi* preset itself. In this case, select here the slot in which that instrument is loaded (see also the relevant section of the following description of the Aux editor).

An appropriate set of choices for the parameters of the aux effects is provided in the Aux editor.

Important note: The output of the Aux Effects is sent only to the mix. If the internal mixer outputs are routed to an Adat or USB output, the effects will not be audible.

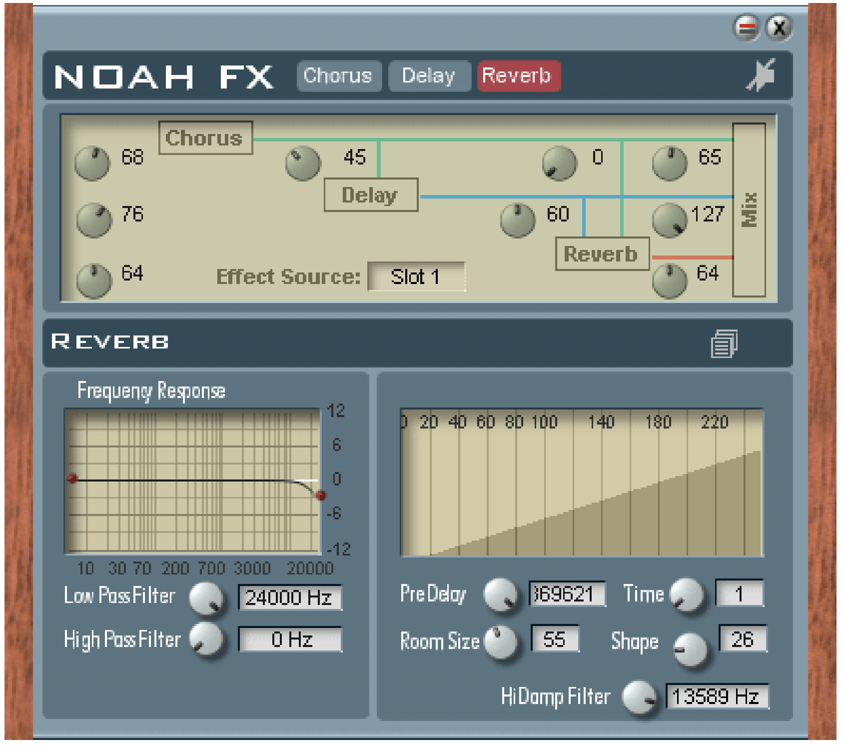
The Aux Effect Editor

You open the Aux Effect Editor via the Live Bar or by clicking the "E" switch in the Master section of the mixer panel. In the Editor you can adjust all of the individual effect parameters as well as establish the routing among the Chorus/Flanger, Delay, and Reverb effects.

Chorus / Delay / Reverb: Use these switches to show the parameters of the desired effect in the display.

Bypass Icon: Removes all effects from the mix.

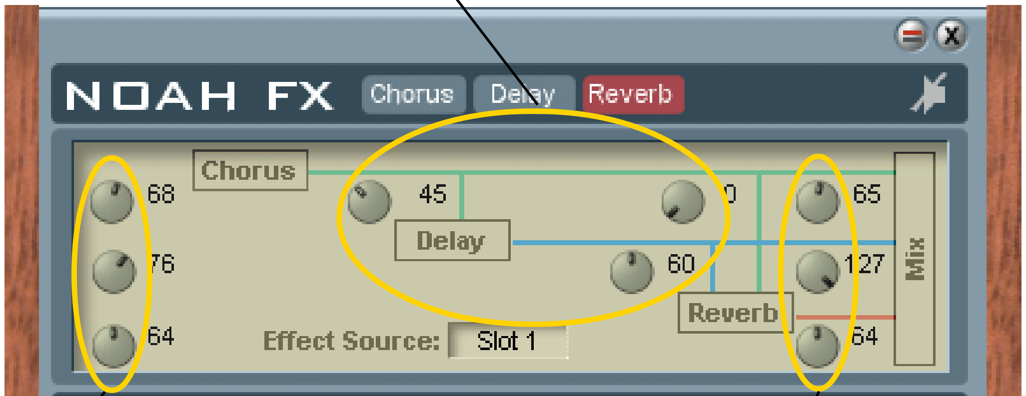
Important note: The output of the Aux Effects is sent only to the mix. If the internal mixer outputs are routed to an Adat or USB output, the effects will not be audible.



Adding the effect component of an Aux effect to the signal of another effect

Often it is desirable for the output of one aux effect to be processed with another aux effect. An example in a typical studio environment would be to add reverb to a delayed signal. If you include the delay and reverb in the mix using two different aux paths, the original signal will have reverb, but the delayed signal will not, and the result will sound unnatural. The experienced recording engineer will probably not route the delay signal back to the mix through a return, but rather through another channel where reverb can be added to it before sending it back along the effects path. Noah, too, allows you to establish such signal routings among the three effects.

For mixing to the inputs of other aux effects

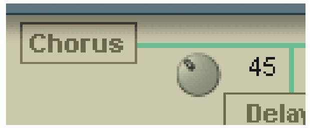


Send controls (same as in Mixer)

Return controls (same as in Mixer)

For example, in the Aux Editor you can send the output of the Chorus to the inputs of the delay and reverb with an independent level for each. Then you can send the output of the delay to the reverb at a level independent of its level in the overall mix.

Chorus (Send1): Controls the level of the chorus effect output added to the delay input signal independent of the chorus level in the overall mix.



Chorus (Send 2): Controls the level of the chorus effect output added to the reverb input signal independent of the chorus level in the overall mix.

Delay (Send 1): Controls the level of the delay effect output added to the reverb input signal independent of the delay level in the overall mix.

The following Send and Return controllers correspond to their respective Send and Return controls in the mixer's Master section.

Send: Controls the effect send level for the respective chorus, delay, or reverb effect.

> This refers to the global send level. The send levels routed to the aux effects from the individual instruments are adjusted via the *Chorus*, *Delay* and *Reverb* controls of the individual channel strips.

Return: Controls the effect return level for the respective chorus, delay, or reverb effect – thus, the level with which the effect appears in the overall mix.

Effect Source – Effect presets in Multi mode

The effect settings for an instrument are usually stored within the presets for that instrument.

This means that in *Multi* mode you have to specify the instrument whose settings you want to use. Because in *Multi* mode only two Insert effects can be loaded simultaneously, it is generally not possible to load the effects from all instrument presets.

Switching to another instrument while working in *Multi* mode would reload this instrument's effects and thereby replace the current effect settings. There may, however, be times when you are happy with the current effect settings but would like to change an instrument.

Therefore, when Multi mode is active, a field is displayed which permits you to choose whether to use the effects settings from the *Multi* configuration or from one of its instruments. This choice is available separately **for the** Insert effects (Effect Source in the Master section of the mixer)

Effect source for insert effects (Mixer)

Effect source for aux effects (Aux effects editor)

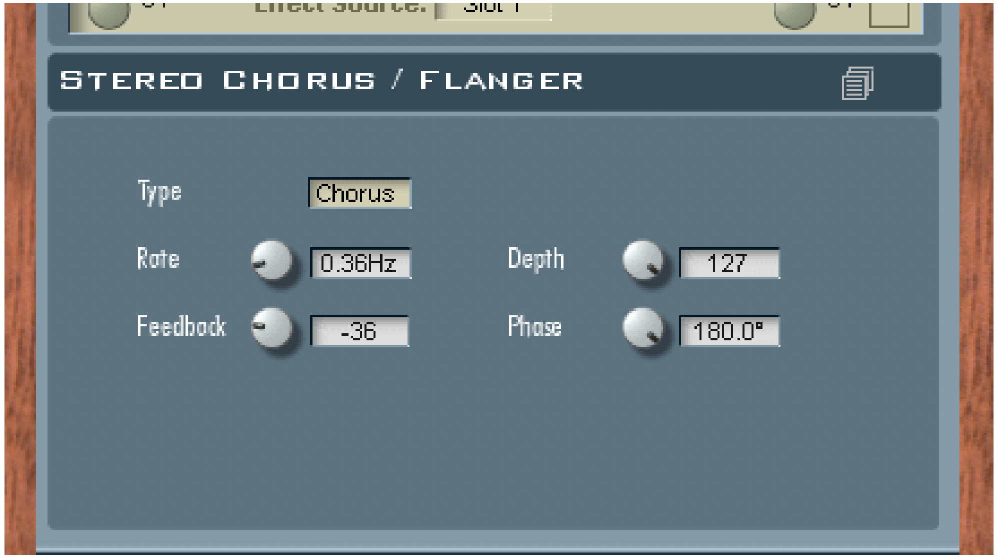
and the Aux effects (Effect Source in the Aux editor). In both places, you have the option of selecting either *Multi* (the Multi's effects settings will be used, instrument preset effects settings will be ignored) or a specific instrument whose presets contain the effect settings to be loaded (specify the desired instrument by selecting the slot into which it has been loaded).

Stereo Chorus / Flanger

The parameters of the Chorus / Flanger are displayed in the lower portion of the Aux effect editor when you click the *Chorus* button in the upper portion.

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 modulated periodically. Mixing this delayed signal with the original produces the chorus effect.

Like the chorus, a flanger is based on a delay line with a periodically modulated delay time. However, the delay times in a flanger are substantially shorter than those of a chorus. In addition, the flanger feeds back the delayed signal 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.



Type

Selects either Chorus or Flanger mode.

Rate

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

Depth

Adjusts the amount of delay time modulation in the chorus/flanger effect.

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.

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.

Delay

When you click the Delay switch in the editor the parameters of the currently selected delay appear in the lower part of the editor. Select one of the various delays in the **Type** field.

Stereo Delay

The signal sent to the delay is delayed for a specified time. The delay time is adjustable independently for each stereo channel, and can be fed back to produce recurring echos. An integrated filter controls the attenuation of the echos depending on frequency and intensity.

ms/BPM Mode

Switches between **ms** mode and **BPM** mode.

Delay L/R (BPM-Mode)

Set delay time in terms of musical note values relative to the **BPM** (tempo) setting. **dot** and **trpl** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is 1/64trpl. The longest attainable note value depends on the tempo – a slower tempo limits the delay to smaller maximum note values. If adjustment of **BPM** causes the maximum delay time (4000 ms) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

The tempo itself (BPM) is adjusted globablly via the MIDI Manager.

Delay L/R (ms-Mode)

Sets delay time directly in milliseconds. The minimum delay setting is 4 ms, the maximum 4000 ms.

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 Feedback

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.

Lo Damp

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

Hi Damp

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

LCR Delay

The LCR Delay sends the delayed input signal to the left, center or right channel. The delay is adjustable per channel (L/C/R) and a feedback loop produces repeated echos if desired. Filters in the feedback loop allow for high or low frequency attenuation of the echos.

ms/BPM Mode

Switches between **ms** mode and **BPM** mode.

Delay L/C/R (BPM-Mode)

Set delay time in terms of musical note values relative to the **BPM** (tempo) setting. **dot** and **trpl** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is 1/64trpl. 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 (4000 ms) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Delay L/C/R (ms-Mode)

Sets delay time directly in milliseconds. The minimum delay setting is 4 ms, the maximum 4000 ms.

Level L/C/R

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

Lo Damp

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

Hi Damp

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

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.

Dual Delay

This delay provides the left and right channels with their own independent feedback loops. Filters are included in the feedback loop to dampen the high or low frequencies of subsequent echos.

ms/BPM-Mode

Switches between **ms** mode and **BPM** mode.

Delay L/R (BPM-Mode)

Set delay time in terms of musical note values relative to the **BPM** (tempo) setting. **dot** and **trpl** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is 1/64trpl. 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 (4000 ms) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Delay L/R (ms-Mode)

Sets delay time directly in milliseconds. The minimum delay setting is 4 ms, the maximum 4000 ms.

Hi Damp L/R

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

Lo Damp L/R

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

Feedback L/R

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.

Multitap

The Multitap provides four delays with independently adjustable volume and pan position. The delay time 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 4000 ms.

ms/BPM-Mode

Switches between **ms** mode and **BPM** mode.

Tap 1-4 (BPM-Mode)

Set delay time in terms of musical note values relative to the **BPM** (tempo) setting. **dot** and **trpl** indicate "dotted" and "triplet", respectively. The shortest note value (delay time) is 1/64trpl. 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 (4000 ms) to be exceeded, the **Note** setting is automatically "stepped down" to the next-largest value.

Tap 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 4000 ms.

Level 1-4

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

Pan 1-4

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

Lo Damp

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

Hi Damp

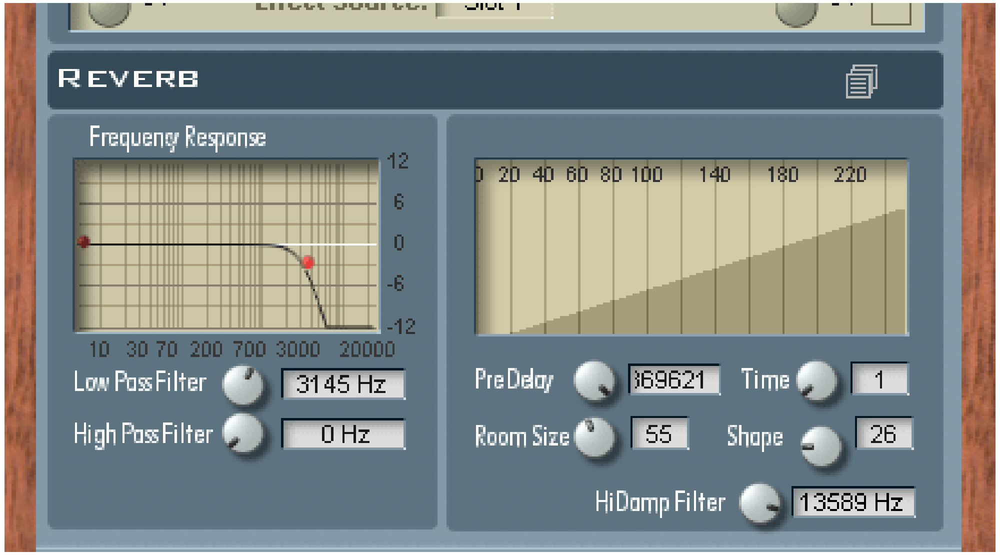
Adjusts the amount of high-frequency damping in the feedback loop.

Feedback

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

Reverb

Click the *Reverb* button in the Aux Effect editor to register the Reverb parameters in the display.



Lowpass Filter

A lowpass filter with a 12dB/Octave slope is located at the reverb input. Adjust the frequency of this filter a) with the rotary control, b) by entering the value in the text field, or c) directly within the filter's graphic control interface by changing the position of the red point.

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.

Highpass Filter

A high pass filter, also with a slope of 12db/ocatve, follows the low pass filter in the signal path. As with the high pass filter, there are three ways to adjust the cutoff frequency: a) use the rotary control, b) enter the value in the text field, or c) adjust the value directly in the graphic display by changing the position of the red point.

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.

HiDamp Filter

This 6 db/octave filter operates on the reverb response by reducing the high frequencies in the response signal depending on how it is adjusted. There are two ways to adjust the cutoff frequency: a) use the rotary control, or b) enter the value directly in the text field.

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

PreDelay

Adjusts the delay of the response in milliseconds.

The PreDelay is used to separate the hall response from the direct signal. This is useful to increase the comprehensibility of vocals or speech. The impression of space is not altered.

Room Size

Adjusts the perceived room size.

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.

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.

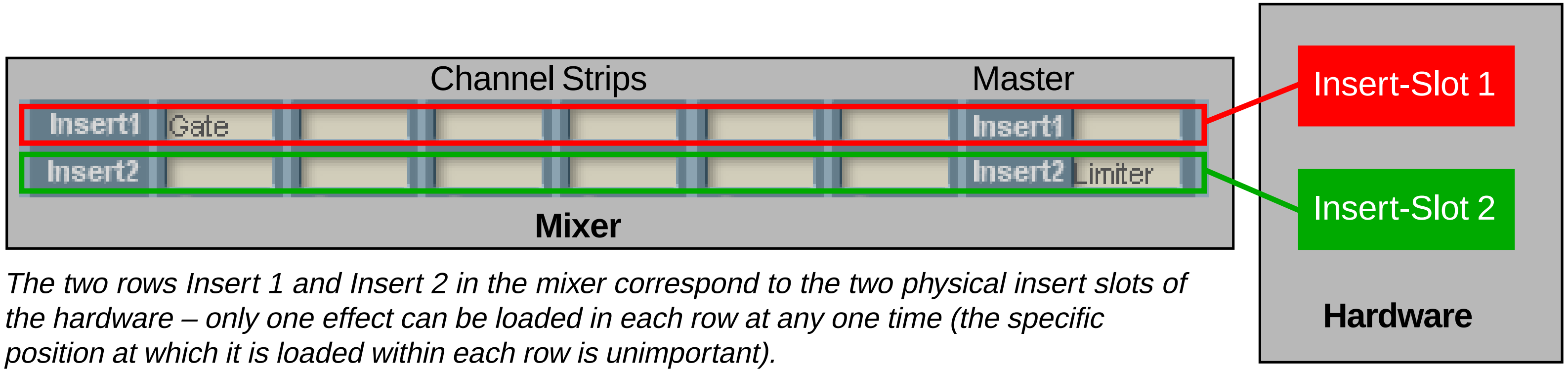
For percussive sounds like drums or other percussion instruments, smaller values for the Shape parameter are best.

Insert Effects

Noah allows for two insert effects to be loaded dynamically. Each channel strip and the Master channel are equipped with two insert fields (Insert1 and Insert2) which are assigned to two hardware insert slots. In other words, if an effect is loaded into Insert1 of one of the channel strips then no other effects can be loaded into Insert1 of any of the other channels.

In the Single operating mode you can load an effect into each of the Insert slots of a channel strip, or of the Master channel. Effects loaded into the Master channel process all individual channel signals as well as the analog and USB input signals.

In the Multi operating mode you can use the effects loaded into a physical insert slot of an instrument by assigning them in the Insert fields of the respective channels. Or you can load the effects into the Insert fields of the Master channel whereby all mixer channels and therefore all instruments will be affected.



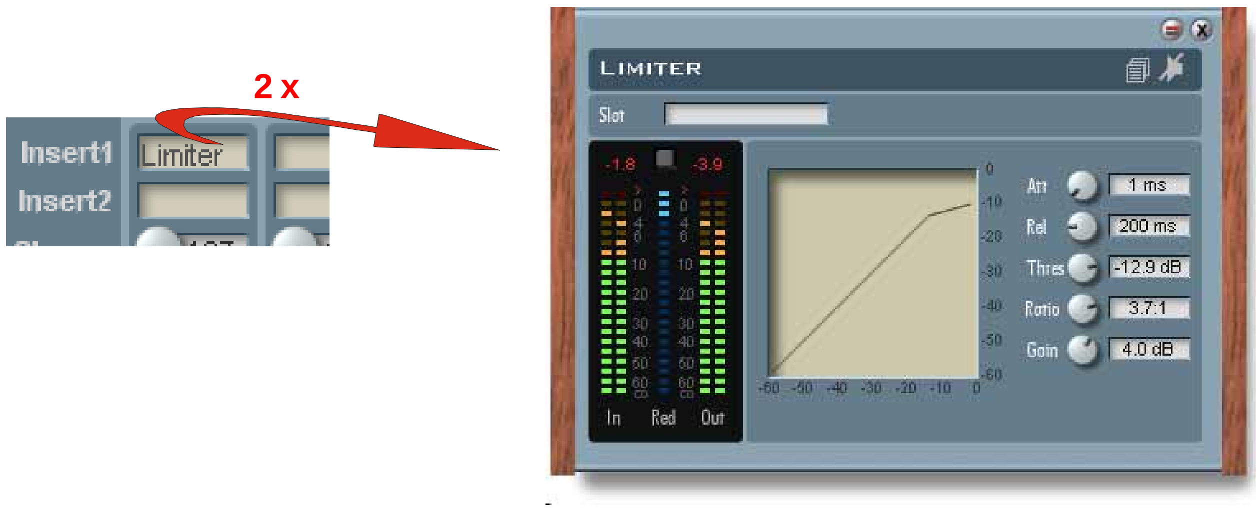
The two rows Insert 1 and Insert 2 in the mixer correspond to the two physical insert slots of the hardware – only one effect can be loaded in each row at any one time (the specific position at which it is loaded within each row is unimportant).

Loading insert effects

An insert effect is **loaded** by right-clicking on an insert field. This pops up a menu containing all available insert effects, from which a selection can then be made.

To **remove** an effect, use the same procedure, selecting the entry *None* at the top of the list. Or, simply load a different effect at a different position within the same row (*Insert 1* or *Insert 2*) – this causes the already-loaded effect to be removed automatically (you will first be asked to confirm this action), since each row can accommodate only one effect at a time.

The **control surface** of an effect can be opened and closed by double-clicking on the insert field.



List of Insert Effects

No	Type	Name
1	Filter	Stereo EQ
2	Filter	Parametric EQ
3	Filter	Graphic EQ
4	Modulation	Ensemble
5	Modulation	Master Chorus
6	Modulation	Harmonic Chorus
7	Modulation	Hexa Chorus
8	Modulation	Triple Chorus
9	Modulation	Master Flanger
10	Modulation	Harmonic Flanger
11	Modulation	Random Flanger
12	Modulation	Space Flanger
13	Modulation	Step Flanger
14	Modulation	Master Phaser
15	Modulation	SSB Phaser
16	Other	2 Voice Pitch Shifter
17	Other	Stereo Pitch Shifter
18	Other	Feedback Pitch Shifter

No	Type	Name
19	Modulation	Auto Pan
20	Modulation	Tremolo
21	Filter	Auto Wah
22	Distortion	Amplifier
23	Distortion	Decimator
24	Distortion	Distortion
25	Distortion	Overdrive
26	Filter	Resonator
27	Filter	Ringmodulator
28	Other	Soft Clip
29	Other	Stereo Expander
30	Other	Tube Processor
31	Dynamic	Compressor
32	Dynamic	Expander
33	Dynamic	Limiter
34	Dynamic	Gate
35	Dynamic	Dynamics

Common Effects Controls

Some of the control elements found on the effects panels are common to all, and therefore not described for each individual effect.

Slot Name

This text field displays the name of the mixer channel strip in which the effect is loaded.

Preset List Switch

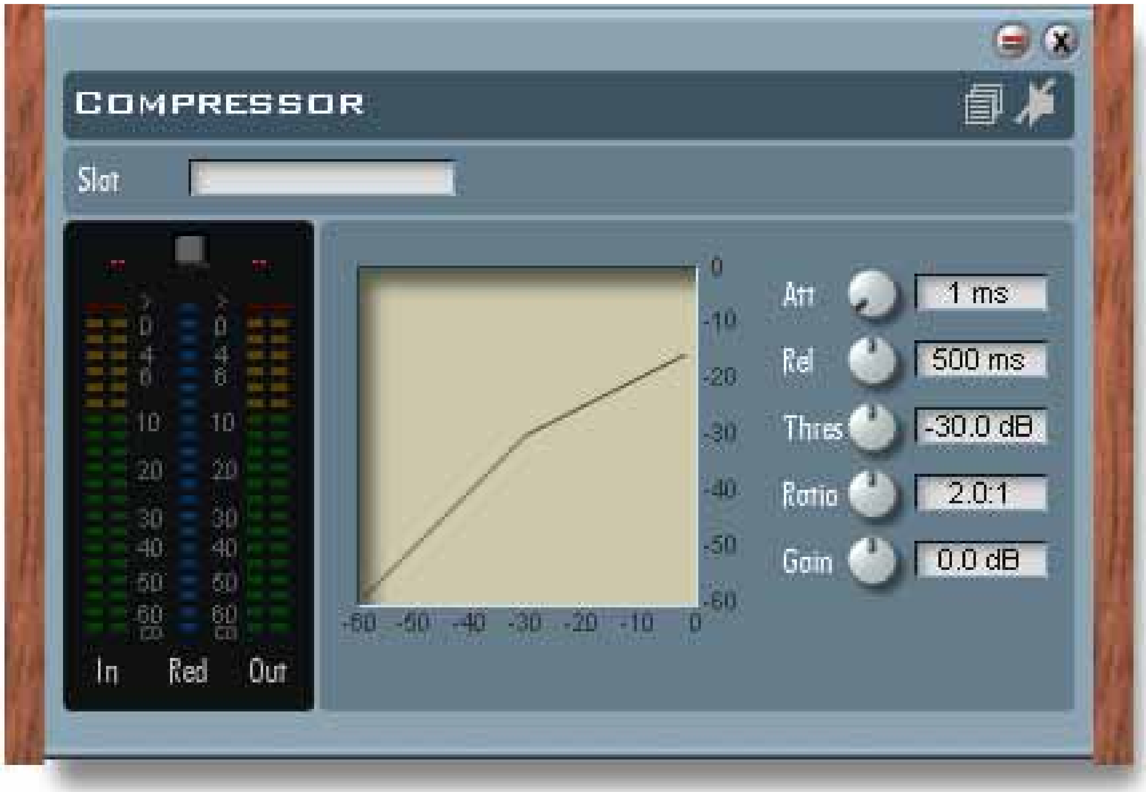
Each effect has its own Preset administration. This switch opens the Preset list.

Bypass Switch

Enable the bypass switch when you want to hear only the original signal (without the effect) for comparison purposes.

Compressor

The compressor modifies the dynamics of a sound by decreasing the volume level of the signal when it becomes louder. 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.



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

Att (Attack)

The *attack* time is the compressor's reaction time; the time it takes for it to respond to an increas in level above the threshold.

Rel (Release)

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

Thres (Threshold)

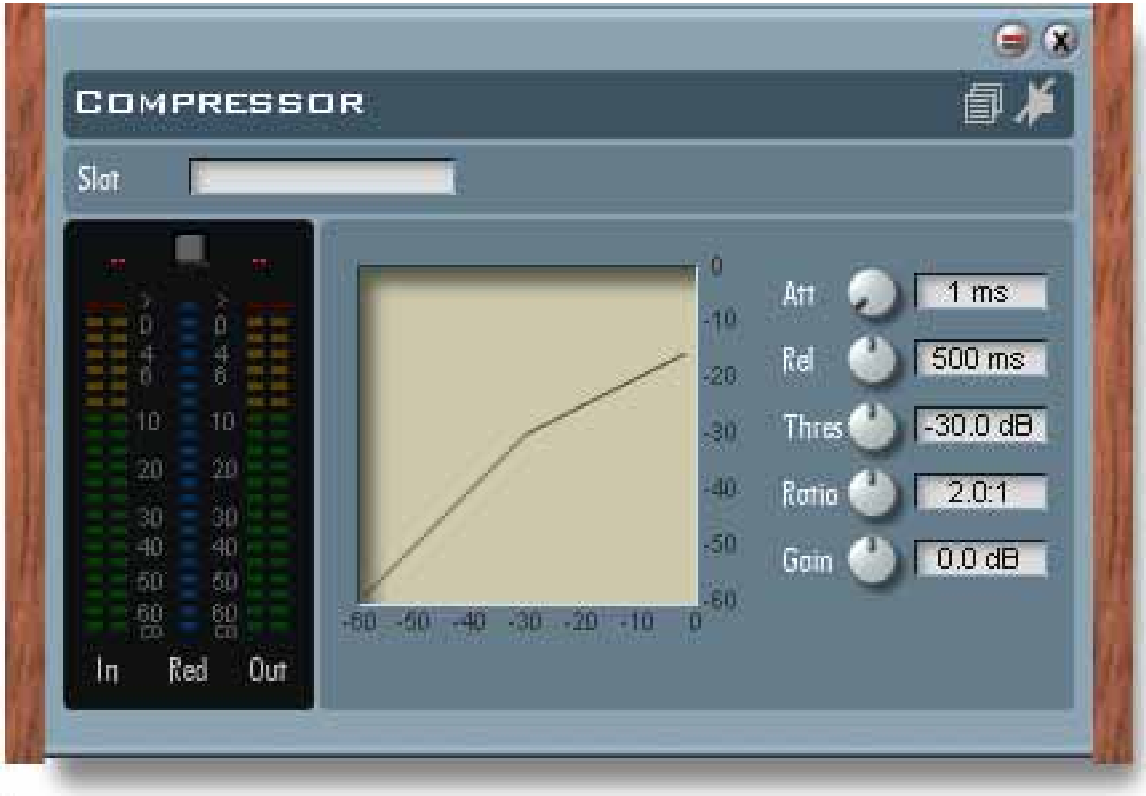
Sets the input signal level above which compression begins.

Ratio

The *ratio* adjusts the degree of compression for signals that exceed the threshold level. Compression is shown as a relative value. 1:1 means no compression. 3:1 means that a rise of +3dB in the input signal results only in a +1dB increase at the output.

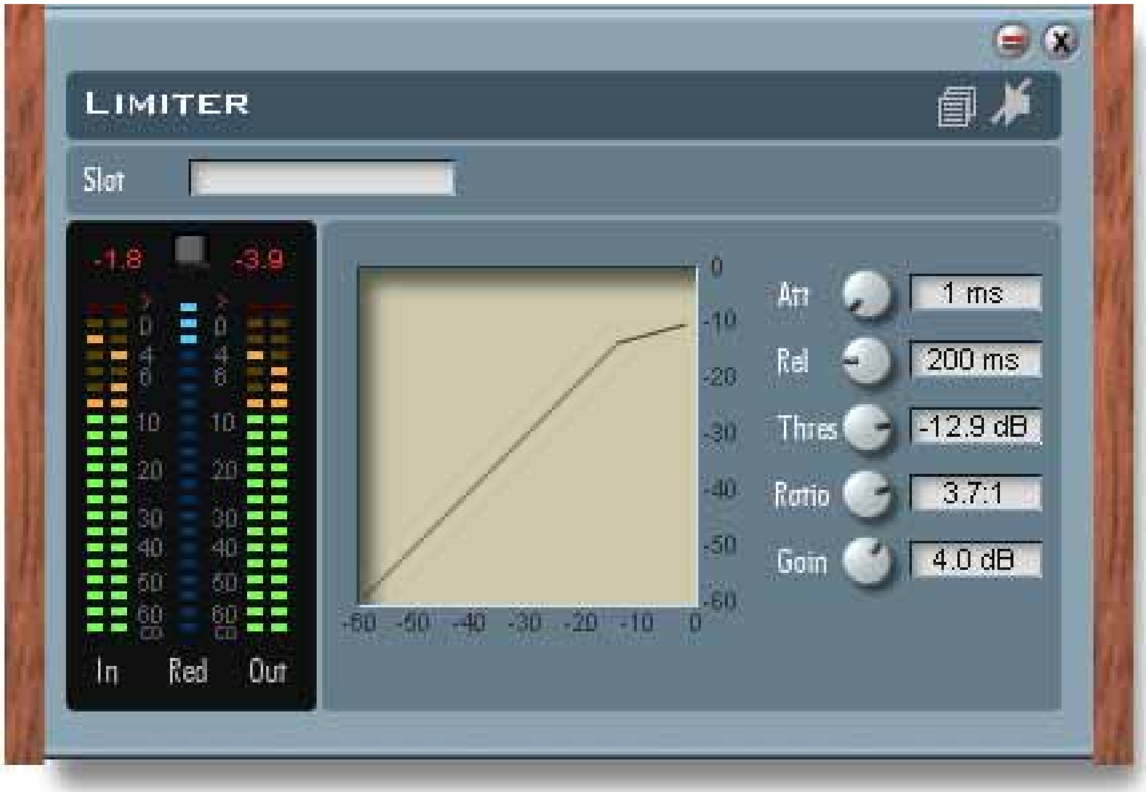
Gain

This increases the overall level of the signal, resulting in a fuller sound. Adjust the volume with the *gain* control.



Limiter

This effect is similar to compression and likewise modifies the dynamics of a signal. 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 ratio of input level change to output level change. The gain control adjusts the overall level of the output signal.



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

Att (Attack)

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

Rel (Release)

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

Tresh (Threshold)

Sets the input signal level above which limitting begins.

Ratio

The *ratio* adjusts the reduction rate for signals that exceed the threshold level. The compression is displayed as a relation value. 1:1 means that there is no reduction. 3:1 means that a an increase of +3dB of the input signal results only in +1dB of gain in the output.

Gain

This increases the overall level of the signal, resulting in a fuller sound. Adjust the volume increase with the Gain control.

Gate

A *gate* is just what its name says - a gate, or door. When it is open, signals 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.

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

Att (Attack)

Amount of attck time for the gate to open once the input signal level has exceeded the 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 gate takes to fully close once it begins to close (i.e., once the input signal level falls below the threshold, and after any remaining hold time has elapsed).

Tresh (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 Hysteresis control. With the Stereo Gate, opening of the gate is determined by the louder of the two channels.

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.

Hyst (Hysteresis)

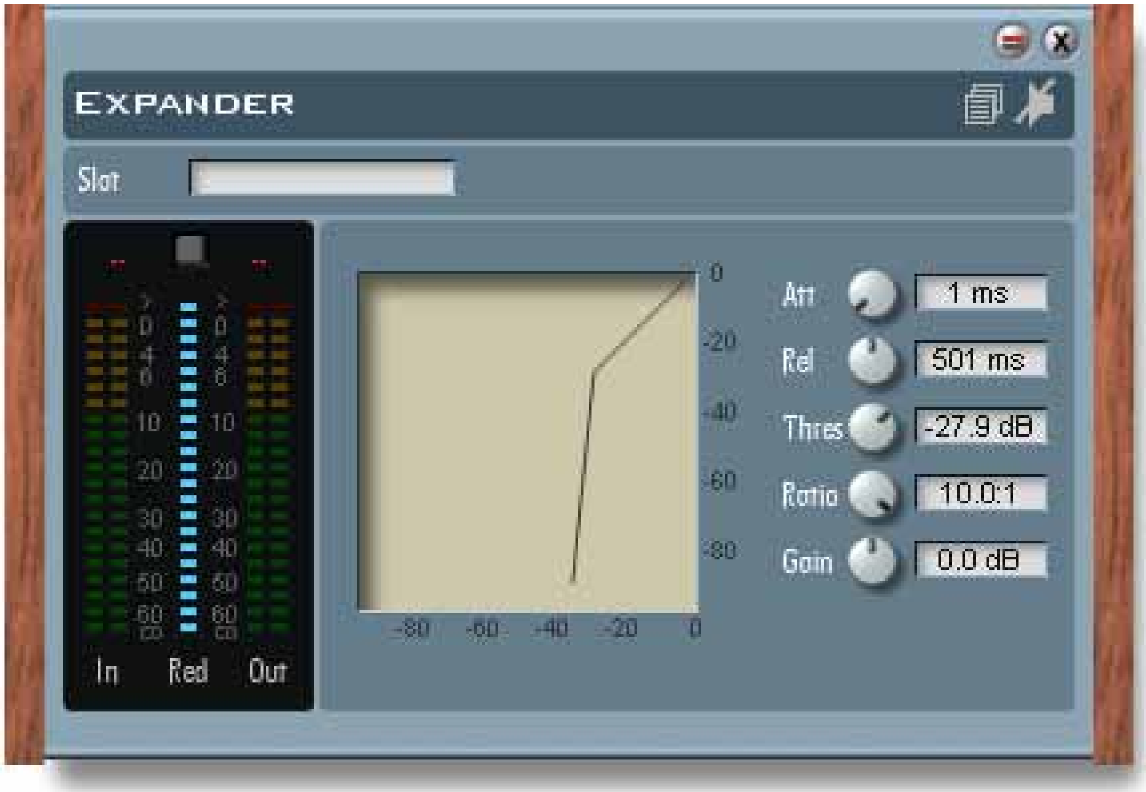
Adjusts the difference between the turn-on (upper) and turn-off (lower) thresholds. The turn-off threshold appears as a red point in the display. It can be set up to 10 dB below the turn-on threshold.

Gain

With the *gain* control you can increase the gate output level by up to 18dB.

Expander

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. 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 disengages when the threshold is crossed. Ratio controls the degree of expansion - how much the volume range is increased by the effect.



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.

Att (Attack)

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

Rel (Release)

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

Tresh (Threshold)

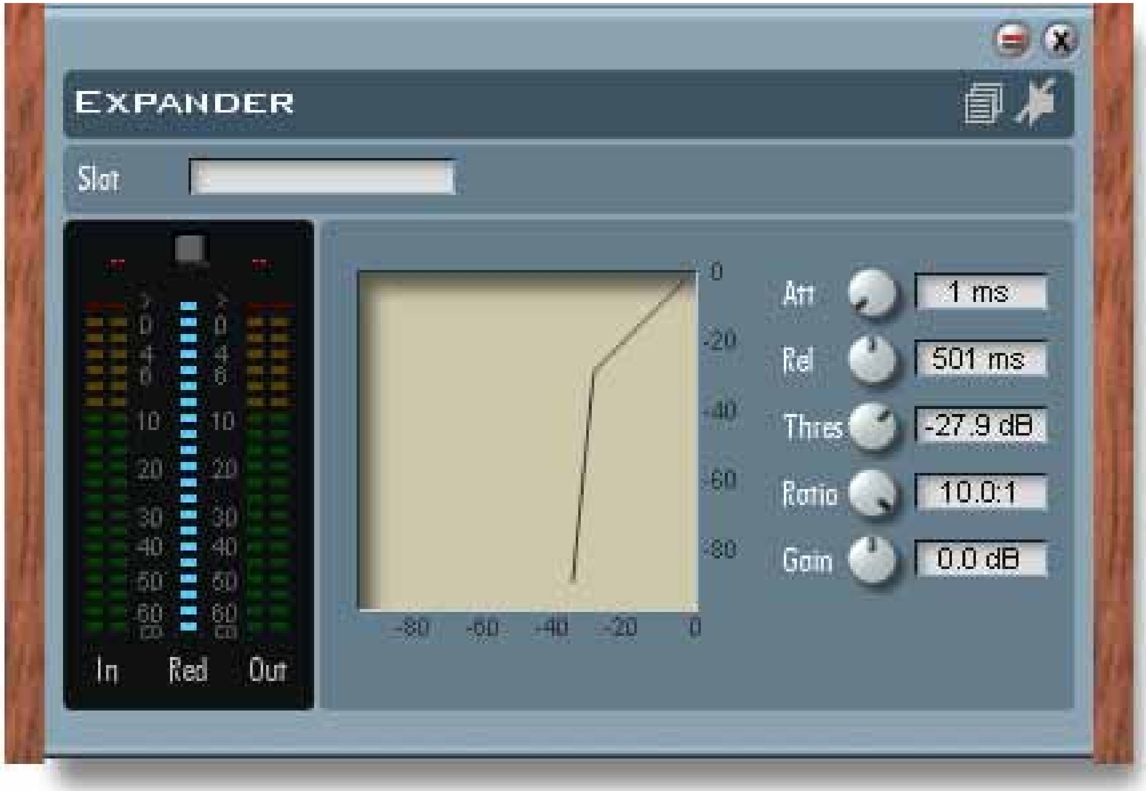
Sets the input signal level below which expanding engages.

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. 3:1e.g. means that a an attenuation of -1dB of the input signal results in -3dB at the output. The maxium Ratio is 10:1.

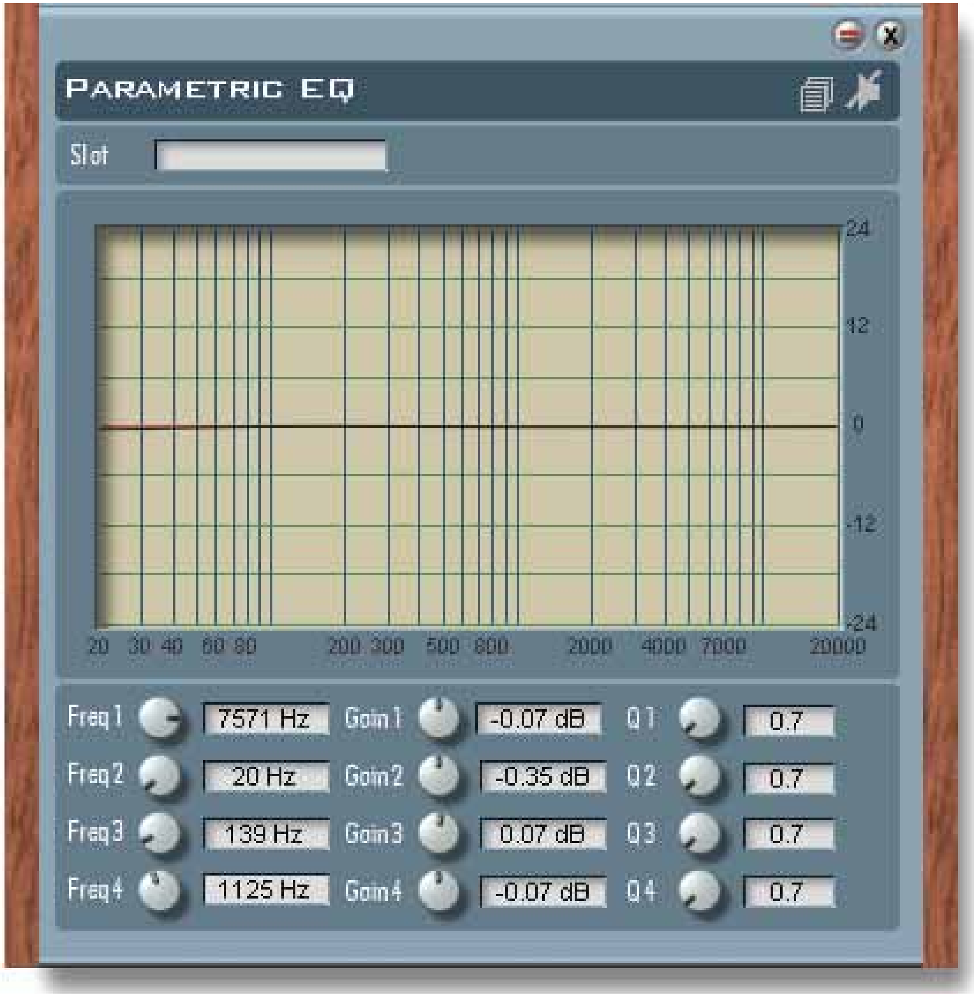
Gain

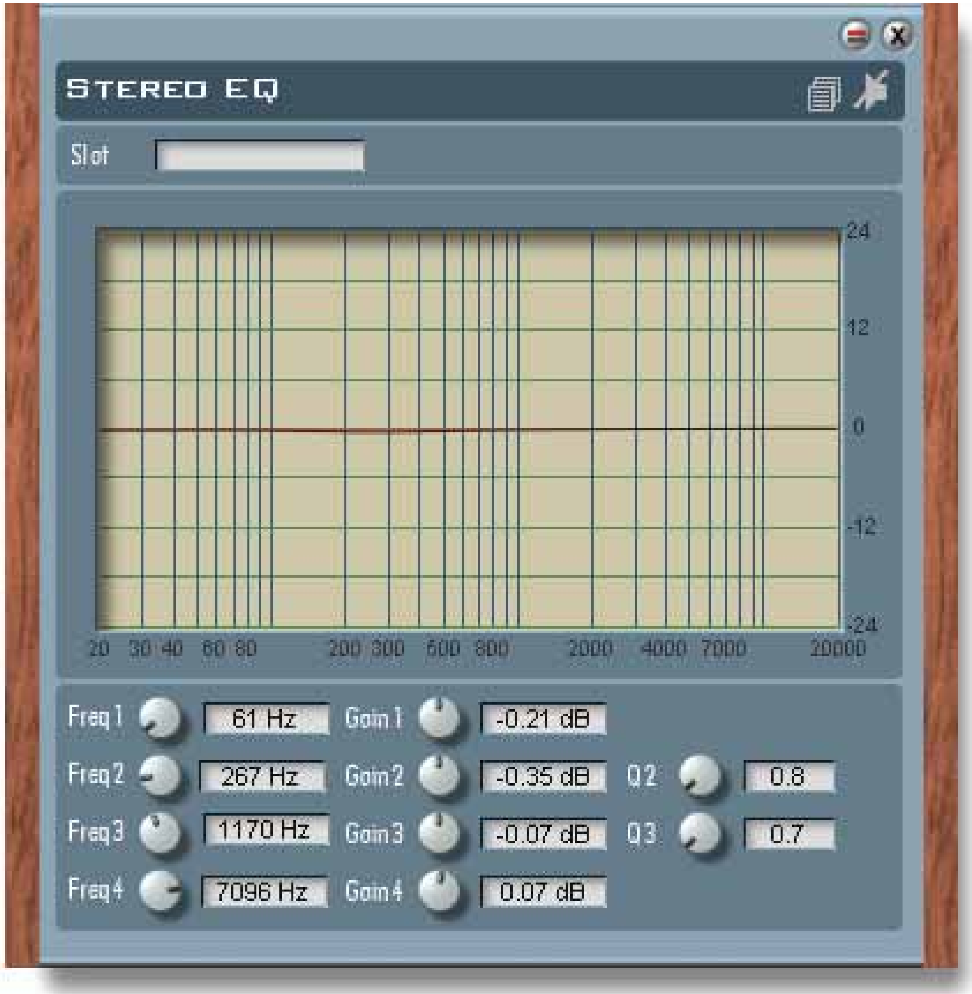
The output level of the expander can be adjusted and amplified up to +18dB.



Parametric EQ / Stereo EQ

The EQ provides four bands of equalization. Each band has an adjustable cut/boost, frequency, and Q factor. In the Stereo EQ bands 1 and 4 are implemented as high and low shelving filters and therefore do not have an adjustable Q.





Graphic Control

Inserting or Deleting a Band

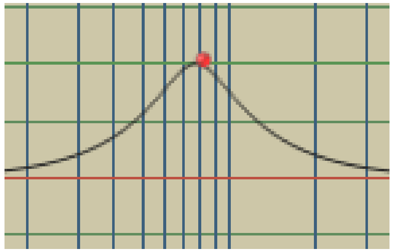
To add a new band, double click in the frequency and level graphic control area. To delete a band, double click on an existing node.

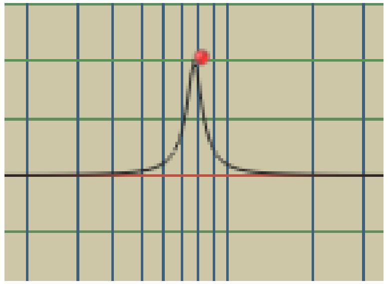
Adjusting a Band's Frequency and Cut or Boost Level

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.





Freq 1 -4

Adjusts the frequencies of the respective filter bands.

Q 1-4

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.

In the Stereo EQ, Q1 and Q4 are missing as the upper and lower bands are implemented as shelving filters.

Gain 1-4

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

Graphic EQ

This equalizer is equipped with 8 fixed-frequency bands. The range and maximum cut/ boost are globally adjustable for all bands.

-

63/125/250/500/1000/2000/4000/8000 Hz: Each control regulates a specific frequency band.

Gain: Via this input gain control, you can reduce the incoming signal level as necessary so that overloading does not occur even when specific frequency ranges are strongly boosted. This control also provides a convenient means of compensating changes in overall loudness resulting from the applied equalization.

Decimator

The Decimator lets you play a signal at a different bit resolution and/or sample rate than that of 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. The sample rate can be modulated by an LFO.

-

Sample Rate

Controls the sample rate which is used for the conversion.

(Sample Rate) Active

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

Bit Depth

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

(Bit) Active

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

LFO Rate

Sets the rate at which the sample rate is modulated by the LFO.

LFO Depth

Sets the intensity of the modulation of the sample rate by the LFO.

Hi Damp (6 dB High Damp)

This filter on the Decimator's output can be used to tone down the high-frequency content.

Distortion

This effect distorts an input signal. The signal can be processed upstream with a highpass filter and downstream with a lowpass filter. The distortion is produced by adjusting the *Drive* control and the resulting signal can be attenuated as appropriate for output with an output gain control.

Highpass

Lets you remove low frequencies from the signal before it is distorted.

Drive

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

Lowpass

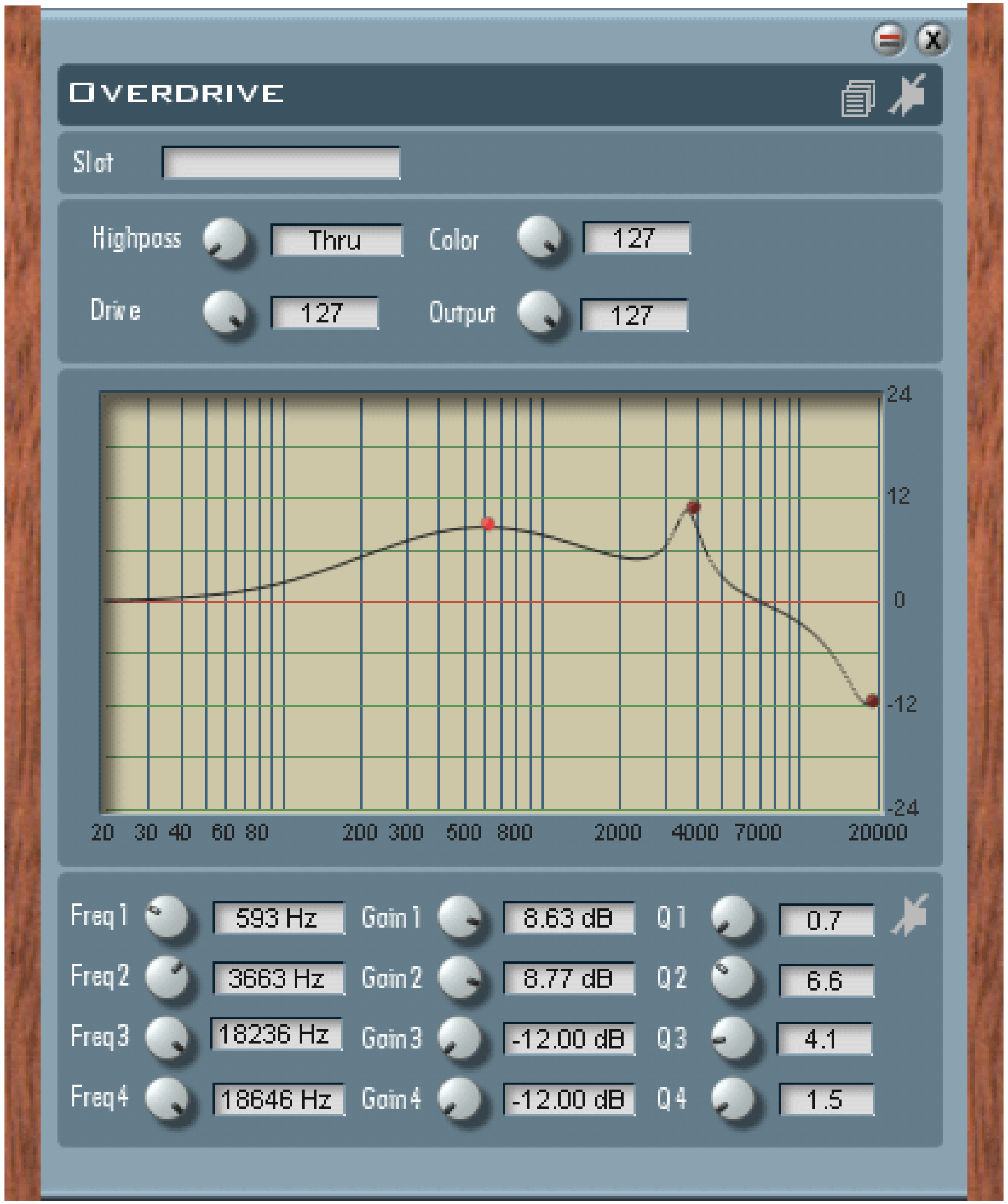
Lets you remove high frequencies from the signal after it has been processed for distortion.

Output

Because distortion processing increases the level of the signal, you can attenuate it with the output control as necessary.

Overdrive

This classic effect distorts the signal with the characteristics of an overdriven tube amplifier. A highpass filter limits the effect to the higher frequencies if desired. Because the *drive* control increases the overall signal level, an *output* control is provided at the output to attenuate the signal as required. A parametric equalizer downstream from the distortion stage allows for detailed fine-tuning of the signal.



Highpass

Use this filter to limit the distortion to the upper frequencies.

Color

Adjusts the tone quality of the tube emulator from soft to hard.

Drive

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

Output

Because distortion processing increases the level of the signal, you can attenuate it with the output gain control as necessary.

Freq 1 -4

Adjusts the frequencies of the respective filters.

Q 1-4

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 upon.

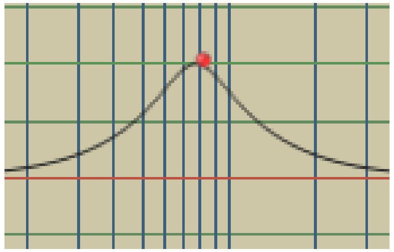
Gain 1-4

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

Graphic Control of the Equalizer

Inserting or Deleting a Band

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

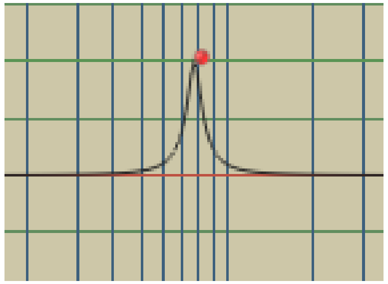


Adjusting a Band's Frequency and Cut or Boost Level

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.



AutoWah

This effect uses a multimode filter with a cutoff frequency controlled by an envelope follower. The envelope follower tracks the level of the original (pre-distortion) signal, since the distortion effect "flattens out" the signal dynamics. When the filter resonance is increased, 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.

Filter Type

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

Cutoff

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

Resonance

Sets the amount of filter resonance adjustable from 0 to 127.

Env Attack

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

Env Decay

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

Env Depth

Some signals are too low to effectively control the filter, while others are too high. Use this control to adjust the influence of the signal over the filter.

Amplifier

The Amplifier is an emulation of a tube guitar amplifier and is in no way inferior to a genuine amplifier. A sophisticated tube simulation allows for extremely realistic amplifier characteristics from a smooth, round tone to a heavily distorted overdrive. Onboard EQs extend the tonal range for the greatest possible flexibility for sound design.

Input

Sets the input volume of the amplifier. It is important to supply the amp with a sufficient level without overdriving it. Otherwise reduce the input level to avoid undesirable digital overs.

Boost

The amplifier has two virtual vacuum tubes. With Boost on, the second tube operates with a higher load in such a way as to provide more distortion.

Drive

Controls the intensity of the distortion. Higher Drive values produce higher volumes, so it is usually necessary to adjust the output **volume** to compensate. Avoid combinations of settings which cause frequent lighting of the red overload LED!

Pre EQ

The Pre EQ lets you control the proportions of the **bass, mid,** and **treble** ranges to be sent to the tube section.

Pre EQ is enabled only if the distortion switch is switched on (Aktive).

Bass

Cuts or boosts the deep bass.

Mid

Cuts or boosts the mid frequencies.

Treble

Cuts or boosts the higher mid frequencies.

PostEQ

Another EQ lies downstream from the Pre EQ to further process the sound.

Post EQ remains in effect even when the Distortion is disabled. You can therefore polish up the sound of a clean (undistorted) signal with Post EQ.

Bass

Permits boost / cut of bass frequencies below approximately 145 Hz.

Mid

Permits boost / cut of lower midrange frequencies around approximately 555 Hz.

Presence

Permits boost / cut of upper midrange frequencies around approximately 1200 Hz.

Treble

Permits boost / cut of treble frequencies above approximately 1550 Hz.

Active (PostEQ)

Permits EQ to be switched on and off for quick with / without comparison.

Keep in mind that strong boosting of the signal in all ranges simltaneously can result in internal distortion which cannot be eliminated via the output level control. However, this applies mainly for extreme combinations which are in any case not particularly useful from the sound point of view.

Volume

Sets the Amplifier's output level. If digital-sounding distortion occurs (usually accompanied by lighting of the red LED), you should reduce the level somewhat.

Distortion

Enables the tube stage.

Speaker

The guitar amp features a speaker simulator further down the chain. This produces an unsurpassed realism for the amplified sound. You can switch this option off, or use it to produce somewhat more atypical, harsh sounds (effects not necessarily imitative of classic speakers) which can be useful nonetheless.

Chorus Effects

In addition to the chorus available as an Aux effect, you can also load the following choruses as Insert effects.

The name "chorus" hints at the sound produced by this effect. It spreads and thickens the signal 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 with a delay time periodically modulated delay time. Mixing this delayed signal with the original produces the chorus effect. The intensity of the effect depends on 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. This effect is also useful for creating a stereo impression from a monaural signal.

-

Ensemble

The Ensemble Chorus is a simple chorus with only a few parameters.

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 Level

Adjusts the level of the original signal.

Wet Level

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.

Harmonic Chorus

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

Split Freq (Split Frequency)

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.

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.

Feedback

Controls the level of the feedback signal that 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.

Phase

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

LoDamp

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

HiDamp

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 a specific band of frequencies.**

Low Level

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 chorus effect.

Hi Level

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

Dry Level

Adjusts the level of the original signal.

Wet Level

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

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.

Predelay L/R

Adjusts the delay time over the range of 0 to 100 ms 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.

Feedback

Controls the level of the feedback signal that 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.

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.

LoDamp

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

HiDamp

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 Level

Adjusts the level of the original signal.

Wet Level

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.

Triple Chorus / Hexa Chorus

With the Triple or Hexa Choruses the signal is delayed not with a single delay line, but with 3 or 6 delay lines respectively. The sound of these choruses is particularly full, rich and nuanced.

-

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.

Spread (Hexa Chorus only)

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 Level

Adjusts the level of the original signal.

Wet Level

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 Effects

Although the chorus supplied as an Aux effect can be switched to Flanger mode, you can also use the following flangers provided as Insert effects.

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. This effect is also useful for creating a stereo field from a monaural signal.

Harmonic Flanger

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

Split Freq (Split Frequency)

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

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.

Feedback

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

Phase

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

LoDamp

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

HiDamp

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.

Low Level

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.

Hi Level

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 Level

Adjusts the level of the original signal.

Wet Level

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

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.

-

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.

Feedback

Controls the level of the feedback signal that 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.

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.

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 Level

Adjusts the level of the original signal.

Wet Level

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

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 that varies continuously in intensity and character.

-

PreDelay L/R

Adjusts the delay time over a range of 0 to 100 msec for both channels of the integrated stereo delay line which is located just 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 the delay time modulation in the flanger effect.

Depth

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

Feedback

Controls the level of the feedback signal that 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.

Phase Invert

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 perceived "width" 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 Level

Adjusts the level of the original signal.

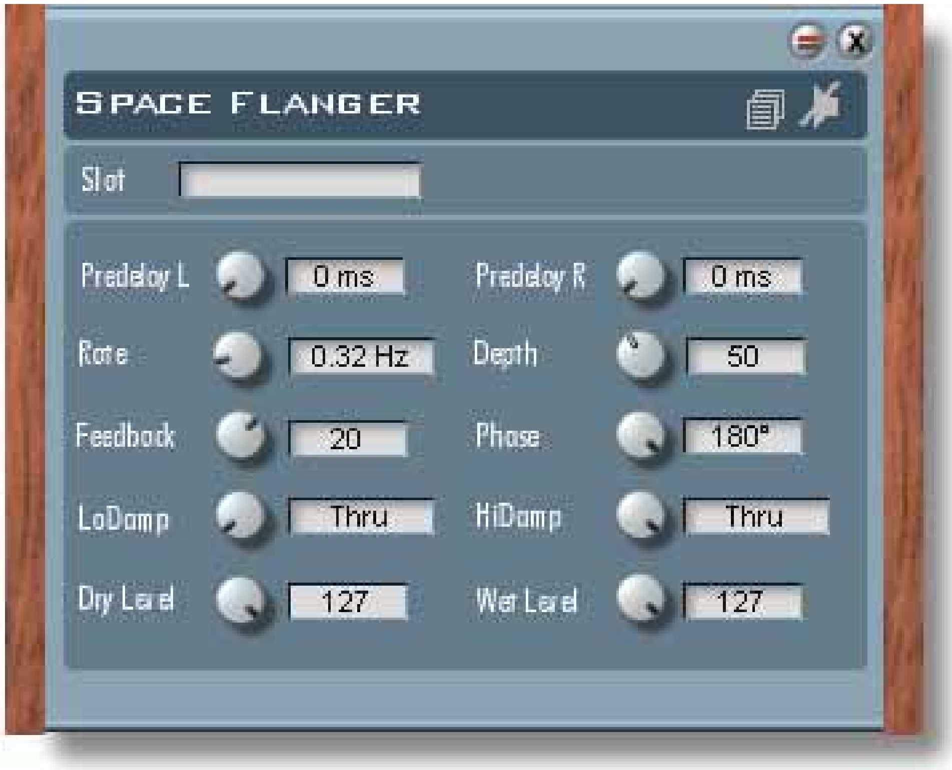
Wet Level

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

In this flanger not only is the delay position varied, but also the delay length. This lends the Space Flanger a very distinctive character.



Pre Delay 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.

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.

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.

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.

LoDamp

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

HiDamp

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 Level

Adjusts the level of the original signal.

Wet Level

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

The Step Flanger applies a sample-and-hold circuit to the modulation waveform. The resulting flanger effects range from a step-wise evolution of the comb filter through its spectrum to abrupt large jumps.

-

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 "sliced" 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.

LoDamp

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

HiDamp

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 Level

Adjusts the level of the original signal.

Wet Level

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 Phaser

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 input signal. 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 sonic 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 also useful for creating a stereo field from a monaural signal.

-

Type

Choose emulation of either a 6-stage or a 12-stage phaser. The number of stages affects the number of frequency ranges which get attenuated away. A larger number of stages results in a more nuanced sound.

Manual

Sets the base operating point of the phaser – thus, the initial positions of the attenuated frequency ranges, before modulation causes them to be shifted up and down.

Resonance

Controls the amount of feedback, and thus the intensity of the resonances and comb-filter effects which are produced by the phasing action.

Waveform

Selects the waveform used for modulation.

Rate

Controls the frequency of modulation of the phasing effect.

Depth

Adjusts the modulation depth – the range of modulation of the phasing effect.

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.

Shape

Alters the shape of the modulation signal, accelerating rising signals and slowing down falling signals.

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 audible.

What is SSB?

SSB is an abbreviation for Single Sideband and refers to a modulation process by means of which a signal's component 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 (880 + 10) is not an octave above 450 Hz (440 + 10) – the shifted frequencies are no longer harmonically related.

SSB Phaser

If SSB is used to shift a signal's frequency spectrum only slightly (by less than 1 Hz). When the shifted signal is mixed with the original, an effect not unlike a pahser is produced, but it differs significantly from a phaser in the following respect: whereas a phaser creates phase-cancellations that 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.

Frequency Shift L/R

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

Link to Left Shift

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

Shift 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 tyhe range appropriately, frequency shifting can be tuned very finely.

Feedback

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

Dry Level

Adjusts the level of the original signal.

Wet Level

Adjusts the level of the SSB effect signal.

2 Voice Pitch Shifter

In contrast to the Stereo Pitch Shifter, the 2 Voice Pitch Shifter does not pitch-shift the two parts of a stereo signal differently, but instead delivers two different shifts of the same signal.

Coarse A/B

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

Fine A/B

Adjusts the detune values in fine increments. The control range is +/- 100 cents, where 100 cents corresponds to one semitone.

Level A/B

Controls the volume level of the effect signal.

Speed

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 require more processing time.

Dry (Dry Level)

Adjusts the level of the original signal.

Wet (Wet Level)

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.

Stereo Pitch Shifter

Use the Stereo 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. The pitch for each channel can be adjusted independently, as well as the volume and pan position.

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 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.

Speed

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 require more processing time.

Dry Level

Adjusts the level of the original signal.

Wet Level

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.

Feedback Pitch Shifter

In the Feedback Pitch Shifter, an adjustable amount of the post-pitch-shift signal is fed back into the shifter and shifted again, resulting in a continually varying frequency content.

Coarse

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

Fine

Adjusts the detune value in fine increments. The control range is +/- 100 cents, where 100 cents corresponds to one semitone.

Feedback

Sets the amount of feedback – thus, the amount of processed output signal which is fed back to the input and reprocessed.

Speed

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 require more processing time.

Dry (Dry Level)

Adjusts the level of the original signal.

Wet (Wet Level)

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.

Resonator

This effect implements a comb filter with optional onboard LFO modulation.

Frequency

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

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.

Rate

Controls the frequency of the LFO (modulation rate).

Waveform

Selects the LFO waveform used to modulate the filter frequency. The following waveforms are available: Sine, Triangle.

Depth

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

Shape

Permits alteration of the waveshape of the modulation signal.

Phase

Controls the difference between the phases of the modulation applied to the two signals of a stereo channel.

Dry Level

Adjusts the level of the original signal.

Wet Level

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

Ringmodulator

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. Modulation of the sine wave frequency via an envelope is also possible.

Carrier Frequency

Controls the basic frequency of the sine wave oscillator. The range is from 1 Hz, to 1000 Hz.

LFO Waveform

Selects the waveform the LFO will use to modulate: Sine, Square, Saw Up, Saw Down, Triangle and Random.

LFO Rate

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

LFO Depth

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

Env Attack

When the sine wave is modulated by the envelope follower, this control sets the speed at which the envelope follower reacts to increasing signals.

Env Decay

When the sine wave is modulated by the envelope follower, this control sets the speed at which the envelope follower reacts to decreasing signals.

Env Depth

Sets the depth of modulation of the sine wave by the envelope follower.

RM Amount

Controls the volume of the ring modulator effect.

Tremolo

The Tremolo effect modulates the amplitude of an input signal periodically according to the waveshape selected in the *Waveform* field. The *Depth* parameter controls the degree of modulation.

Depth

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

Phase

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

Shape

Permits alteration of the waveshape of the modulation signal.

Rate

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

Waveform

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

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*.

Rate

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

Waveform

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

Depth

Controls the depth, or strength of the pan modulation.

Shape

Permits alteration of the waveshape of the modulation signal.

Phase

Controls the difference between the phases of the modulation applied to the two signals of a stereo channel.

Output

Controls the level of the effect output.

Stereo Expander

The Stereo Expander increases or decreases the perceived "width" of a stereo image. The processing is mono-compatible and frequency-independent.

Amount

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

Wide - Stereo - Mono: Here, the breadth of the stereo image resulting from the Amount setting is displayed graphically.

Tube Processor

This classic effect distorts the signal with the characteristics of an overdriven tube amplifier. A highpass filter limits the effect to the higher frequencies if desired. Because the *drive* control increases the overall signal level, an *output* control is provided at the output to attenuate the signal as required.

Color

Adjusts the tone quality of the tube emulator from soft to hard.

Lowpass (Post EQ Low)

Permits removal of high frequencies from the signal after it has been distorted.

Output

Because distortion processing increases the level of the signal, you can attenuate it with the output gain control as necessary.

Highpass

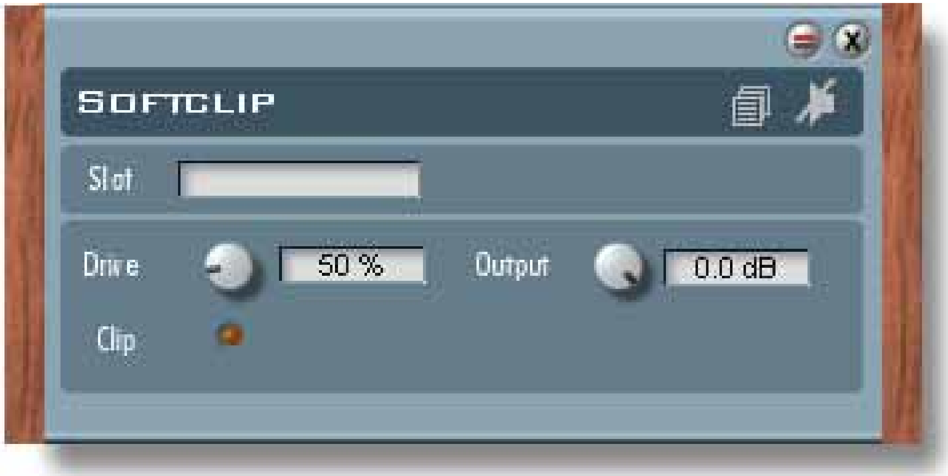
Use this filter to limit the distortion to the upper frequencies.

Drive

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

Softclip

The Softclip module permits the loudness of a signal to be boosted without introducing digital overload distorion (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.



Drive

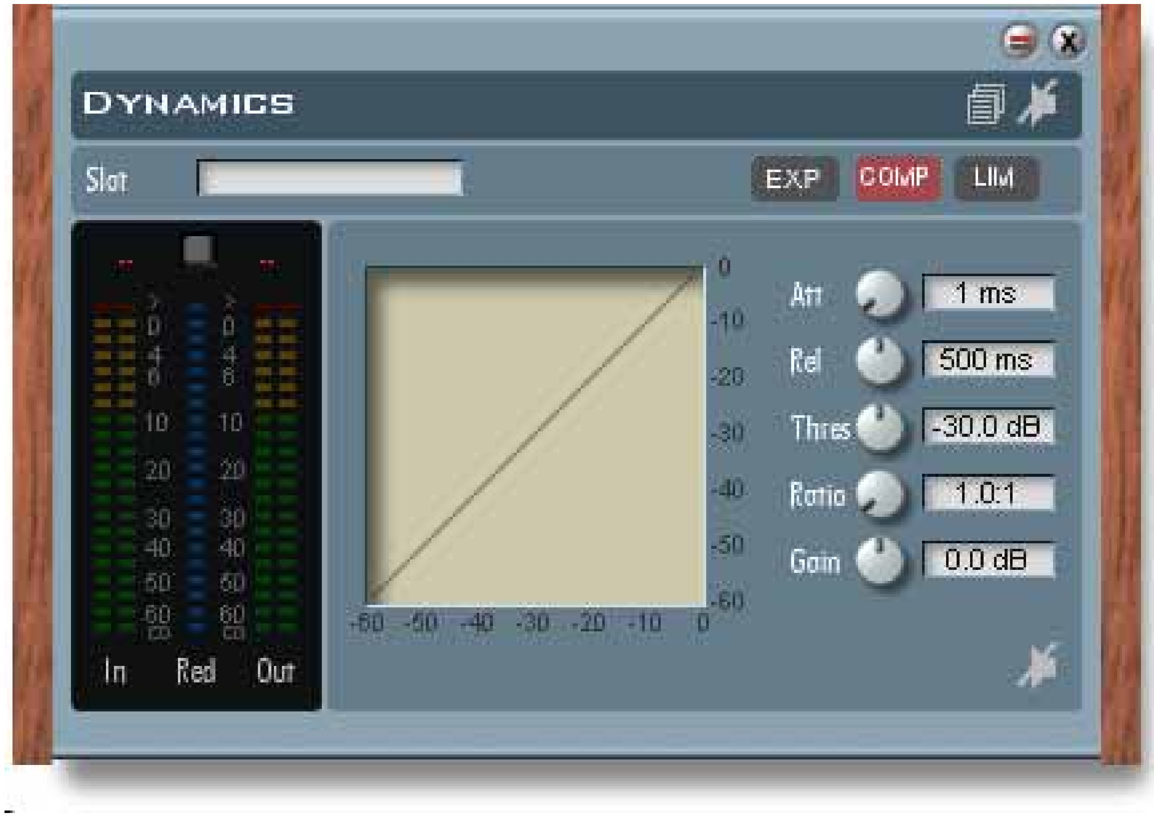
Adjusts the intensity of the effect. An orange Clip LED lights when the effect begins working.

Output

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

Dynamics

This module unites an expander, a compressor and a limiter, thereby permitting you to use all three of these effects at one time without needing to use more than one effects slot. For each of the three effects there is a control page (displayed by clicking the corresponding button) containing the effect's parameters. These parameters are identical to those of the expander, compressor and limiter effects described earlier in this section and are therefore not described here. The only difference is the Bypass option:



Bypass [On, Off]

Permits each of the three effects to be individually switched on or off.

Index

A

Amount 67, 69
Attack 22, 26, 36
Autopan 66

B

Bit on/off 32
Boost 37
Bypass-Icon 5

C

C 4
Carrier Frequency 64
Chorus 3, 5, 8, 40
Chorus (Send 2) 6
Chorus (Send1) 6
Coarse L/R 62
Color 34, 68
Compressor M/S 21
Cross Feedback 10
Cutoff 36

D

D 4
Decay 36
Decimator 32
Delay 3, 5
Delay (Send 1) 7
Delay L/C/R 11
Delay LCR 11
Depth
8, 40, 41, 43, 45, 46, 48, 50, 52, 54, 57, 63, 65, 66
Diffuse 15
Distortion 33, 39
Drive 33, 34, 37, 68
Dual Delay 12

E

E 4
Effect Source 4
Ensemble 40
Env Attack 64
Env Decay 36
Env Depth 36, 64

F

FB 14, 47, 60, 61
Feedback
10, 11, 14, 41, 44, 49, 51, 62
Feedback L/R 12
Filter Type 36
Fine L/R 60, 62
Floor 26
Freq 30, 35
Frequency 63

G

Gain 22, 24, 26, 28, 31
Graphic EQ 31

H

Hexa Chorus 45
Hi Damp 10, 11, 12, 14, 32
Hi Level 42, 47
HiDamp 42
HiDamp Filter 15
Highpass 33, 34, 68
Highpass Filter 15
Hold 26

I

Insert 4

L

Level 13
Level L/C/R 11
LFO Depth 32
LFO Rate 32
Limiter 23
Link to Left Shift 59
Lo Damp 10, 11, 14
Lo Damp L/R 12
LoDamp 42
Low Level 42, 47
Lowpass 33
Lowpass Filter 15

O

Out 25, 27
Output 33, 34, 68
Overdrive 34

P

Pan 13
Parametric EQ 29
Phase 41, 44, 47, 49, 57, 63, 65, 66
Phase Invert 51
Phaser 56
PreDelay 16

R

R 4
Rate 40, 41, 46, 56, 63, 65, 66
Ratio 22
Release 22, 24, 26, 28
Resonance 36, 63
Resonator 63
Return 4, 7
Reverb 3, 5, 15
Ringmodulator 64

S

Sample Rate 32
Send 4
Shape 16, 63
Softclip 69
Speed 60, 61, 62
Split Freq 41, 46
Spread 45
Stereo Delay 9
Stereo EQ 29
Stereo Expander 67

T

Tap 13
Threshold 26
Time 9, 11, 12
Tremolo 65
Type 8

W

Waveform 56, 63, 65, 66