Plug-in Reference

CUBASE 5

Advanced Music Production System

steinberg

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The included effect plug-ins

Introduction

This chapter contains descriptions of the included plug-in effects and their parameters.

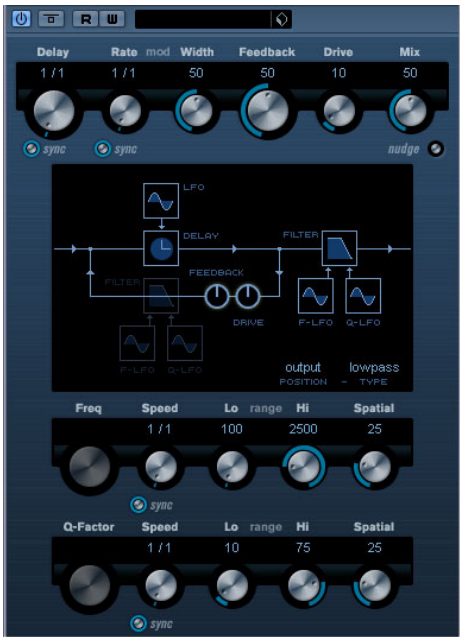
In Cubase, the plug-in effects are arranged in a number of different categories. This chapter is arranged in the same fashion, with the plug-ins listed in separate sections for each effect category.

⇨ Most of the included effects are compatible with VST3, this is indicated by an icon in front of the name of the plug-in as displayed in plug-in selection menus (for further information, see the chapter “Audio Effects” in the Operation Manual).

Delay plug-ins

This section contains descriptions of the plug-ins in the “Delay” category.

ModMachine (Cubase only)



ModMachine combines delay modulation and filter frequency/resonance modulation and can provide many interesting modulation effects. It also features a Drive parameter for distortion effects.

The parameters are as follows:

Parameter	Description
Delay	This is where you specify the base note value for the delay if tempo sync is on (1/1–1/32, straight, triplet or dotted). If tempo sync is off, the delay time can be set freely in milliseconds.
Tempo sync Delay on/off	The button below the Delay knob turns tempo sync for the delay parameter on or off. If set to off, the delay time can be set freely with the Delay knob.
Rate	The Rate parameter sets the base note value for tempo syncing the delay modulation (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the rate can be set freely with the Rate knob.

Parameter	Description
Tempo sync Rate on/off	The button below the Rate knob turns tempo sync for the rate parameter on or off. If set to off, the rate can be set freely with the Rate knob.
Width	This sets the amount of delay pitch modulation. Note that although the modulation affects the delay time, the sound is mostly perceived as a vibrato or chorus-like effect.
Feedback	This sets the number of repeats for the delay.
Drive	This parameter adds distortion to the feedback loop. The longer the Feedback, the more the delay repeats become distorted over time.
Mix	Sets the level balance between the dry signal and the effect. If ModMachine is used as a send effect, this should be set to maximum (100%) as you can control the dry/effect balance with the send.
Nudge	Clicking the Nudge button once will momentarily speed up the audio coming into the plug-in, simulating an analog tape nudge type sound effect.
Signal path graphic	You can click on the Filter sections displayed in the graphic in the center of the plug-in to place the Filter section either before or after the Drive and Feedback parameters in the signal path.
Output/Loop	The Filter can either be placed in the feedback loop of the delay or in its output path (see above).
Filter type	This toggle button allows you to select a filter type. Low-pass/bandpass/hipass filter types are available.
Freq	This sets the cutoff frequency for the filter. This is only available if filter frequency LFO tempo sync is deactivated and the Speed parameter (see below) is set to “0”.
Speed	This sets the speed of the filter frequency LFO modulation. If tempo sync is activated the Speed parameter sets the base note value for tempo syncing the modulation (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the rate can be set freely with the Speed knob.
Range Lo/Hi	These knobs specify the range (in Hz) of the filter frequency modulation. Both positive (e.g. Lo set to 50 and Hi set to 10000) and negative (e.g. Lo set to 5000 and Hi set to 500) ranges can be set. If tempo sync is off and the Speed is set to zero, these parameters are inactive and the filter frequency is instead controlled by the Freq parameter.
Spatial	This introduces an offset between the channels to create a stereo panorama effect for the filter frequency modulation. Turn clockwise for a more pronounced stereo effect.
Q-Factor	This controls the resonance of the filter. This is only available if filter resonance LFO tempo sync is deactivated and the Speed parameter (see below) is set to “0”. If tempo sync is on, the resonance is controlled by the Speed and Range parameters.
Speed	This sets the speed of the filter resonance LFO modulation. If tempo sync is activated, the Speed parameter sets the base note value for tempo syncing the modulation (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the rate can be set freely with the Speed knob.
Range Lo/Hi	These knobs specify the range of filter resonance modulation. Both positive (e.g. Lo set to 50 and Hi set to 100) and negative (e.g. Lo set to 100 and Hi set to 50) ranges can be set. If tempo sync is off and the Speed is set to zero, these parameters are inactive and the filter resonance is controlled by the Q-Factor parameter instead.
Spatial	This introduces an offset between the channels to create a stereo panorama effect for the filter resonance modulation. Turn clockwise for a more pronounced stereo effect.

MonoDelay

This is a mono delay effect that can either be tempo-based or use freely specified delay time settings. The delay can also be controlled from another signal source via the Side-Chain input.

The parameters are as follows:

Parameter	Description
Delay	This is where you specify the base note value for the delay if tempo sync is on (1/1–1/32, straight, triplet or dotted). If tempo sync is off, it sets the delay time in milliseconds.
Tempo sync on/off	The button below the Delay Time knob is used to turn tempo sync on or off. If set to off, the delay time can be set freely with the Delay Time knob, without sync to tempo.
Feedback	This sets the number of repeats for the delay.
Filter Lo	This filter affects the feedback loop of the effect signal and allows you to roll off low frequencies from 10 Hz up to 800 Hz. The button below the knob activates/deactivates the filter.
Filter Hi	This filter affects the feedback loop of the effect signal and allows you to roll off high frequencies from 20 kHz down to 1.2 kHz. The button below the knob activates/deactivates the filter.

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If MonoDelay is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Side-Chain on/off	When this is activated, the delay can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold the delay repeats are silenced. When the signal drops below the threshold the delay repeats reappear. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

PingPongDelay

This is a stereo delay effect that alternates each delay repeat between the left and right channels. The effect can either be tempo-based or use freely specified delay time settings.

The parameters are as follows:

Parameter	Description
Delay	This is where you specify the base note value for the delay if tempo sync is on (1/1–1/32, straight, triplet or dotted). If tempo sync is off, it sets the delay time in milliseconds.
Tempo sync on/off	The button below the Delay Time knob is used to turn tempo sync on or off. If set to off, the delay time can be set freely with the Delay Time knob, without sync to tempo.
Feedback	This sets the number of repeats for the delay.
Filter Lo	This filter affects the feedback loop and allows you to roll off low frequencies up to 800Hz. The button below the knob activates/deactivates the filter.
Filter Hi	This filter affects the feedback loop and allows you to roll off high frequencies from 20kHz down to 1.2kHz. The button below the knob activates/deactivates the filter.
Spatial	This parameter sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo “ping-pong” effect.

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If PingPongDelay is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Side-Chain on/off	When this is activated, the delay can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

StereoDelay

StereoDelay has two independent delay lines which either use tempo-based or freely specified delay time settings.

The parameters are as follows:

Parameter	Description
Delay 1	This is where you specify the base note value for the delay, if tempo sync is on (1/1–1/32, straight, triplet or dotted). If tempo sync is off, it sets the delay time in milliseconds.
Delay 2	As above.
Tempo sync on/off	The buttons below each respective Delay knob are used to turn tempo sync on or off for the respective delay. If set to off, the delay time can be set freely with the Delay Time knobs.
Feedback 1 & 2	This sets the number of repeats for each delay.
Filter Lo 1 & 2	This filter affects the feedback loop and allows you to roll off low frequencies up to 800Hz. The button below the knob activates/deactivates the filter.
Filter Hi 1 & 2	This filter affects the feedback loop and allows you to roll off high frequencies from 20kHz down to 1.2kHz. The button below the knob activates/deactivates the filter.
Pan1 & 2	This sets the stereo position for each delay.
Mix	Sets the level balance between the dry signal and the effect. If StereoDelay is used as a send effect, this should be set to maximum (100%) as you can control the dry/effect balance with the send.
Side-Chain on/off	When this is activated, the delay can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

Distortion plug-ins

This section contains descriptions of the plug-ins in the “Distortion” category.

AmpSimulator

AmpSimulator is a distortion effect, emulating the sound of various types of guitar amp and speaker cabinet combinations. A wide selection of amp and cabinet models is available.

The parameters are as follows:

Parameter	Description
Drive	Governs the amount of amp overdrive.
Bass	Tone control for the low frequencies.
Middle	Tone control for the mid frequencies.
Treble	Tone control for the high frequencies.
Presence	Use this to boost or damp the higher frequencies.
Volume	This controls the overall output level.
Amplifier	This allows you to select between various amplifier models. Click on the currently selected amplifier name to open a pop-up with all the available amplifier models. This section can be bypassed by selecting “No Amp”.
Cabinet	Various speaker cabinet models. Click on the currently selected cabinet name to open a pop-up with all the available amplifier models. This section can be bypassed by selecting “No Speaker”.
Damping Lo/Hi	Further tone controls for shaping the sound of the selected speaker cabinet. Click on the values, enter a new value and press the [Enter] key.

DaTube

This effect emulates the characteristic warm, lush sound of a tube amplifier.

The parameters are as follows:

Parameter	Description
Drive	Regulates the pre-gain of the “amplifier”. Use high values if you want an overdriven sound just on the verge of distortion.
Balance	This controls the balance between the signal processed by the Drive parameter and the dry input signal. For maximum drive effect, set this to its highest value.
Output	Adjusts the post-gain, or output level, of the “amplifier”.

Distortion

Distortion will add crunch to your tracks.

The parameters are as follows:

Parameter	Description
Boost	Increases the distortion amount.
Feedback	This parameter feeds part of the output signal back to the effect input, increasing the distortion effect.
Tone	Lets you select a frequency range to which to apply the distortion effect.
Spatial	Changes the distortion characteristics of the left and right channel, thus creating a stereo effect.
Output	Raises or lowers the signal going out of the effect.

SoftClipper (Cubase only)

This effect adds soft overdrive, with independent control over the second and third harmonic.

The parameters are as follows:

Parameter	Description
Input	Regulates the pre-gain. Use high values if you want an overdriven sound just on the verge of distortion.
Mix	Setting Mix to 0 means that no processed signal is added to the original signal.
Output	Adjusts the post-gain, or output level.
Second	This allows you to adjust the amount of the second harmonic in the processed signal.
Third	This allows you to adjust the amount of the third harmonic in the processed signal.

Dynamics plug-ins

This section contains descriptions of the plug-ins in the “Dynamics” category.

Compressor

Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. Compressor features separate controls for threshold, ratio, attack, hold, release and make-up gain parameters. Compressor features a separate display that graphically illustrates the compressor curve shaped according to the Threshold and Ratio parameter settings. Compressor also features a Gain Reduction meter that shows the amount of gain reduction in dB, Soft knee/Hard knee compression modes and a program-dependent Auto feature for the Release parameter.

The available parameters work as follows:

Parameter	Description
Threshold (-60 to 0 dB)	This setting determines the level where Compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1:1 to 8:1)	Ratio determines the amount of gain reduction applied to signals over the set threshold. A ratio of 3:1 means that for every 3 dB the input level increases, the output level will increase by only 1 dB.
Soft Knee (On/Off)	If this is off, signals above the threshold will be compressed instantly according to the set ratio (hard knee). When Soft Knee is activated, the onset of compression will be more gradual, producing a less drastic result.
Make-up (0–24 dB or “Auto mode”)	This parameter is used to compensate for output gain loss, caused by compression. If the Auto button is activated, the knob becomes dark and the output is instead automatically adjusted for gain loss.
Attack (0.1–100 ms)	This determines how fast Compressor will respond to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
Hold (0–2000 ms)	Sets the time the applied compression will affect the signal after exceeding the Threshold.
Release (10–1000 ms or “Auto mode”)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the Threshold level. If the “Auto” button is activated, Compressor will automatically find an optimal release setting that varies depending on the audio material.
Analysis (0–100) (Pure Peak to Pure RMS)	This parameter determines whether the input signal is analysed according to peak or RMS values (or a mixture of both). A value of 0 is pure peak and 100 pure RMS. RMS mode operates using the average power of the audio signal as a basis, whereas Peak mode operates more on peak levels. As a general guideline, RMS mode works better on material with few transients such as vocals, and Peak mode better for percussive material, with a lot of transient peaks.
Live mode (On/Off)	When activated, Live mode disengages the “look ahead” feature of the Compressor. Look ahead does produce more accurate processing but will add a certain amount of latency as a trade-off. When Live mode is activated, there is no latency, which might be better for “live” processing.
Side-Chain (On/Off)	When this is activated, the compression can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold, the compression is triggered. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

DeEsser (Cubase only)

A de-esser is used to reduce excessive sibilance, primarily for vocal recordings. Basically, it is a special type of compressor that is tuned to be sensitive to the frequencies produced by the “s” sound, hence the name de-esser. Close proximity microphone placement and equalizing can lead to situations where the overall sound is just right, but there is a problem with sibilants. Conventional compression and/or equalizing will not easily solve this problem, but a de-esser can.

The SPL DeEsser has the following parameters:

Parameter	Description
S-Reduction	Controls the intensity of the de-essing effect. We recommend that you start with a value between 4 and 7.
Level display	Indicates the dB value by which the level of the sibilant or s-frequency is reduced. The display shows values between 0 dB (no reduction) and minus 20 dB (the s-frequency level is lowered by 20 dB). Each segment in the display represents a level reduction of 2 dB.
Auto Threshold	See separate description below.
Male/Female	This sets the s-frequency and sibilant recognition to the characteristic frequency ranges of the female or male voice. The center frequency of the bandwidth at which the SPL DeEsser operates is located in the 7 kHz range for the female voice and in the 6 kHz range for the male voice.

About the Auto Threshold function

Conventional de-essing devices all have a threshold parameter. This is used to set a threshold for the incoming signal level, above which the device starts to process the signal. The SPL DeEsser however has been designed for utmost ease-of-use. With Auto Threshold on (the button lights up) it automatically and constantly readjusts the threshold to achieve an optimum result. If you still wish to determine for yourself at which signal level the SPL DeEsser should start to process the signal, deactivate the Auto Threshold button. The SPL DeEsser will then use a fixed threshold.

When recording a voice, usually the de-esser's position in the signal chain is located after the microphone pre-amp and before a compressor/limiter. This is useful, as it keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics by reacting to excessive sibilants and s-frequencies.

The Auto Threshold function keeps the processing on a constant level. The input threshold value is automatically and constantly adjusted to the audio input level. Even level differences of say 20 dB do not have a negative impact on the result of the processing. The input levels may vary, but processing remains constant.

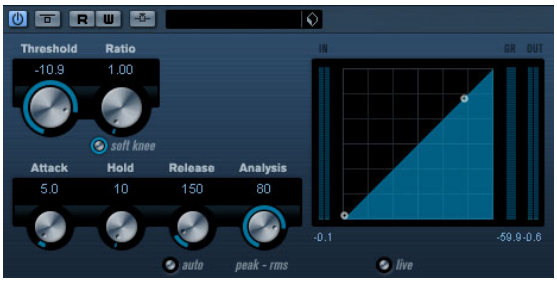
EnvelopeShaper (Cubase only)

EnvelopeShaper can be used to cut or boost the gain of the Attack and Release phase of the audio material. You can either use the knobs or drag the breakpoints in the graphic display to change parameter values. Be careful with levels when boosting the gain and if needed reduce the Output level to avoid clipping.

The following parameters are available:

Parameter	Description
Attack (-20–20 dB)	Changes the gain of the Attack phase of the signal.
Length (5–200 ms)	This determines the length of the Attack phase.
Release (-20–20 dB)	Changes the gain of the Release phase of the signal.
Output (-24–12 dB)	Sets the output level.

Expander (Cubase only)



Expander reduces the output level in relation to the input level for signals below the set threshold. This is useful, when you want to enhance the dynamic range or reduce the noise in quiet passages. You can either use the knobs or drag the breakpoints in the graphic display to change the Threshold and the Ratio parameter values.

The following parameters are available:

Parameter	Description
Threshold (-60–0dB)	This setting determines the level where expansion “kicks in”. Signal levels below the set threshold are affected, but signal levels above are not processed.
Ratio (1:1–8:1)	Ratio determines the amount of gain boost applied to signals below the set threshold.
Soft Knee (On/Off)	If this is off, signals below the threshold will be expanded instantly according to the set ratio (“hard knee”). When Soft Knee is activated, the onset of expansion will be more gradual, producing a less drastic result.
Attack (0.1–100ms)	This determines how fast Expander will respond to signals below the set threshold. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
Hold (0–2000ms)	Sets the time the applied expansion will affect the signal below the Threshold.
Release (10–1000ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level when the signal exceeds the Threshold level. If the “Auto” button is activated, Expander will automatically find an optimal release setting that varies depending on the audio material.
Analysis (0–100) (Pure Peak to Pure RMS)	This parameter determines whether the input signal is analysed according to peak or RMS values (or a mixture of both). A value of 0 is pure peak and 100 pure RMS. RMS mode operates using the average power of the audio signal as a basis, whereas Peak mode operates more on peak levels. As a general guideline, RMS mode works better on material with few transients such as vocals, and Peak mode better for percussive material, with a lot of transient peaks.
Live mode (On/Off)	When activated, Live mode disengages the look ahead feature of Expander. Look ahead does produce more accurate processing but will add a certain amount of latency as a trade-off. When Live mode is activated, there is no latency.
Side-Chain (On/Off)	When this is activated, the expansion can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold, the expansion is triggered. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

Gate

Gating, or noise gating, silences audio signals below a certain set threshold level. As soon as the signal level exceeds the set threshold, the gate opens to let the signal through.

The available parameters are as follows:

Parameter	Description
Threshold (-60–0dB)	This setting determines the level where Gate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold will close the gate.
State LED	This indicates whether the gate is open (LED lights up in green), closed (LED lights up in red) or something in between (LED lights up in yellow).
Filter buttons	When the Side-Chain button (see below) is activated, you can use these buttons to set the filter type to either Low Pass, Band Pass or High Pass.
Side-Chain (On/Off)	This button (below the Center knob) activates the filter. The input signal can then be shaped according to set Center and Q-Factor parameters which may be useful in tailoring how the Gate operates.
Center (50Hz–20000Hz)	Sets the center frequency of the filter.

Parameter	Description
Q-Factor (0.01–10000)	Sets the Resonance of the filter.
Monitor (On/Off)	Allows you to monitor the filtered signal.
Attack (0.1–1000 ms)	This parameter sets the time it takes for the gate to open after being triggered. If the Live button (see below) is deactivated, it will ensure that the gate will already be open when a signal above the threshold level is played back. Gate manages this by “looking ahead” in the audio material, checking for signals loud enough to pass the gate.
Hold (0–2000 ms)	This determines how long the gate stays open after the signal drops below the threshold level.
Release (10–1000 ms or “Auto”)	This parameter sets the amount of time it takes for the gate to close (after the set hold time). If the “Auto” button is activated, Gate will find an optimal release setting, depending on the audio program material.
Analysis (0–100) (Pure Peak to Pure RMS)	This parameter determines whether the input signal is analysed according to Peak or RMS values (or a mixture of both). A value of 0 is pure Peak and 100 pure RMS. RMS mode operates using the average power of the audio signal as a basis, whereas Peak mode operates more on peak levels. As a general guideline, RMS mode works better on material with few transients such as vocals, and Peak mode better for percussive material, with a lot of transient peaks.
Live mode (On/Off)	When activated, Live mode disengages the “look ahead” feature of the Gate. Look ahead does produce more accurate processing but will add a certain amount of latency as a trade-off. When Live mode is activated, there is no latency, which might be better for “live” processing.

Limiter

Limiter is designed to ensure that the output level never exceeds a certain set output level, to avoid clipping in following devices. Limiter can adjust and optimize the Release parameter automatically according to the audio material, or it can be set manually. Limiter also features separate meters for the input, output and the amount of limiting (middle meters).

The available parameters are the following:

Parameter	Description
Input (-24–+24 dB)	Allows you to adjust the input gain.
Output (-24–+6 dB)	This setting determines the maximum output level.
Release (0.1–1000 ms or Auto mode)	This parameter sets the amount of time it takes for the gain to return to its original level. If the “Auto” button is activated, Limiter will automatically find an optimal release setting that varies depending on the audio material.

Maximizer

Maximizer can be used to raise the loudness of audio material without the risk of clipping. Optionally, there is a soft clip function that removes short peaks in the input signal and introduces a warm tubelike distortion to the signal.

The available parameters are the following:

Parameter	Description
Output (-24–+6 dB)	This setting determines the maximum output level. Should normally be set to 0 (to avoid clipping).
Optimize (0–100)	This setting determines the loudness of the signal.
Soft Clip (On/Off)	Soft Clipper starts limiting (or clipping) the signal “softly”, at the same time generating harmonics which add a warm, tubelike characteristic to the audio material.

MIDI Gate

Gating, in its fundamental form, silences audio signals below a certain set threshold level. That means, when a signal rises above the set level, the Gate opens to let the signal through while signals below the set level are cut off. MIDI Gate, however, is a Gate effect that is not triggered by threshold levels, but instead by MIDI notes. Hence it needs both audio and MIDI data to function.

Setting up

MIDI Gate requires both an audio signal and a MIDI input to function.

To set it up, proceed as follows:

1. Select the audio to be affected by the MIDI Gate.
This can be audio material from any audio track, or even a live audio input (provided you have a low latency audio card).

2. Select the MIDI Gate as an insert effect for the audio track.
The MIDI Gate control panel opens.

3. Select a MIDI track to control the MIDI Gate.
This can be an empty MIDI track, or a MIDI track containing data, it doesn't matter. However, if you wish to play the MIDI Gate in real-time – as opposed to having a recorded part playing it – the track has to be selected for the effect to receive the MIDI output.

4. Open the Output Routing pop-up menu for the MIDI track and select the MIDI Gate option.
The MIDI Output from the track is now routed to the MIDI Gate.

What to do next depends on whether you are using live or recorded audio and whether you are using real-time or recorded MIDI. We will assume for the purposes of this manual that you are using recorded audio, and play the MIDI in real-time.

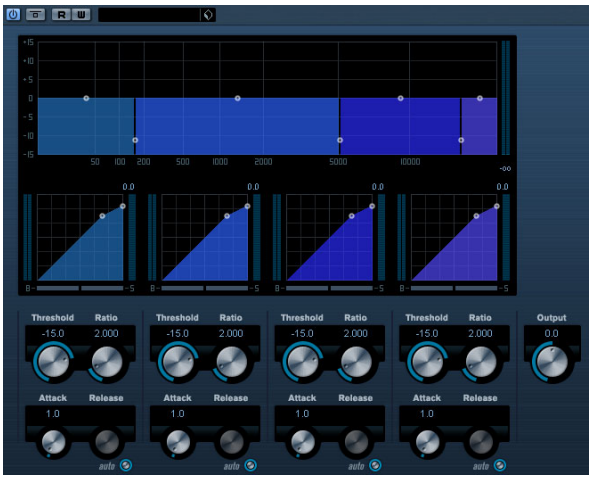
Make sure the MIDI track is selected and start playback.

5. Now play a few notes on your MIDI keyboard.
As you can hear, the audio track material is affected by what you play on your MIDI keyboard.

The following MIDI Gate parameters are available:

Parameter	Description
Attack	This is used for determining how long it should take for the Gate to open after receiving a signal that triggers it.
Hold	Regulates how long the Gate remains open after a Note On or Note Off message (see Hold Mode below).
Release	This determines how long it takes for the Gate to close (in addition to the value set with the Hold parameter).
Note To Attack	The value you specify here determines to which extent the velocity values of the MIDI notes should affect the Attack. The higher the value, the more the Attack time will increase with high note velocities. Negative values will give shorter Attack times with high velocities. If you do not wish to use this parameter, set it to the 0 position.
Note To Release	The value you specify here determines to which extent the velocity values of the MIDI notes should affect the Release. The higher the value, the more the Release time will increase. If you do not wish to use this parameter, set it to the 0 position.
Velocity To VCA	This controls to which extent the velocity values of the MIDI notes determine the output volume. A value of 127 means that the volume is controlled entirely by the velocity values, while a value of 0 means that velocities will have no effect on the volume.
Hold Mode	Use this switch to set the Hold Mode. In Note-On mode, the Gate will only remain open for the time set with the Hold and Release parameters, regardless of the length of the MIDI note that triggered the Gate. In Note-Off mode on the other hand, the Gate will remain open for as long as the MIDI note plays, and then apply the Hold and Release parameters.

MultibandCompressor (Cubase only)



The MultibandCompressor allows a signal to be split in up to four frequency bands, each with its own freely adjustable compressor characteristic. The signal is processed on the basis of the settings that you have made in the Frequency Band and Compressor sections. You can specify the level, bandwidth and compressor characteristics for each band by using the various controls.

The Frequency Band editor

The Frequency Band editor in the upper half of the panel is where you set the width of the frequency bands as well as their level after compression. Two value scales and a number of handles are available. The vertical value scale to the left shows the input gain level of each frequency band. The horizontal scale shows the available frequency range.

The handles provided in the Frequency Band editor can be dragged with the mouse. You use them to set the corner frequency range and the input gain levels for each frequency bands.

- The handles at the sides are used to define the frequency range of the different frequency bands.
- By using the handles on top of each frequency band, you can cut or boost the input gain by +/- 15dB after compression.

Bypassing frequency bands

Each frequency band can be bypassed using the “B” button in each compressor section.

Soloing frequency bands

A frequency band can be soloed using the “S” button in each compressor section. Only one band can be soloed at a time.

Using the Compressor section

By moving breakpoints or using the corresponding knobs, you can specify the Threshold and Ratio. The first breakpoint from which the line deviates from the straight diagonal will be the threshold point. The compressor parameters for each of the four bands are as follows:

Parameter	Description
Threshold (-60–0dB)	This setting determines the level where Compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1000–8000) (1:1 to 8:1)	Ratio determines the amount of gain reduction applied to signals over the set threshold. A ratio of 3000 (3:1) means that for every 3dB the input level increases, the output level will increase by only 1dB.
Attack (0.1–100ms)	This determines how fast the compressor will respond to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
Release (10–1000ms or “Auto”)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the Threshold level. If the “Auto” button is activated, the compressor will automatically find an optimal release setting that varies depending on the audio material.

The Output dial

The Output dial controls the total output level that the MultibandCompressor passes on to Cubase. The range available is +/- 24dB.

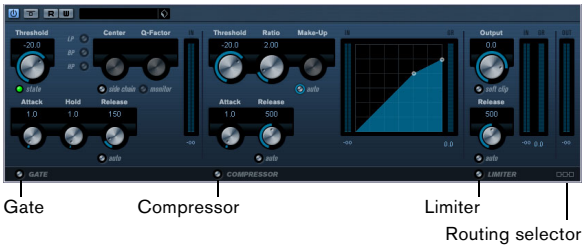
VintageCompressor (Cubase only)

This is modelled after vintage type compressors. Compressor features separate controls for input gain, attack, release and output gain parameters. In addition, there is a Punch mode which preserves the attack phase of the signal and a program dependent Auto feature for the Release parameter.

The available parameters work as follows:

Parameter	Description
Input gain (-24–48dB)	This setting, together with the Output gain parameter determines the compression amount. The higher the Input gain setting, and the lower the Output gain setting, the more compression is applied.
Output gain (-48–24dB)	Sets the output gain.
Attack (0.1–100ms)	This determines how fast Compressor will respond. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
Punch (On/Off)	When this is activated, the early attack phase of the signal is preserved, retaining the original “punch” in the audio material, even with short Attack settings.
Release (10–1000ms or “Auto mode”)	Sets the amount of time it takes for the gain to return to its original level. If the “Auto” button is activated, Vintage Compressor will automatically find an optimal release setting that varies depending on the audio material.
Side-Chain (On/Off)	When this is activated, the compression can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold the compression is triggered. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

VSTDynamics



VSTDynamics is an advanced dynamics processor. It combines three separate processors: Gate, Compressor and Limiter, covering a variety of dynamic processing functions. The window is divided into three sections, containing controls and meters for each processor.

Activating the individual processors

You activate the individual processors using the buttons at the bottom of the plug-in panel.

The Gate section

Gating, or noise gating, is a method of dynamic processing that silences audio signals below a certain set threshold level. As soon as the signal level exceeds the set threshold, the gate opens to let the signal through. The Gate trigger input can also be filtered using an internal side-chain.

The available parameters are as follows:

Parameter	Description
Threshold (-60–0dB)	This setting determines the level where Gate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold will close the gate.
state	This indicates whether the gate is open (LED lights up in green), closed (LED lights up in red) or something in between (LED lights up in yellow).
Side-Chain (On/Off)	This button activates the internal side-chain filter. This lets you filter out parts of the signal that might otherwise trigger the gate in places you don’t want it to, or to boost frequencies you wish to accentuate, allowing for more control over the gate function.
LP (Lowpass), BP (Bandpass), HP (Highpass)	These buttons set the basic filter mode.
Center (50–22000Hz)	This sets the center frequency of the filter.

Parameter	Description
Q-Factor (0.001–10000)	This sets the resonance or width of the filter.
Monitor (On/Off)	Allows you to monitor the filtered signal.
Attack (0.1–100 ms)	This parameter sets the time it takes for the gate to open after being triggered.
Hold (0–2000 ms)	This determines how long the gate stays open after the signal drops below the threshold level.
Release (10–1000 ms or “Auto”)	This parameter sets the amount of time it takes for the gate to close (after the set hold time). If the “Auto” button is activated, Gate will find an optimal release setting, depending on the audio program material.

The Compressor section

Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. Compressor functions like a standard compressor with separate controls for threshold, ratio, attack, release and make-up gain parameters. Compressor features a separate display that graphically illustrates the compressor curve shaped according to the Threshold, Ratio and MakeUp Gain parameter settings. Compressor also features a Gain Reduction meter that shows the amount of gain reduction in dB, and a program dependent Auto feature for the Release parameter.

The available parameters work as follows:

Parameter	Description
Threshold (-60–0 dB)	This setting determines the level where Compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1:1–8:1)	Ratio determines the amount of gain reduction applied to signals over the set threshold. A ratio of 3:1 means that for every 3 dB the input level increases, the output level will increase by only 1 dB.
Make-Up (0–24 dB)	This parameter is used to compensate for output gain loss, caused by compression. When Auto is on, gain loss will be compensated automatically.
Attack (0.1–100 ms)	This determines how fast Compressor will respond to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
Release (10–1000 ms or “Auto”)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the Threshold level. If the “Auto” button is activated, Compressor will automatically find an optimal release setting that varies depending on the audio material.
Graphic display	Use the graphic display to graphically set the Threshold or the Ratio value.

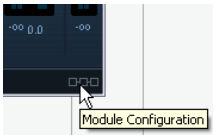
The Limiter section

Limiter is designed to ensure that the output level never exceeds a certain set output level, to avoid clipping in following devices. Conventional limiters usually require very accurate setting up of the attack and release parameters, to prevent the output level from going beyond the set threshold level. Limiter adjusts and optimizes these parameters automatically, according to the audio material. You can also adjust the Release parameter manually.

The available parameters are the following:

Parameter	Description
Output (-24–+6 dB)	This setting determines the maximum output level. Signal levels above the set threshold are affected, but signal levels below are left unaffected.
Soft Clip (On/Off)	Soft Clipper acts differently compared to the limiter. When the signal level exceeds -6 dB, SoftClip starts limiting (or clipping) the signal “softly”, at the same time generating harmonics which add a warm, tubelike characteristic to the audio material.
Release (10–1000 ms or “Auto”)	This parameter sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the “Auto” button is activated, Limiter will automatically find an optimal release setting that varies depending on the audio material.

The Module Configuration button



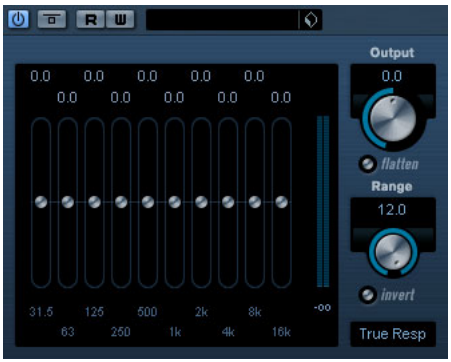
In the bottom right corner of the plug-in panel you will find a button with which you can set the signal flow order for the three processors. Changing the order of the processors can produce different results, and the available options allow you to quickly compare what works best for a given situation. Simply click the Module Configuration button to change to a different configuration. There are three routing options:

- C-G-L (Compressor-Gate-Limit)
- G-C-L (Gate-Compressor-Limit)
- C-L-G (Compressor-Limit-Gate)

EQ plug-ins

This section describes the plug-ins in the “EQ” category.

GEQ-10/GEQ-30 (Cubase only)



These graphic equalizers are identical in every respect except for the number of available frequency bands (10 and 30 respectively). Each band can be cut or boosted by up to 12 dB allowing for fine control of the frequency response. In addition there are several preset modes available which can add “color” to the sound of the GEQ-10/GEQ-30.

- You can draw response curves in the main display by click-dragging with the mouse.

  Note that you have to click on one of the sliders first before dragging across the display. You can also point and click to change individual frequency bands or enter values numerically by clicking on a gain value at the top of the display.

- At the bottom of the window the respective frequency bands are shown in Hz.

- At the top of the display, the amount of cut/boost is shown in dB.

Apart from the frequency bands, the following parameters are available:

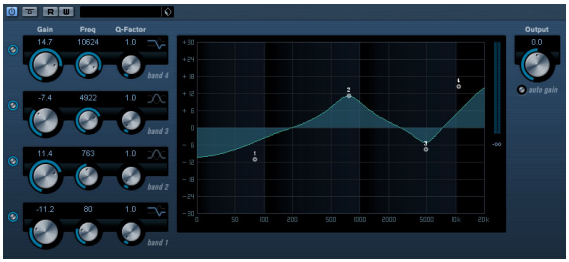
Parameter	Description
Output	This controls the overall gain of the equalizer.
Range	This allows you to relatively adjust how much a set curve cuts or boosts the signal. If the Range parameter is turned fully clockwise, +/- 12 dB is the available range.
Flatten button	Resets all the frequency bands to 0 dB.
Invert range	This will invert the current response curve.
Mode	The filter mode set here determines how the various frequency band controls interact to create the response curve. See also below.

About the filter modes

On the pop-up in the lower right corner there are several different EQ modes available. These modes can add color or character to the equalized output in various ways, which is sometimes desirable. As always, let your ears be the judge! Here follow brief descriptions of the filter modes:

- True Response – serial filters with accurate frequency response.
- Digi Standard – resonance of last band depends on sample rate.
- Variable Q – parallel filters where the resonance depends on the amount of gain. Musical sounding.
- Constant Q u – parallel filters where the resonance of the first and last bands depends on the sample rate (u=unsymmetric).
- Constant Q s – parallel filters where the resonance is raised when boosting the gain and vice versa (s=symmetric).
- Resonant – serial filters where a gain increase of one band will lower the gain in adjacent bands.

StudioEQ (Cubase only)



This is a high-quality 4-band parametric stereo equalizer with two fully parametric midrange bands. The low and high bands can act as either shelving filters (three types) or as a Peak (bandpass) or Cut (lowpass/highpass) filter.

Making settings

1. Click the corresponding On button to the left of the EQ curve display to activate any or all of the Low, Mid 1, Mid 2 or High equalizer bands.

When a band is activated, a corresponding eq point appears in the EQ curve display.

2. Set the parameters for an activated EQ band.

This can be done in several ways:

- By using the knobs.
- By clicking a value field and entering values numerically.
- By using the mouse to drag points in the EQ curve display window.

By using this method, you control both the Gain and Frequency parameters simultaneously. The knobs turn accordingly when you drag points.

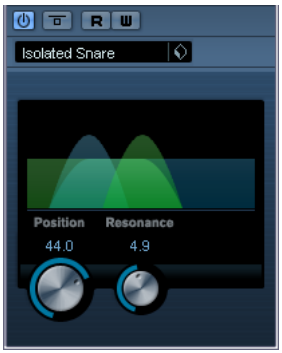
The following parameters are available:

Parameter	Description
Low Freq (20 to 2000 Hz)	This sets the frequency of the Low band.
Low Gain (-20 to +24 dB)	This sets the amount of cut/boost for the Low band.
Low Q-Factor	This controls the width or resonance of the Low band.
Low Filter mode	For the Low band, you can select between three types of shelving filters or Peak (bandpass) or Cut (lowpass/highpass) filters. The Gain parameter will be fixed if Cut mode is selected. -Shelf I adds resonance in the opposite gain direction slightly over the set frequency. -Shelf II adds resonance in the gain direction at the set frequency. -Shelf III is a combination of Shelf I and II.
Mid 1 Freq (20 to 20000 Hz)	This sets the center frequency of the Mid 1 band.
Mid 1 Gain (+/- 24 dB)	This sets the amount of cut/boost for the Mid 1 band.
Mid 1 Q-Factor (0.5 to 10)	This sets the width of the Mid 1 band. The higher this value, the “narrower” the bandwidth.
Mid 2 Freq (20 to 20000 Hz)	This sets the center frequency of the Mid 2 band.
Mid 2 Gain (-20 to +24 dB)	This sets the amount of cut/boost for the Mid 2 band.
Mid 2 Q-Factor (0.5 to 10)	This sets the width of the Mid 2 band. The higher this value, the “narrower” the bandwidth.
High Freq (200 to 20000 Hz)	This sets the frequency of the High band.
High Gain (-20 to +24 dB)	This sets the amount of cut/boost for the High band.
High Q-Factor	This parameter controls the width or resonance of the High band.
High Filter mode	For the High band, you can select between three types of shelving filters, and Peak or Cut filters. The Gain parameter will be fixed if Cut mode is selected. -Shelf I adds resonance in the opposite gain direction slightly below the set frequency. -Shelf II adds resonance in the gain direction at the set frequency. -Shelf III is a combination of Shelf I and II.
Output (-24 to +24 dB)	This parameter allows you to adjust the overall output level.
Auto Gain	When this is activated, the gain is automatically adjusted, keeping the output level constant regardless of the EQ settings.

Filter plug-ins

This section contains descriptions of the plug-ins in the “Filter” category.

DualFilter

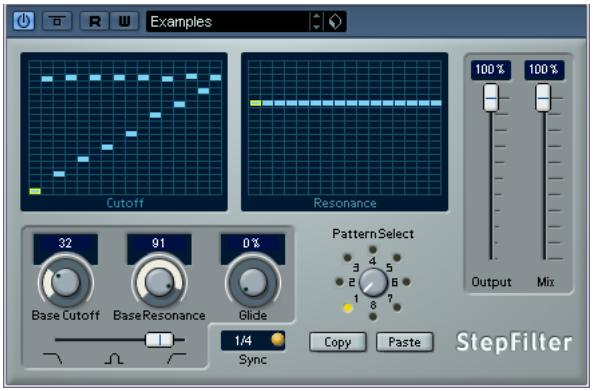


This effect filters out certain frequencies while allowing others to pass through.

The following parameters are available:

Parameter	Description
Position	This parameter sets the filter cutoff frequency. If you set this to a negative value, DualFilter will act as a low-pass filter. Positive values cause DualFilter to act as a high-pass filter.
Resonance	Sets the sound characteristic of the filter. With higher values, a ringing sound is heard.

StepFilter



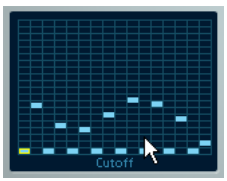
StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

General operation

StepFilter can produce two simultaneous 16-step patterns for the filter cutoff and resonance parameters, synchronized to the sequencer tempo.

Setting step values

- Setting step values is done by clicking in the pattern grid windows.
- Individual step entries can be freely dragged up or down the vertical axis, or directly set by clicking in an empty grid box. By click-dragging left or right, consecutive step entries will be set to the pointer position.



Setting filter cutoff values in the grid window.

- The horizontal axis shows the pattern steps 1–16 from left to right, and the vertical axis determines the (relative) filter cutoff frequency and resonance setting.

The higher up on the vertical axis a step value is entered, the higher the relative filter cutoff frequency or filter resonance setting.

- By starting playback and editing the patterns for the cutoff and resonance parameters, you can hear how your filter patterns affect the sound source connected to StepFilter directly.

Selecting new patterns

- Created patterns are saved with the project, and up to 8 different cutoff and resonance patterns can be saved internally.

Both the cutoff and resonance patterns are saved together in the 8 Pattern memories.

- To select new patterns you use the pattern selector.

New patterns are all set to the same step value by default.

Pattern Selector

Using pattern copy and paste to create variations

You can use the Copy and Paste buttons below the pattern selector to copy a pattern to another pattern memory location, which is useful for creating variations on a pattern.

- Select the pattern you wish to copy, click the Copy button, select another pattern memory location and click Paste.

The pattern is copied to the new location, and can now be edited to create variations using the original pattern as a starting point.

StepFilter parameters

Parameter/ Value	Description
Base Cutoff	This sets the base filter cutoff frequency. Cutoff values set in the Cutoff grid window are values relative to the Base Cutoff value.
Base Resonance	This sets the base filter resonance. Resonance values set in the Resonance grid window are values relative to the Base Resonance value. Note that very high Base Resonance settings can produce loud ringing effects at certain frequencies.
Glide	This will apply glide between the pattern step values, causing values to change more smoothly.
Filter Mode	This slider selects between lowpass (LP), bandpass (BP) or highpass (HP) filter modes (from left to right respectively).
Sync 1/1 to 1/32 (Straight, Triplet or Dotted)	This sets the pattern beat resolution, i.e. what note values the pattern will play in relation to the tempo.
Output	Sets the overall volume.
Mix	Adjusts the mix between dry and processed signal.

ToneBooster

ToneBooster is a filter that allows you to raise the gain in a selected frequency range. It is particularly useful when inserted before AmpSimulator in the plug-in chain (see “AmpSimulator” on page 9), greatly enhancing the tonal varieties available.

The following parameters are available:

Parameter	Description
Tone	This sets the center filter frequency.
Gain	Allows you to adjust the gain of the selected frequency range by up to 24 dB.
Width	This sets the resonance of the filter.
Mode	This sets the basic operational mode of the filter; Peak or Bandpass.

Tonic – Analog Modeling Filter (Cubase only)

Tonic is a versatile and powerful analog modeling filter plug-in based on the filter design of the Monologue monophonic synthesizer. Its variable characteristics plus the powerful modulation functions make it an excellent choice for all current music styles. Designed to be more a creative tool rather than a tool to fix audio problems, it can add color and punch to your tracks while being light on CPU usage.

The Tonic Analog Modeling Filter has the following properties:

- Dynamic multimode analog modeling filter (mono/stereo).
- 24dB low pass, 18dB low pass, 12dB low pass, 6dB low pass, 12dB band pass and 12dB high pass modes.
- Adjustable drive and resonance up to self-oscillation.
- Envelope follower for dynamic filter control with an audio signal.
- Audio and MIDI trigger modes.
- Powerful step LFO with smoothing and morphing.
- X/Y matrix pad for additional real-time modulation with access to all Tonic parameters.

Filter

Parameter	Description
Mode	Sets the filter type. Available filter types are: 24dB Low pass, 18dB Low pass, 12dB Low pass, 6dB Low pass, 12dB Band pass and 12dB High pass.
Cutoff	Sets the filter cutoff frequency. How this parameter operates is governed by the filter type.
Res	Changes the resonance of the multi-mode filter. Full resonance puts the filter into self-oscillation.
Drive	Drive adds a soft, tube-like saturation to the sound. Like for an analog filter, the amount of saturation also depends on the input signal level.
Mix	Sets the balance between dry and effect signal.
Ch.	Choose between mono or stereo operation. When set to mono, the output signal of Tonic will be mono regardless of the input signal.

Env Mod

Parameter	Description
Mode	Tonic offers three types of envelope modulation: “Follow” tracks the input signal’s volume envelope for dynamic control of the filter cutoff. “Trigger” uses the input signal to trigger the envelope and have it run through a single envelope cycle. “MIDI” uses any MIDI note to trigger the envelope. The filter cutoff tracks the keys played on the keyboard. In addition velocities higher than 80 will add an accent to the envelope by increasing the envelope depth and reducing the decay time. For MIDI control, set up a separate MIDI control track and select “Tonic” from the output pop-up menu for the track.
Attack	Controls the attack time of the envelope. Higher attack times result in slower rise times when the envelope is triggered.
Release	Controls the release time of the envelope. Higher release times result in slower envelope tails.
Depth	Controls the amount of envelope control applied to the filter cutoff level.
LFO Mod	Using this parameter, envelope level modulates the LFO speed. A rather stunning effect.

X/Y Pad

Parameter	Description
X Par	Sets the parameter to be modulated on the x axis of the XY Pad. All of Tonic's parameters are available as destinations
Y Par	Sets the parameter to be modulated on the y axis of the XY Pad.
XY Pad	Use the mouse to control any two of Tonic's parameters in combination. By moving the mouse horizontally, you can control the x parameter, by moving it vertically, you can control the y parameter. You can also record controller movements as automation data.

LFO Mod

Parameter	Description
Mode	Sets the direction of the step LFO modulation. The available modes are: Forward, Reverse, Alternating, and Random.
Depth	Controls the amount of LFO modulation applied to the filter cutoff level.
Rate	Controls the speed of the LFO modulation. The LFO rate is always in sync with the song tempo. For example: a rate of 4.00 steps per beat advances the step sequencer in 16th notes at a 4/4 time signature. A rate of 4.00 beats per step would advance the LFO at only one step per bar in a 4/4 time signature.
Smooth	The smooth parameter controls the smoothing of the LFO steps. This works like a glide effect applied to the filter cutoff.
Morph	Morph controls the playback value of the LFO step sequencer. It makes the LFO steps drift about randomly. Experiment freely with the morph parameter. As you return the knob to its zero position the step pattern will return to its original setting.
Steps	Sets the number of steps played in sequence. Deactivated steps are grayed out in the step window.
Preset	Offers a number of step LFO waveform patterns. Choices include: Sine, Sine+, Cosine, Triangle, Sawtooth, Square, Random and User (which is the pattern saved with the respective program).
Step Matrix	Click into the step matrix to set the level for each of the 16 LFO steps. A higher amount results in a deeper filter cutoff modulation. Click and drag along the matrix to "draw" a waveform.

WahWah

WahWah is a variable slope bandpass filter that can be auto-controlled by a side-chain signal or via MIDI modeling the well-known analog pedal effect (see below). You can independently specify the frequency, width and the gain for the Lo and Hi Pedal positions. The crossover point between the Lo and Hi Pedal positions is at 50.

The parameters are as follows:

Parameter	Description
Pedal	This controls the filter frequency sweep.
Freq Lo/Hi	Sets the frequency of the filter for the Lo and Hi Pedal positions.
Width Lo/Hi	Sets the width (resonance) of the filter for the Lo and Hi Pedal positions.
Gain Lo/Hi	Sets the gain of the filter for the Lo and Hi Pedal positions.
Slope	Specifies the slope of the filter; 6dB or 12dB.
Side-Chain On/Off	A signal routed to the Side-Chain input of the effect can control the Pedal parameter when this is activated. The louder the signal, the more the filter frequency (Pedal) is raised so the plug-in acts as an "auto-wha" effect. For a description of how to set up Side-Chain routing, see the chapter "Audio effects" in the Operation Manual.

MIDI control

For real-time MIDI control of the Pedal parameter, MIDI must be directed to the WahWah plug-in.

- Whenever the WahWah has been added as an insert effect (for an audio track or an FX channel), it will be available on the Output Routing pop-up menu for MIDI tracks. If WahWah is selected on the Output Routing menu, MIDI will be directed to the plug-in from the selected track.

Mastering – UV 22 HR

The UV22 HR is a dithering plug-in, based on an advanced algorithm developed by Apogee. For an introduction to the concept of dithering, see the chapter “Audio Effects” in the Operation Manual.

The following options can be set in the UV 22 HR control panel:

Option	Description
Hi	Try this first, it is the most “all-round” setting.
Low	This applies a lower level of dither noise.
Auto black	When this is activated, the dither noise is gated (muted) during silent passages in the material.
Bit Resolution	The UV22 HR supports dithering to multiple resolutions: 8, 16, 20 or 24 bits. You select the desired resolution by clicking the corresponding button.

⚠ Dither should always be applied post output bus fader.

Modulation plug-ins

This section contains descriptions of the plug-ins in the “Modulation” category.

AutoPan

This is a simple autopan effect. It can use different waveforms to modulate the left-right stereo position (pan), either using tempo sync or manual modulation speed settings.

The parameters are as follows:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the auto-pan speed can be set freely with the Rate knob, without sync to tempo.
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on (the button lights up) or off.
Width	Sets the depth of the Autpan effect.
Shape	Sets the modulation waveform. Sine and Triangle waveforms are available.
Side-Chain On/Off	A signal routed to the Side-Chain input of the effect can control the Width parameter when this is activated. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

Chorus

This is a single stage chorus effect. It works by doubling whatever is sent into it with a slightly detuned version. See also “StudioChorus” on page 30.

The parameters are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the chorus sweep (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Width	This determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Spatial	This sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
Mix	Sets the level balance between the dry signal and the effect. If Chorus is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Shape	This changes the shape of the modulating waveform, altering the character of the chorus sweep. Sine and triangle waveforms are available.
Filter Lo/Hi	These parameters allow you to roll off low and high frequencies of the effect signal, respectively.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold the modulation will be controlled by the side-chain signal’s envelope. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

Cloner (Cubase only)

The Cloner plug-in adds up to four detuned and delayed voices to the signal, for rich modulation and chorus effects.

The parameters are as follows:

Parameter	Description
Voices	This allows you to select the number of voices (up to four). For each added voice, a Detune and a Delay slider are added in the right half of the panel.
Spatial	This spreads the added voices across the stereo spectrum. Turn clockwise for a deeper stereo effect.
Mix	Sets the level balance between the dry signal and the effect. If Cloner is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Output	Allows you to reduce or increase the output gain by up to +/- 12dB.
Detune slider 1–4	This controls the relative detune amount for each voice. Positive and negative values can be set, from -100 to 100. A value of zero means no detune for that voice.
Delay slider 1–4	This controls the relative delay amount for each voice. A value of zero means no delay for that voice.
Detune	This parameter governs the overall depth of the detuning for all voices. If this is set to zero, no detuning takes place, regardless of the Detune slider settings. By clicking the natural button below this knob, you can change the pitch algorithm.
Humanize Detune knob	Humanize is turned on and off with the Static Detune button below this knob. When activated, the detune settings are subtly varied, for a richer effect. Values range from 0 to 100 (strongest detune variation). If deactivated, the set detune amount is static, and the knob is blacked out.
Delay	This parameter governs the overall depth of the delay for all voices. If this is set to zero, no delay takes place, regardless of the Delay slider settings.
Humanize Delay knob	Humanize is turned on and off with the Static Delay button button below this knob. When activated the delay settings are subtly varied, for a richer effect. Values range from 0 to 100 (strongest delay variation). If deactivated, the set delay amount is static, and the knob is blacked out.

Flanger

Flanger is a classic flanger effect with added stereo enhancement.

The parameters are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the flanger sweep (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Range Lo/Hi	This sets the frequency boundaries for the flanger sweep.
Feedback	This determines the character of the flanger effect. Higher settings produce a more “metallic” sounding sweep.
Spatial	This sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
Mix	Sets the level balance between the dry signal and the effect. If the Flanger is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Shape	This changes the shape of the modulating waveform, altering the character of the flanger sweep.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Manual	If this is activated, the flanger sweep will be static, i.e. no modulation. You can instead change the sweep position manually by turning this knob.
Filter Lo/Hi	These parameters allow you to roll off low and high frequencies of the effect signal, respectively.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold the modulation will be controlled by the side-chain signal’s envelope. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

Metalizer

The Metalizer feeds the audio signal through a variable frequency filter, with tempo sync or time modulation and feedback control.

Parameter	Description
Feedback	The higher the value, the more “metallic” the sound.
Sharpness	Governs the character of the filter effect. The higher the value, the narrower the affected frequency area, producing sharper sound and a more pronounced effect.
Tone	Governs the feedback frequency. The effect of this will be more noticeable with high Feedback settings.
On button	Turns filter modulation on and off. When turned off, the Metalizer will work as a static filter.
Mono button	When this is on, the output of the Metalizer will be in mono.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the modulation speed can be set freely with the Speed knob, without sync to tempo.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Output	Sets the overall volume.
Mix	Sets the level balance between the dry signal and the effect. If Metalizer is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.

Phaser

Phaser produces the well-known “swooshing” phasing effect with additional stereo enhancement.

The parameters are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the phaser sweep (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Width	The width of the modulation effect between higher and lower frequencies.
Feedback	This determines the character of the phaser effect. Higher settings produce a more pronounced effect.
Spatial	When using multi-channel audio, Spatial creates a 3-dimensional impression by delaying modulation in each channel.
Mix	Sets the level balance between the dry signal and the effect. If the Phaser is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Manual	If this is activated, the phaser sweep will be static, i.e. no modulation. You can instead change the sweep position manually by turning this knob.
Filter Lo/Hi	These parameters allow you to roll off low and high frequencies of the effect signal, respectively.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold the modulation will be controlled by the side-chain signal’s envelope. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

Ringmodulator

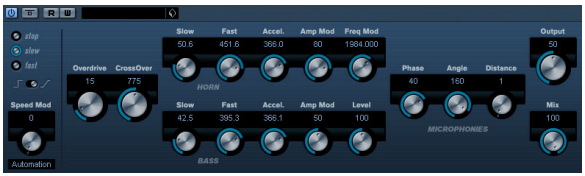
The Ringmodulator can produce complex, bell-like enharmonic sounds. Ring modulators work by multiplying two audio signals. The ring modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals.

The Ringmodulator has a built-in oscillator that is multiplied with the input signal to produce the effect.

Parameter	Description
Oscillator LFO Amount	Controls how much the oscillator frequency is affected by the LFO.
Oscillator Env. Amount	Controls how much the oscillator frequency is affected by the envelope (which is triggered by the input signal). Positive and negative values can be set, with center position representing no modulation. Left of center, a loud input signal will decrease the oscillator pitch, whereas right of center the oscillator pitch will increase when fed a loud input.
Oscillator Wave	Selects the oscillator waveform; square, sine, saw or triangle.
Oscillator Range	Determines the frequency range of the oscillator in Hz.
Oscillator Frequency	Sets the oscillator frequency +/- 2 octaves within the selected range.
Oscillator Roll-Off	Cuts high frequencies in the oscillator waveform, to soften the overall sound. This is best used when harmonically rich waveforms are selected (e.g. square or saw).
LFO Speed	Sets the LFO Speed.
LFO Env. Amount	Controls how much the input signal level – via the envelope generator – affects the LFO speed. Positive and negative values can be set, with center position representing no modulation. Left of center, a loud input signal will slow down the LFO, whereas right of center a loud input signal will speed it up.

Parameter	Description
LFO Waveform	Selects the LFO waveform; square, sine, saw or triangle.
Invert Stereo	This inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.
Envelope Generator (Attack and Decay dials)	The Envelope Generator section controls how the input signal is converted to envelope data, which can then be used to control oscillator pitch and LFO speed. It has two main controls: Attack sets how fast the envelope output level rises in response to a rising input signal. Decay controls how fast the envelope output level falls in response to a falling input signal.
Lock L<R	When this button is enabled, the L and R input signals are merged, and produce the same envelope output level for both oscillator channels. When disabled, each channel has its own envelope, which affects the two channels of the oscillator independently.
Output	Sets the overall volume.
Mix	Adjusts the mix between dry and processed signal.

Rotary



The Rotary plug-in simulates the classic effect of a rotary speaker. A rotary speaker cabinet features variable speed rotating speakers to produce a swirling chorus effect, commonly used with organs. Rotary features all the parameters associated with the real thing.

The parameters are as follows:

Parameter	Description
Speed (Stop/Slow/Fast)	This controls the speed of the Rotary in three steps.
Mode	Selects whether the Slow/Fast setting is a switch or a variable control. When switch mode is selected and Pitchbend is the controller, the speed will switch with an up or down flick of the bender. Other controllers switch at 64.
Speed Mod	Selects the Rotary speed from 0 (Stop) to 100 (Fast).
Overdrive	Applies a soft overdrive or distortion.
Crossover Freq.	Sets the crossover frequency (200–3000 Hz) between the low and high frequency loudspeakers.
Slow	Fine adjustment of the high rotor Slow speed.
Accel.	Fine adjustment of the high rotor acceleration time.
Fast	Fine adjustment of the high rotor Fast speed.
Amp Mod	High rotor amplitude modulation.
Freq Mod	High rotor frequency modulation.
Slow	Fine adjustment of the low rotor Slow speed.
Fast	Fine adjustment of the low rotor Fast speed.
Accel	Fine adjustment of the low rotor acceleration time.
Amp Mod.	Adjusts amplitude modulation depth.
Level	Adjusts overall bass level.
Phase	Adjusts the phasing amount in the sound of the high rotor.
Angle	Sets the simulated microphone angle. 0 = mono, 180 = one mic on each side.
Distance	Sets the simulated microphone distance from the speaker in inches.
Output	Adjusts the overall output level.
Mix	Adjusts the mix between dry and processed signals.

Directing MIDI to the Rotary

For real-time MIDI control of the Speed parameter, MIDI must be directed to the Rotary.

- Whenever the Rotary has been added as an insert effect (for an audio track or an FX channel), it will be available on the Output Routing pop-up menu for MIDI tracks. If Rotary is selected on the “out:” menu, MIDI will be directed to the plug-in from the selected track.

StudioChorus

The StudioChorus plug-in is a two stage chorus effect which adds short delays to the signal and pitch modulates the delayed signals to produce a “doubling” effect. The two separate stages of chorus modulation are completely independent and are processed serially (cascaded).

The parameters for each stage are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the chorus sweep (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Width	This determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Spatial	This sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
Mix	Sets the level balance between the dry signal and the effect. If StudioChorus is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Shape	This changes the shape of the modulating waveform, altering the character of the chorus sweep. Sine and triangle waveforms are available.
Filter Lo/Hi	These parameters allow you to roll off low and high frequencies of the effect signal, respectively.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold, the modulation will be controlled by the side-chain signal’s envelope. For a description of how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

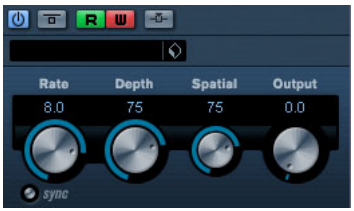
Tranceformer

Tranceformer is a ring modulator effect, in which the incoming audio is ring modulated by an internal, variable frequency oscillator, producing new harmonics. A second oscillator can be used to modulate the frequency of the first oscillator, in sync with the Song tempo if needed.

Parameter	Description
Waveform buttons	Sets the pitch modulation waveform.
Tone	Sets the frequency (pitch) of the modulating oscillator (1 to 5000 Hz).
Depth	Governs the depth of the pitch modulation.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the modulation speed can be set freely with the Speed knob, without sync to tempo.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
On button	Turns modulation of the pitch parameter on or off.
Mono button	Governs whether the output will be stereo or mono.
Output	Adjusts the output level of the effect.
Mix	Sets the level balance between the dry signal and the effect.

⇨ Note that clicking and dragging in the display allows you to adjust the Tone and Depth parameters at the same time!

Tremolo



Tremolo produces amplitude (volume) modulation.

Parameters are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the modulation speed can be set freely with the Rate knob, without sync to tempo.
Depth	This governs the depth of the amplitude modulation.
Spatial	This will add a stereo effect to the modulation.
Output	Adjusts the output volume.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold, the modulation will be controlled by the side-chain signal's envelope. For a description of how to set up Side-Chain routing, see the chapter "Audio effects" in the Operation Manual.

Vibrato

The Vibrato plug-in produces pitch modulation.

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet or dotted). If tempo sync is off, the modulation speed can be set freely with the Rate knob, without sync to tempo.
Depth	This governs the depth of the pitch modulation.
Spatial	This will add a stereo effect to the modulation.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshold, the modulation will be controlled by the side-chain signal's envelope. For a description of how to set up Side-Chain routing, see the chapter "Audio effects" in the Operation Manual.

Other plug-ins

This section contains descriptions of the plug-ins in the “Others” category.

Bitcrusher

If you’re into lo-fi sound, Bitcrusher is the effect for you. It offers the possibility of decimating and truncating the input audio signal by bit reduction, to get a noisy, distorted sound. You can for example make a 24-bit audio signal sound like an 8 or 4-bit signal, or even render it completely garbled and unrecognizable. The parameters are:

Parameter	Description
Mode	Select one of four operating modes for the Bitcrusher. Each mode will produce a result sounding a bit different. Modes I and III are nastier and noisier, while modes II and IV are more subtle.
Sample Divider	This sets the amount by which the audio samples are decimated. At the highest setting (65), nearly all of the information describing the original audio signal will be eliminated, turning the signal into unrecognizable noise.
Depth	Use this to set the desired bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 will create mostly noise.
Output	Governs the output level from the Bitcrusher. Drag the slider upwards to increase the level.
Mix	This slider regulates the balance between the output from the Bitcrusher and the original audio signal. Drag the slider upwards for a more dominant effect, and drag it downwards if you want the original signal to be more prominent.

Chopper

Chopper is a combined tremolo and autopan effect. It can use different waveforms to modulate the level (tremolo) or left-right stereo position (pan), either using tempo sync or manual modulation speed settings. The parameters are as follows:

Parameter	Description
Waveform buttons	Sets the modulation waveform.
Depth	Sets the depth of the Chopper effect. This can also be set by clicking in the graphic display.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the tremolo/auto-pan speed can be set freely with the Speed knob, without sync to tempo.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on (the button lights up) or off.
Stereo/Mono button	Determines whether the Chopper will work as an autopanner (button set to “Stereo”) or a tremolo effect (button set to “Mono”).
Mix	Sets the level balance between the dry signal and the effect. If Chopper is used as a send effect, this should be set to maximum.

Octaver

This plug-in can generate two additional voices that track the pitch of the input signal one octave and two octaves below the original pitch, respectively. Octaver is best used with monophonic signals. The parameters are as follows:

Parameter	Description
Direct	This adjusts the mix of the original signal and the generated voice(s). A value of 0 means only the generated and transposed signal is heard. By raising this value, more of the original signal is heard.
Octave 1	This adjusts the level of the generated signal one octave below the original pitch. Set to 0 means the voice is muted.
Octave 2	This adjusts the level of the generated signal two octaves below the original pitch. Set to 0 means the voice is muted.

Tuner

This is a guitar tuner. Simply connect a guitar or other instrument to an audio input and select the Tuner as an insert effect (make sure you deactivate any other effect that alters pitch, like chorus or vibrato). When the instrument is connected, proceed as follows:

- Play a note.

The key is shown in the middle of the display. In addition, the frequency in Hz is shown in the bottom left corner and the octave range in the bottom right corner. If the key is wrong (e.g. if you wish to tune the E string and the key is shown as Fb), first tune the string so that the correct key is shown.

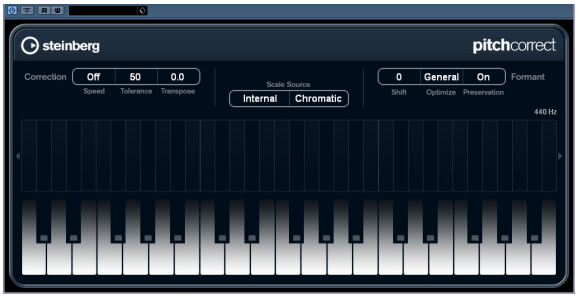
- The two arrows indicate any deviation in pitch by their position. If the pitch is flat, they will be positioned in the left half of the display, if the pitch is sharp they will be in the right half.

The deviation is also shown (in Cent) in the upper area of the display.

- Tune the instrument so that the two arrows are in the middle.

Repeat this procedure for each string.

PitchShift – Pitch Correct



Pitch Correct automatically detects, adjusts and fixes slight pitch and intonation inconsistencies in monophonic vocal and instrumental performances in real time. The advanced algorithms of this plug-in preserve the formants of the original sound thus allowing for natural sounding pitch correction without the typical “Micky Mouse” effect.

Furthermore, you can use Pitch Correct creatively. You could e.g. create backing vocals by modifying the lead vocals or vocoder sounds by using extreme values. You can use an external MIDI controller, a MIDI or Instrument track or the virtual keyboard to “play” a note or a scale of target pitches that determine the current scale notes to which the audio should be shifted. This allows you to change your audio in a very quick and easy way, which is extremely useful for live performances. In the keyboard display, the original audio will be displayed in blue while the changes are displayed in orange.

The parameters are:

Parameter	Description
Correction – Speed	This parameter determines the sensitivity of analysis.
Correction – Tolerance	With this parameter, you can determine the smoothness of the pitch change. Higher values cause the pitch shift to occur immediately. 100 is a very drastic setting that is designed mainly for special effects (e.g. the famous “Cher” effect).
Correction – Transpose (-12 to 12)	With this parameter you can adjust (or “retune”) the pitch of the incoming audio in semitone steps. You can set positive and negative values from -12 to 12. A value of zero means the signal is not Transposed.
Scale Source – Internal	If you choose the Internal option from the Scale Source pop-up, another pop-up menu will be shown to the right, where you can choose the key to which the source audio will be adapted. The available options are: Chromatic: The audio will be pitched to the closest semitone. Major/minor: The audio will be pitched to the major/minor scale specified in the pop-up menu to the right. This will be reflected on the keyboard display. Custom: The audio will be pitched to the notes that you specify by clicking the desired keys on keyboard display. If you want to reset the keyboard, click on the orange line below the display.
Scale Source – External MIDI Scale	Here you can specify how the audio should be changed by your external MIDI controller, the virtual keyboard or your MIDI or Instrument track. The audio will be shifted to a scale of target pitches. Note that you have to assign the audio track as output for your MIDI or Instrument track and that the Correct parameter has to be turned up.
Scale Source – External MIDI Note	Here you can specify how the audio should be changed by your external MIDI controller, the virtual keyboard or your MIDI or Instrument track. The audio will be shifted to a scale of target pitches. Note that you have to assign the audio track as output for your MIDI or Instrument track and that the Tolerance parameter has to be turned up.
Formant – Shift (-60 to 60)	With this parameter you can change the natural timbre, i.e. the characteristic frequency components of the source audio.
Formant – Optimize (General, Male, Female)	The Type button lets you specify if the sound source. While General is the default setting, Male is designed for low pitches and Female for high pitches.
Formant – Preservation (On/Off)	If you deactivate this button, formants are raised and lowered with the pitch, provoking strange vocal effects. Higher values result in “Micky Mouse” effects, lower values in “Monster” sounds. If you activate this button, the formants will be kept, maintaining the character of the audio.
Master Tuning	Detunes the output signal. The default setting is 44,0 kHz.

Restoration plug-ins – Grungelizer

The Grungelizer adds noise and static to your recordings – kind of like listening to a radio with bad reception, or a worn and scratched vinyl record. The available parameters are as follows:

Parameter	Description
Crackle	This adds crackle to create that old vinyl record sound. The farther to the right you turn the dial, the more crackle is added.
RPM switch	When emulating the sound of a vinyl record, this switch lets you set the RPM (revolutions per minute) speed of the record (33/45/78 RPM).
Noise	This dial regulates the amount of static noise added.
Distort	Use this dial to add distortion.
EQ	Turn this dial to the right to cut off the low frequencies, and create a more hollow, lo-fi sound.
AC	This emulates a constant, low hum of AC current.
Frequency switch	This sets the frequency of the AC current (50 or 60 Hz), and thus the pitch of the AC hum.
Timeline	This dial regulates the amount of overall effect. The farther to the right (1900) you turn this dial, the more noticeable the effect.

Reverb plug-ins

This section contains descriptions of the plug-ins in the “Reverb” category.

REVerence (Cubase only)

REVerence is a convolution tool that allows you to apply room characteristics (reverb) to the audio. This is done by processing the audio signal according to an impulse response – a recording of an impulse in a room or another location that is used as a fingerprint of the room. As a result, the processed audio will sound as if it were played in the same location. Included with the plug-in are top quality samples of real spaces to create reverberation. REVerence also contains an integrated equalizer with a high shelving filter, a parametric filter and a low shelving filter. You can also reverse and trim impulse responses and use the integrated program matrix to automate smooth parameter changes.

⇨ REVerence is a plug-in that can be very demanding in terms of RAM. This is because the impulse responses that you load into the REVerence program slots are loaded into RAM in order to guarantee an artefact-free switching between the different programs.

Loading programs

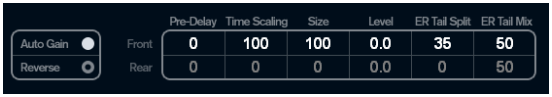
A program is the combination of an impulse response and its settings. These include reverb settings (see “Changing the reverb settings” on page 36), EQ settings (see “Making EQ settings” on page 37), pictures (see “Loading pictures” on page 38) and output settings (see “Making Output settings” on page 38). The program matrix allows you to load programs and to view the name of the current program, i.e. the impulse response (see “Program handling” on page 39).

The available parameters are as follows:

Parameter	Description
Program name	In the upper left corner of the plug-in panel, either the name of the loaded impulse response file, or the name of the program will be shown. After loading the impulse response, its number of channels and the length in seconds will be displayed for a few seconds. If you click on an other program slot, the name of this program will be displayed below the name of the loaded program. If you click the import button, you can import your own impulse response file. In this case, the file name will be shown.
browse	This opens a browser window showing the included programs. A program consists of an impulse response file, its settings and its picture. Click the Categories button in the browser to open the Filter section where you can sort the impulse responses, e.g. by Room Type, Channels and Family Name.
import	Clicking this button allows you to load your own impulse response file from disk. These should be ordinary wave or aiff audio files, with a maximum duration of 10 seconds. Longer files will be cut automatically.
Program slots (1 to 36)	The active program slot is indicated by a white frame. An armed program slot is indicated by a blinking white frame. A program slot with stored settings is indicated by a blue background. For more details, see “Program handling” on page 39.
Impulse Transition Blending Active	When changing impulse responses, a crossfade between the old and the new program is performed. The Impulse Transition Blending indicator shows the fade time between the two programs.
store	Stores the active impulse response and its settings as a program, see “Program handling” on page 39.
recall	Recalls the stored program from the program slot, see “Program handling” on page 39.
erase	Deletes the stored program from the matrix, see “Program handling” on page 39.

Changing the reverb settings

The reverb settings allow you to change the characteristics of the room. Here, you can revert the impulse file and change its volume, length or size.

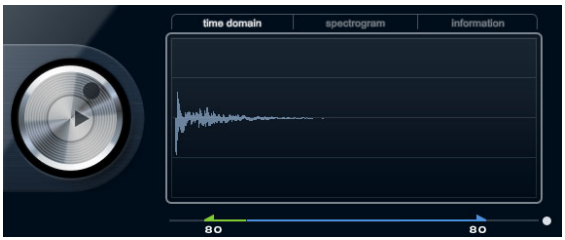


The available parameters are as follows:

Parameter	Description
AutoGain	When this is activated, the impulse response is automatically normalized.
Reverse	When this is activated, the impulse response is reversed.
Pre-Delay	The amount of time between the dry signal and the onset of the reverb. This allows you to simulate larger spaces by choosing higher pre-delay values.
Time Scaling	This parameter controls the reverb time.
Size	This determines the size of the simulated room.
Level	A level control for the impulse response. Usually, this governs the volume of the reverb.
ER Tail Split	This sets a split point between the early reflections and the tail allowing you to determine where the reverb tail begins. A value of 60 means that the early reflections will be heard for 60 ms.
ER Tail Mix	Allows you to set up the relation of early reflections and tail. Values above 50 attenuate the early reflections and values below 50 will attenuate the tail.
Rear	If you are working with surround tracks up to 5.1, you can set up an offset for the rear channels (in relation to the upper parameter).

Displaying impulse responses

The impulse display section allows you to view the impulse response details and to change the length of the response (trimming).

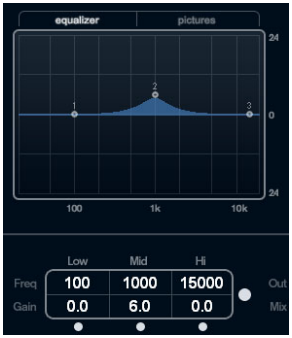


The available parameters are as follows:

Parameter	Description
Play button/ Time scaling wheel	When clicking the play button to apply the loaded impulse response, a short click is played. This provides a neutral test sound that makes it easier for you to know how different settings influence the reverb characteristics. The Time scaling wheel lets you adjust the Reverb time.
time domain	The Time Domain display shows the waveform of the impulse response.
spectrogram	The Spectrogram display shows the analyzed spectrum of the impulse response. The time is displayed along the horizontal axis, frequency along the vertical axis and the volume is represented by the color.
information	The Information display shows additional information, e.g. the name of the program and the loaded impulse response, the number of channels, the length and the Broadcast Wave File information.
Activate Impulse trimming (button)	When Impulse trimming is active, a slider is shown below the Impulse display that allows you to set a trim value for the start and the end of the impulse response from 1 to 100.
Trim slider	When Impulse trimming is active you can trim your impulse response. Drag the front handle to trim the start of the direct part of the impulse response or the end handle to trim the reverb tail. You can also use the mouse wheel for trimming. Note that the impulse response will be cut without any fading.

Making EQ settings

In the equalizer/pictures section you can make equalizer settings to tune the sound of the reverb.

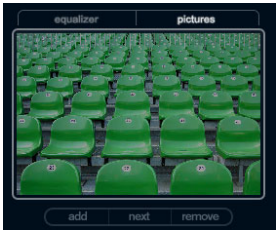


The available parameters are as follows:

Parameter	Description
EQ curve display	Shows the EQ curve. You can make settings by using the EQ parameters below the display or by changing the curve manually by dragging the curve points.
Activate EQ (button)	This activates the EQ for the effect plug-in.
Low Shelf On (button)	Activates the low shelf filter that boosts or cuts frequencies below the cutoff frequency by the specified amount.
Low Freq (20 to 500)	This sets the frequency of the Low band.
Low Gain (-24 to +24)	This sets the amount of cut/boost for the Low band.
Mid Peak On (button)	Activates the mid peak filter that creates a peak or notch in the frequency response.
Mid Freq (100 to 10000)	This sets the center frequency of the Mid band.
Mid Gain (-12 to +12)	This sets the amount of cut/boost for the Mid band.
Hi Shelf On (button)	This activates the high shelf filter that boosts or cuts frequencies above the cutoff frequency by the specified amount.
Hi Freq (5000 to 20000)	This sets the frequency of the Hi band.
Hi Gain (-24 to +24)	This sets the amount of cut/boost for the Hi band.

Loading pictures

In the equalizer/pictures section you can load or change pictures to illustrate the setting, i.e. the recording location or microphone arrangement of the loaded impulse response. If you import your own impulse response by clicking on the import button in the program matrix, all folders that are located on the same level will automatically be scanned for images. Up to five images can be loaded. Note that the images will only be referenced and not copied to the project folder.



The available parameters are as follows:

Parameter	Description
add	If you click this button, a file dialog opens where you can navigate to the picture file to import. Supported file formats are jpg, gif and png.
next	If several images are loaded, you can click on this button display the next image.
remove	If you click this button, the active picture will be deleted. Note that this will not remove the picture file from your hard disk (you must do this in the Explorer/Finder).

Making Output settings

In the Output section you can control the overall level and determine the dry/wet mix.

The available parameters are as follows:

Parameter	Description
Output activity indicator	Displays the output level, giving you an indication of the overall level of the impulse response and its settings.
Output slider	This slider allows you to adjust the overall output level.
Out (-24 to +12)	Raises or lowers the signal output of the plug-in.
Mix (0 to 100)	Sets the level balance between the dry and the wet signal.

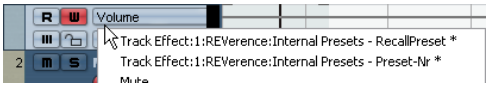
Program handling

You can save your REVerence settings as VST presets. These presets will contain all loaded impulse responses along with all parameter settings for the REVerence plug-in.

You can set up and use the REVerence program matrix which allows for quick and easy recall of your impulse responses and settings. This has the following advantages:

- The impulse responses are preloaded into RAM allowing for shorter loading times.
- When automating your project you can save programs with different impulse response settings and different scenes and load them, again saving load time.

If you work with programs you do not have to load a complete VST preset which would lead to the writing of unnecessary automation data compared to the two automation events that are written when you load a REVerence program.



Automation parameters written when changing REVerence programs

Proceed as follows to set up REVerence programs:

1. Activate a program slot by clicking on it.

A blinking white frame will be shown to indicate that this program slot is armed and that your next steps will take effect on this slot.

2. Click on the browse button, or click on the empty slot again to load one of the included impulse responses, or on the import button if you want to open your own impulse response file. In the browser that appears, select the file that you would like to use as an impulse response and click OK.

The name of the loaded impulse response will be shown in the upper left corner of the REVerence panel.

3. Set up the REVerence parameters and click the store button to save the impulse response and the current settings as a program.

The program slot will show a blue background to indicate that an impulse response is loaded for this slot.

4. Set up as many programs (up to 36) as you need by following the steps above.

5. Recall a saved program by double-clicking on the program slot.

A white frame will be shown to indicate that this program is active.

⇨ Note that when switching programs, the new program, i.e. impulse response, will not be loaded immediately. Instead the plug-in create a crossfade between the old and the new program. The fade time depends on the length of the impulse responses and your RAM and is displayed by the progress bar to the left of the program matrix.

The Impulse Transition Blending indicator

- By clicking the recall button to the left of the program matrix you can recall the saved settings of a program. This is useful, if you changed the parameters and are not satisfied with the result.
- By clicking the erase button to the left of the program matrix you can erase a program.

6. If you would like to use your programs for another project, save your settings as a VST preset by opening the corresponding pop-up and proceeding as usual.

RoomWorks

RoomWorks is a highly adjustable reverb plug-in for creating realistic room ambience and reverb effects in stereo and surround formats. The CPU usage is adjustable to fit the needs of any system. From short room reflections to cavern-sized reverb, this plug-in delivers high quality reverberation. RoomWorks has the following parameters:

Parameter	Description
Low Freq	Frequency at which the low shelving filter takes effect.
High Freq	Frequency at which the high shelving filter takes effect. Both the high and low filters EQ the input signal prior to reverb processing.
Low Gain	The amount of boost or cut for the low shelving filter.
High Gain	The amount of boost or cut for the high shelving filter.
Pre-Delay	The amount of time before the onset of reverb. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.
Reverb Time	Reverb Time in milliseconds.
Size	This alters the delays times of early reflections to simulate larger or smaller spaces.
Diffusion	This affects the character of the reverb tail. Higher diffusion is smoother while less diffusion can be clearer. This emulates changing the types of surfaces in a room (brick vs. carpet for instance).
Width	This controls the width of the stereo image. 100% gives you full stereo reverb. At 0%, the reverb is all in mono.
Variation	Pressing this button will generate a new version of the same reverb program using altered reflection patterns. This is helpful when certain sounds are causing odd ringing or undesirable results. Creating a new variation will often solve these issues. There are 1000 possible variations.
Hold	Pressing this button freezes the reverb buffer in an infinite loop (yellow circle around button). You can create some interesting pad sounds using this feature.
Low Range	This determines the frequency below which low damping will occur.
High Range	This determines the frequency above which high frequency damping will occur.
Low Level	This affects the decay time of low frequencies. Normal room reverb decays quicker in the high and low frequency range than in the midrange. Lowering the level percentage will cause low frequencies to decay quicker. Values above 100% will cause low frequencies to decay longer than the midrange.
High Level	This affects the decay time of high frequencies. Normal room reverb decays quicker in the high and low frequency range than in the midrange. Lowering the level percentage will cause high frequencies to decay quicker. Values above 100% will cause high frequencies to decay longer than the midrange.
Amount	This determines how much effect the envelope attack and release controls have on the reverb itself. Lower numbers have a more subtle effect while higher numbers sound more drastic.
Attack	The envelope settings in RoomWorks control how the reverb will follow the dynamics of the input signal in a fashion similar to a noise gate or downward expander. Attack determines how long in milliseconds it takes for the reverb to reach full volume after a signal peak. This is similar to a predelay but the reverb is ramping up instead of starting all at once.
Release	The release determines how long after a signal peak the reverb can be heard before being cut off, similar to a gate's release time.
Mix	Determines the blend of dry (unprocessed) signal to wet (processed) signal. When using RoomWorks inserted in an FX channel, you will most likely want to set this to 100% or use the Send button.
Wet only	This button defeats the mix parameter, setting the effect to 100% wet or affected signal. This button should normally be pressed when RoomWorks is being used as a send effect inserted on an FX or group channel.
Distance	This control is only available for surround configurations. With this parameter you can control where the virtual listening position is within the room. Positive values position the listener closer to the front of the room and negative values place the listener towards the rear of the room.
Rotate	This button is only available for surround configurations. When active, the perspective of the room is shifted 90°.
Balance	This control is only available for surround configurations. Balance controls the relative levels between the forward and rear speakers. Positive values favor the front speakers and negative values favor the rear speakers. Note that when the Rotate option is activated, these relationships will shift 90°.
Efficiency	This unique control determines how much of the CPU is used for RoomWorks. The lower the percentage of efficiency, the more CPU resources will be used. This will yield a higher quality reverb than higher percentage settings. Interesting effects can be created with very high Efficiency settings (>90%). Experiment for yourself.

Parameter	Description
Export	This button determines if during audio export RoomWorks will use the maximum CPU power for the highest quality reverb or not. You may wish to keep a higher efficiency setting for a desired effect during export. If you want the highest quality reverb during export make sure this is selected (yellow circle around button).

⇨ Cubase only: Note that the options in the Surround section on the far right of the RoomWorks panel are available only when using the plug-in as an insert for a surround-enabled track.

RoomWorks SE

RoomWorks SE is a “lite” version of the RoomWorks plug-in. This plug-in delivers high quality reverberation, but has fewer parameters and is less CPU demanding than the full version. RoomWorks SE has the following parameters:

Parameter	Description
Pre-Delay	The amount of time before the onset of reverb. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.
Reverb Time	Reverb Time in seconds.
Diffusion	This affects the character of the reverb tail. Higher diffusion is smoother while less diffusion can be clearer. This emulates changing the types of surfaces in a room (brick vs. carpet for instance).
High Level Amount	This affects the decay time of high frequencies. Normal room reverb decays quicker in the high and low frequency range than in the midrange. Lowering the level percentage will cause high frequencies to decay quicker. Values above 100% will cause high frequencies to decay longer than the midrange.
Low Level Amount	This affects the decay time of low frequencies. Normal room reverb decays quicker in the high and low frequency range than in the midrange. Lowering the level percentage will cause low frequencies to decay quicker. Values above 100% will cause low frequencies to decay longer than the midrange.
Mix	Determines the blend of dry (unprocessed) signal to wet (processed) signal. When using RoomWorks SE inserted in an FX channel, you will most likely want to set this to 100% or use the Send button.

Spatial plug-ins

This section contains descriptions of the plug-ins in the “Spatial” category.

MonoToStereo

This effect will turn a mono signal into a “pseudo-stereo” signal. The plug-in must be inserted on a stereo track playing a mono file to work.

The parameters are as follows:

Parameter	Description
Width	This controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.
Delay	This parameter increases the amount of differences between the left and right channels to further increase the stereo effect.
Color	This parameter also generates differences between the channels to increase the stereo effect.
Mono	This switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when creating an artificial stereo image.

StereoEnhancer

This plug-in will expand the stereo width of (stereo) audio material. It cannot be used with mono files.

The parameters are as follows:

Parameter	Description
Width	This controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.
Delay	This parameter increases the amount of differences between the left and right channels to further increase the stereo effect.
Color	This parameter also generates differences between the channels to increase the stereo enhancement.
Mono	This switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when enhancing the stereo image.

Surround plug-ins (Cubase only)

This section describes the plug-ins in the “Surround” category.

Mix6To2 (Cubase only)

The Mix6To2 effect allows you to control the levels of up to six surround channels, and to mix these down to a stereo output. The pop-up menu contains a number of speaker arrangement presets that correspond to some default surround formats. The Mix6To2 lets you quickly mix down your surround mix format to stereo, and to include parts of the surround channels in the resulting mix.

- Note that Mix6To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. Also note that the Mix6To 2 should be placed in one of the post fader insert effect slots for the output bus.

Each of the surround channels has the following parameters:

- Two volume faders that govern the levels of the surround bus to the left and right side of the (master) bus.
- A Link button that links the two volume faders.
- Two Invert buttons allow you to invert the phase of the left and right side of the surround bus.

The Master bus has the following parameters:

- A Link button that links the two Master faders.
- A Normalize button. If activated, the mixed output will be normalized, i.e. the output level will automatically be adjusted so that the loudest signal is as loud as possible without clipping.

SurroundPan (Cubase only)

For a description of the SurroundPan plug-in, see the chapter “Surround Sound (Cubase only)” in the Operation Manual.

Tools plug-ins

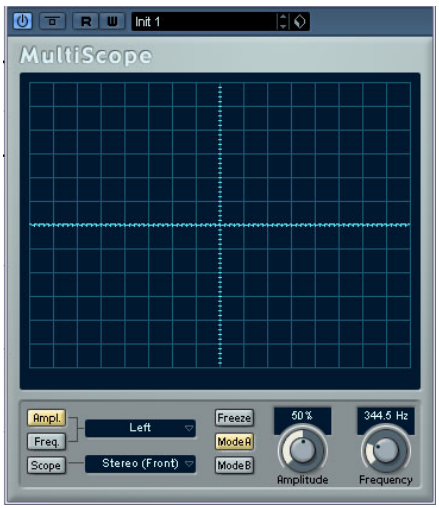
This section describes the plug-ins in the “Tools” category.

MultiScope (Cubase only)

The MultiScope can be used for viewing the waveform, phase linearity or frequency content of a signal. There are three different modes:

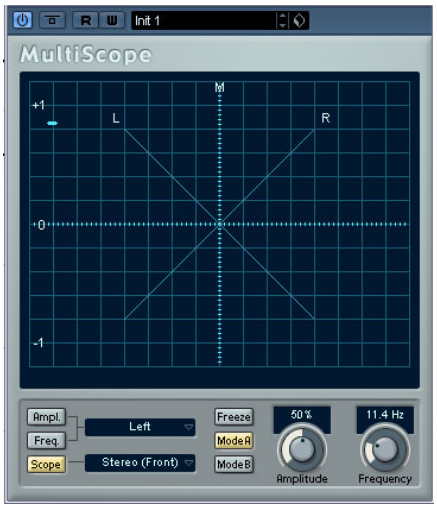
- Oscilloscope (Ampl.)
- Phase Correlator (Scope)
- Frequency Spectrum analyzer (Freq.)

Ampl (Oscilloscope) mode



- To view a signal waveform, open the MultiScope control panel and make sure that the button “Ampl.” in the lower left corner is lit.
- If the source signal is stereo you can now select either the Left or Right channel for viewing, or Stereo for both channels to be shown in the window. If it is a Mono signal, this won’t matter.
- If the MultiScope is used with a multi-channel track or output bus, you can select any speaker channel for viewing, or All Channels to view them all at once.
- You can now adjust the Amplitude knob to increase/decrease the vertical size of the waveform, and the frequency knob to select the frequency area for viewing.
- The “Freeze” button can be used to freeze the display for all three Scope modes.
  Click it again to exit freeze mode.

Phase Correlator mode



To select the phase correlator, click the “Scope” button so that it lights up. The phase correlator indicates the phase and amplitude relationship between channels in a stereo pair or a surround configuration.

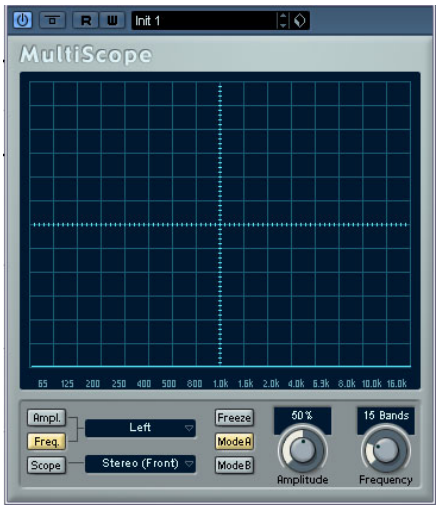
For stereo pairs, the indications work in the following way:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but fairly round shape indicates a well balanced stereo signal. If the shape “leans” to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phase Meter will show a straight line, angled 90° to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90° on the other.
- Generally, the more you can see a “thread”, the more bass in the signal, and the more “spray-like” the display, the more high frequencies in the signal.

When the MultiScope is used with a surround channel in Scope mode, the pop-up menu to the right of the Scope button determines the result:

- If “Stereo (Front)” is selected, the display will indicate the phase and amplitude relationship between the front stereo channels.
- If “Surround” is selected, the display indicates the energy distribution in the surround field.

Frequency Spectrum Analyzer



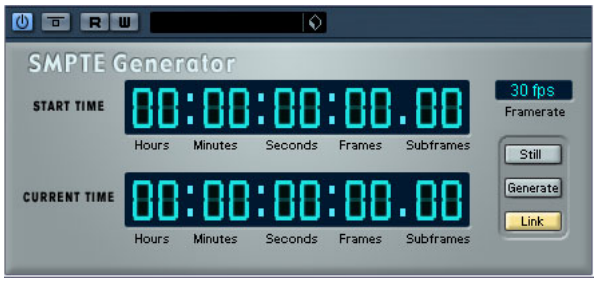
- Click on the “Freq” button so that it lights up in yellow.

The MultiScope is now in Frequency Spectrum analyze mode, and will divide the frequency spectrum into separate vertical bands, which allows you to get a visual overview of the different frequencies’ relative amplitude. The frequency bands are shown left to right, starting with the lower frequencies.

- If the source signal is stereo you can now select either the Left or Right channel for viewing, or Stereo for both channels to be shown in the window. If it is a Mono signal, this won’t matter.
- If the MultiScope is used with a multi-channel track or output bus, you can select any speaker channel for viewing, or All Channels to view them all at once.
- Adjust the Amplitude knob to increase/decrease the vertical range of the bands.
- By adjusting the Frequency knob, you can divide the frequency spectrum into 8, 15, or 31 bands, or you can select “Spectrum”, which shows a high resolution view.
- Use the Mode A and Mode B buttons to switch between different view modes.

Mode A is more graphically detailed, showing a solid, blue amplitude bar for each band. Mode B is less detailed, showing a continuous blue line that displays the peak levels for each band. These view modes don’t have any effect if you have selected “Spectrum” with the Frequency knob.

SMPTEGenerator (Cubase only)



This plug-in is not an effect device. It sends out SMPTE time code to an audio output, allowing you to synchronize other equipment to Cubase (provided that the equipment can sync directly to SMPTE time code). This can be very useful if you don’t have access to a MIDI-to-time code converter.

The following items and parameters are available:

- Still Button

Activate this to make the device generate SMPTE time code at the current cursor position in stop mode.

- Generate Button

Activate this to make the device generate SMPTE time code.

- Link Button

This synchronizes the time code output to the Transport time positions. When Link is activated, the time code output will exactly match the play position in Cubase.
Activating the Generate button makes the device send the SMPTE time code in “free run” mode, meaning that it will output continuous time code, independently from the transport status in Cubase. If you wish to “stripe” a tape with SMPTE, you should use this mode.

- Start Time

This sets the time at which the SMPTE Generator starts, when activated in “free run” mode (Link button off). To change the Start time, click on a digit and move the mouse up or down.

- Current Time

When Link is on this shows the current position in Cubase. If Link is off it shows the current time of the SMPTE Generator in “free run” mode. This cannot be set manually.

- Framerate

This defaults to the frame rate set in the Project Setup dialog. If you wish to generate time code in another frame rate than the Project is currently set to (for example to stripe a tape), you can select another format on the Framerate pop-up (provided that “Link” is off).
Note, however, that for the other device to synchronize correctly with Cubase, the framerate has to be the same in the Project Setup dialog, the SMPTE Generator and in the receiving device.

Example – Synchronizing a device to Cubase

Proceed as follows:

1. Connect the SMPTE Generator as an insert effect on an audio channel, and route the output of that channel to a separate output.
Make sure that no other insert or send effects are used on the time code channel. You should also disable EQ, if this is active.

2. Connect the corresponding output on the audio hardware to the time code input on the device you wish to synchronize to Cubase.
Make all necessary settings in the other device, so that it is set to synchronize to incoming timecode.

3. Adjust the level of the time code if needed, either in Cubase or in the receiving device.
Activate Generate button (make the device send the SMPTE time code in “free run” mode) to test the level.

4. Make sure that the frame rate in the receiving device matches the frame rate set in the SMPTE Generator.

5. Activate the Link button.
The SMPTE Generator will now output time code that matches the position of the Cubase Transport panel.

- Press Play on the Cubase Transport panel.

The other device is now synchronized and will follow any position changes set with the Cubase transport controls.

Drag offset for display

If you want to enter an offset, click with the mouse into the display and drag upwards or downwards to change the values. This enters a display offset – the current cursor position will not be affected. In Generate mode this offsets the Start Time, in Link mode it offsets the generated Timecode.

TestGenerator (Cubase only)

This utility allows you to generate an audio signal, which can be recorded as an audio file. The resulting file can then be used for a number of purposes:

- For testing the specifications of audio equipment.
- For measurements of various kinds, including calibrating tape recorders.
- For testing signal processing methods.
- For educational purposes.

The TestGenerator is based on a waveform generator which can generate a number of basic waveforms such as sine and saw and various types of noise. In addition, you can also set the frequency and amplitude of the generated signal.

As soon as you add the TestGenerator as an effect to an audio track and activate it, a signal is generated. You can then activate recording as usual to record an audio file according to the signal specifications:

Parameter	Description
Waveforms	By clicking these buttons, you select the basis for the signal generated by the waveform generator. You can select between four basic waveforms: Sine, Square, Sawtooth and Triangle, or three types of noise (white, brown and pink noise – from left to right).
Frequency	This controls the frequency of the generated signal, from 1 Hz to 20000 Hz.
Gain	This controls the amplitude of the signal. The higher the value (up to 0 dB) the stronger the signal.

2

The included VST Instruments

Introduction

This chapter contains descriptions of the included VST instruments and their parameters.

⇨ Most of the included instruments are compatible with VST3, this is indicated by an icon in front of the name (for further information, see the section “About VST 3” in the chapter “Audio Effects” in the Operation Manual).

Prologue

Prologue is modelled on subtractive synthesis, the method used in classic analog synthesizers. It has the following basic features:

- Multimode filter

Variable slope lowpass and hipass, plus bandpass and notch filter modes – see “About the filter types” on page 51.

- Three oscillators, each with 4 standard waveforms plus an assortment of specialized waveforms.

See “Selecting Waveforms” on page 47.

- Frequency modulation.

See “About frequency modulation” on page 49.

- Ring Modulation.

See “Ring modulation” on page 50.

- Built-in effects.

See “Effects (EFX) page” on page 55.

- Prologue receives MIDI in Omni mode (on all MIDI channels).

You don’t need to select a MIDI channel to direct MIDI to the Prologue.

⇨ The signal flow of the Prologue synth is illustrated in the section “Diagrams” on page 86.

Sound parameters

Oscillator section

This section contains parameters affecting the 3 oscillators. These are located in upper half of the instrument panel.

Selecting Waveforms

Each oscillator has a number of waveforms which are selectable by clicking on the waveform name in the box located in each oscillator section.

Selecting waveforms.

The following waveforms are available:

Waveform	Description
Sawtooth	This waveform contains all harmonics and produces a bright and rich sound.
Parabolic	This could be described as a “rounded” sawtooth waveform, producing a softer timbre.
Square	Square waveforms only contain odd number harmonics, which produces a distinct, hollow sound.

Waveform	Description
Triangle	The Triangle waveform generates only a few harmonics, spaced at odd harmonic numbers, which produces a slightly hollow sound.
Sine	The sine wave is the simplest possible waveform, with no harmonics (overtones). The sine wave produces a neutral, soft timbre.
Formant 1–12	Formant waveforms emphasizes certain frequency bands. Like the human voice, musical instruments have a fixed set of formants, which give it a unique, recognizable tonal color or timbre, regardless of pitch.
Vocal 1–7	These are also formant waveforms, but specifically vocal-oriented. Vowel sounds (A/E/I/O/U) are among the waveforms found in this category.
Partial 1–7	Partials, also called harmonics or overtones, are a series of tones which accompany the prime tone (fundamental). These waveforms could be described as producing intervals with two or more frequencies heard simultaneously with equal strength.
Reso Pulse 1–12	This waveform category begins with a complex waveform (Reso Pulse 1), that emphasizes the fundamental frequency (prime). For each consecutive waveform in this category, the next harmonic in the harmonic series is emphasized.
Slope 1–12	This waveform category begins with a complex waveform (Slope 1), with gradually decreasing harmonic complexity the higher the number selected. Slope 12 produces a sine wave (no harmonics).
Neg Slope 1–9	This category also begins with a complex waveform (NegSlope 1), but with gradually decreasing low frequency content the higher the number selected.

- To hear the signal generated by the oscillator(s), the corresponding Osc dial in the oscillator sections must be turned clockwise to a suitable value.

OSC 1 parameters

Oscillator 1 acts as a master oscillator. It determines the base pitch for all three oscillators. Oscillator 1 features the following parameters:

Parameter	Value	Description
Osc 1	0–100	This controls the output level of the oscillator.
Coarse	+/- 48 semitones	This determines the base pitch used by all oscillators.
Fine	+/- 50 cent	Fine tunes the oscillator pitch in cent increments (100th of a semitone). This also affects all oscillators.
Wave Mod	+/- 50	This parameter dial is only active if the Wave Mod button is activated beside the waveform selection box. Wave modulation works by adding a phase-shifted copy of the oscillator output to itself, which produces waveform variations. For example if a sawtooth waveform is used, activating WM will produce a pulse waveform. By modulating the WM parameter with for example a LFO, classic PWM (pulse width modulation) is produced. Wave modulation can, however, be applied to any waveform.
Phase button	On/Off	When Phase synchronization is activated, all oscillators will restart their waveform cycles with every note played. With Phase deactivated, the oscillators generate a waveform cycle continuously, which produces slight variations when playing as each note will start from a random phase in the cycle, adding warmth to the sound. But when synthesizing bass sounds or drum sounds, it is usually desired that the attack of every note played sounds the same, so for these purposes you should activate Phase sync. Phase sync also affects the noise generator.
Tracking button	On/Off	When Tracking is activated, the oscillator pitch will track the notes played on the keyboard. If Tracking is deactivated the oscillator pitch remains constant, regardless of what note is played.
Wave Mod button	On/Off	This switches wave modulation on or off.
Waveform pop-up	See "Selecting Waveforms" on page 47.	Sets the basic waveform for the oscillator.

OSC 2 parameters

Oscillator 2 has the following parameters:

Parameter	Value	Description
Osc 2	0–100	This controls the output level of the oscillator.
Coarse	+/- 48 semitones	This determines the coarse pitch for Osc 2. If FM is enabled, this determines frequency ratio of the oscillator regarding Osc 1.
Fine	+/- 50 cent	Fine tunes the oscillator pitch in cent increments (100th of a semitone). If FM is enabled, this determines the frequency ratio of the oscillator regarding Osc 1.

Parameter	Value	Description
Wave Mod	+/- 50	This parameter dial is only active if the Wave Mod button is activated beside the waveform selection box. Wave modulation works by adding a phase-shifted copy of the oscillator output to itself, which produces waveform variations. For example if a sawtooth waveform is used, activating WM will produce a pulse waveform. By modulating the WM parameter with for example a LFO, classic PWM (pulse width modulation) is produced. Wave modulation can, however, be applied to any waveform.
Ratio	1–16	This parameter dial (which is only active if the Freq Mod button is activated) adjusts the amount of frequency modulation applied to oscillator 2. See “About frequency modulation” on page 49. Is normally referred to as FM index.
Sync button	On/Off	When Sync is activated, Osc 2 is slaved to Osc 1. This means that every time Osc 1 completes its cycle, Osc 2 is forced to reset (start its cycle from the beginning). This produces a characteristic sound, suitable for lead playing. Osc 1 determines the pitch, and varying the pitch of Osc 2 produces changes in timbre. For classic sync sounds, try modulating the pitch of Osc 2 with an envelope or a LFO. The Osc 2 pitch should also be set higher than the pitch of Osc 1.
Tracking button	On/Off	When Tracking is activated, the oscillator pitch will track the notes played on the keyboard. If Tracking is deactivated the oscillator pitch remains constant, regardless of what note is played.
Freq Mod button	On/Off	This switches frequency modulation on or off.
Wave Mod button	On/Off	This switches wave modulation on or off.
Waveform selector pop-up	See “Selecting Waveforms” on page 47.	Sets the basic waveform for the oscillator.

OSC 3 parameters

Oscillator 3 has the following parameters:

Parameter	Value	Description
Osc 3	0–100	This controls the output level of the oscillator.
Coarse	+/- 48 semitones	This determines the coarse pitch for Osc 3. If FM is enabled, this determines the frequency ratio of the oscillator regarding Osc 1/2.
Fine	+/- 50 cent	Fine tunes the oscillator pitch in cent increments (100th of a semitone). If FM is enabled, this determines the frequency ratio of the oscillator regarding Osc 1/2.
Ratio	1–16	This parameter dial (which is only active if the Freq Mod button is activated) adjusts the amount of frequency modulation applied to oscillator 3. See “About frequency modulation” on page 49. Is normally referred to as FM index.
Sync button	On/Off	When Sync is activated, Osc 3 is slaved to Osc 1. This means that every time Osc 1 completes its cycle, Osc 3 is forced to reset (start its cycle from the beginning). This produces a characteristic sound, suitable for lead playing. Osc 1 determines the pitch, and varying the pitch of Osc 3 produces changes in timbre. For classic sync sounds, try modulating the pitch of Osc 3 with an envelope or a LFO. The Osc 3 pitch should also be set higher than the pitch of Osc 1.
Tracking button	On/Off	When Tracking is activated, the oscillator pitch will track the notes played on the keyboard. If Tracking is deactivated the oscillator pitch remains constant, regardless of what note is played.
Freq Mod button	On/Off	This switches frequency modulation on or off.
Wave Mod button	On/Off	This switches wave modulation on or off.
Waveform selector pop-up	See “Selecting Waveforms” on page 47.	Sets the basic waveform for the oscillator.

About frequency modulation

Frequency modulation or FM means that the frequency of one oscillator (called the carrier) is modulated by the frequency of another oscillator (called the modulator).

- In Prologue, Osc 1 is the modulator, and Osc 2 and 3 are carriers.

  Osc 2 could be said to be both carrier and modulator as if Freq Mod is applied to Osc 2 it is modulated by Osc 3. If Osc 2 also uses frequency modulation, Osc 3 will be modulated by both Osc 1 and Osc 2.

- The “pure” sound of frequency modulation is output through the modulator oscillator(s).

  This means that you should turn off the Osc 1 output when using frequency modulation.

- The Freq Mod button switches frequency modulation on or off.

- The Ratio parameter determines the amount of frequency modulation.

Portamento

This parameter makes the pitch glide between the notes you play. The parameter setting determines the time it takes for the pitch to glide from one note to the next. Turn the knob clockwise for longer glide time.

The “Mode” switch allows you to apply glide only when you play a legato note (when switch is set to Legato). Legato is when you play a note without releasing the previously played note. Note that Legato mode only works with monophonic Parts.

Ring modulation

Ring modulators multiply two audio signals. The ring-modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals. In Prologue, Osc 1 is multiplied with Osc 2 to produce sum and difference frequencies. Ring modulation is often used to create bell-like sounds.

- To hear the ring modulation, you should turn down the output level for Osc 1 and 2, and turn up the “R.Mod” level all the way.

- If Osc 1 and 2 are tuned to the same frequency, and no modulation is applied to the Osc 2 pitch, nothing much will happen.
If you change the pitch of Osc 2, however, drastic changes in timbre can be heard. If the oscillators are tuned to a harmonic interval such as a fifth or octave, the ring modulated output will sound harmonic, other intervals will produce inharmonious, complex timbres.

- Oscillator Sync should be deactivated when using ring modulation.

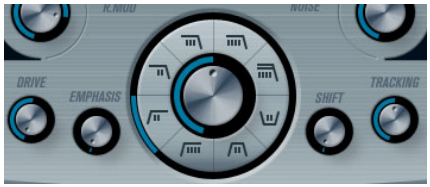
Noise Generator

A noise generator generates noise (all frequencies at equal levels). Applications include simulating drum sounds and breath sounds for wind instruments.

- To hear only the sound of the noise generator, you should turn down the output level for the oscillators, and turn up the Noise parameter.

- The Noise generator level is routed to Envelope 1 by default.
See “Envelope page” on page 53 for a description of the Envelope generators.

Filter section



The circle in the middle contains the filter parameters. The central dial sets the filter cut off parameter and the outer ring the filter type:

Parameter	Description
Filter type	Sets the filter type to either lowpass, highpass, bandpass or notch. The filter types are described on “About the filter types” on page 51.
Cut off	This dial controls the filter frequency or “cut off”. If a lowpass filter is used, it could be said to control the opening and closing of the filter, producing the classic “sweeping” synthesizer sound. How this parameter operates is governed by the filter type mode (see “About the filter types” on page 51).
Emphasis	This is the resonance control for the filter. For lowpass and highpass filters, raising the Emphasis value will emphasize the frequencies around the set cutoff frequency. This produces a generally thinner sound, but with a sharper, more pronounced cutoff sweep. The higher the filter Emphasis value, the more resonant the sound becomes until it starts to ring (self-oscillate), generating a distinct pitch. For Bandpass or Notch filters, the Emphasis setting adjusts the width of the band. When you raise the value, the band where frequencies are let through (Bandpass), or cut (Notch) will become narrower.
Drive	This can be used to adjust the filter input level. Levels above 0 dB will gradually introduce a soft distortion of the input signal, and a decrease of the filter resonance.
Shift	Internally, each filter consists of two or more “subfilters” connected in series. This parameter shifts the cutoff frequency of the subfilters. The result depends on the selected filter type: For Lowpass and Highpass filter types it changes the filter slope. For Bandpass and Notch filter types it changes the bandwidth. The Shift parameter has no effect if either the 12 dB LP or 12 dB HP filter type is selected.
Tracking	If this parameter is set to values over the 12 o’clock position, the filter cutoff frequency will increase the further up on the keyboard you play. Negative values invert this relationship. If the Track parameter is set fully clockwise, the cutoff frequency will track the keyboard by a semitone per key.

About the filter types

You select which filter type to use using the buttons around the filter cut off knob. The following filter types are available (listed clockwise from 9 o'clock):

Type	Description
12 db LP	This lowpass filter has a gentler slope (12 dB/Octave above the cutoff frequency), leaving more of the harmonics in the filtered sound.
18 dB LP	This lowpass filter also has a cascade design, attenuating frequencies above the cutoff frequency with a 18 dB/Octave slope, as used in the classic TB 303 synth.
24 dB LP	Lowpass filters let low frequencies pass and cut out the high frequencies. This filter type attenuates frequencies above the cutoff frequency with a 24 dB/Octave slope, which produces a warm and fat sound.
24 dB LP II	This lowpass filter has a cascade design which attenuates frequencies above the cutoff frequency with a 24 dB/Octave slope, which produces a warm and dark sound.
12 dB Band	A bandpass filter cuts both high and low frequencies above and below the cutoff frequency with a 12 dB/Octave slope, producing a nasal and thin sound.
12 dB Notch	A notch filter cuts off frequencies near the cutoff frequency by 12 dB/Octave, letting the frequencies below and above through. This produces a phaser-like sound.
12 dB HP	This highpass filter has a 12 dB/Octave slope, giving a bright and thin sound.
24 dB HP	A highpass filter is the opposite of a lowpass filter, cutting out the lower frequencies and letting the high frequencies pass. This filter has a 24 dB/Octave slope, giving a bright and sharp sound.

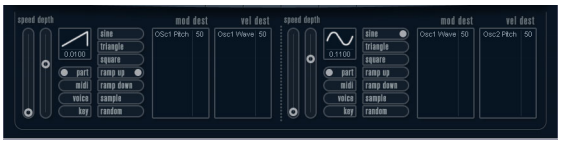
Master Volume and Pan

The master Volume controls the master volume (amplitude) of the instrument. By default this parameter is controlled by Envelope 1, to generate an amplitude envelope for the oscillators.

The Pan dial controls the position in the stereo spectrum for the instrument. You can use Pan as a modulation destination.

Modulation and controllers

The lower half of the control panel displays the various modulation and controller assignment pages available as well as the effect page. You switch between these pages using the buttons below the Filter section.



The following pages are available:

- The LFO page has two low frequency oscillators (LFOs) for modulating parameters – see below.
- The Envelope page contains the four Envelope generators which can be assigned to control parameters – see "Envelope page" on page 53.
- The Event page contains the common MIDI controllers (Mod wheel, Aftertouch etc. and their assignments – see "Event page" on page 54.
- The Effect page has three separate effect types available; Distortion, Delay and Modulation – see "Effects (EFX) page" on page 55.

LFO page

This is opened by clicking the LFO button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for two independent LFOs. Depending on the currently selected Preset, there may already be modulation destinations assigned, in which case these are listed in the "Mod Dest" box for each LFO – see "Assigning LFO modulation destinations" on page 52. A low frequency oscillator (LFO) is used for modulating parameters, for example the pitch of an oscillator (to produce vibrato), or for any parameter where cyclic modulation is desired.

The two LFOs have identical parameters:

Parameter	Description
Speed	This governs the rate of the LFO. If MIDI Sync is activated (see below), the available rate values are selectable as note values, e.g. beat increments of the sequencer tempo in Cubase.
Depth	This controls the amount of modulation applied by the LFO. If set to zero, no modulation is applied.
Waveform	This sets the LFO waveform.
Sync mode (Part/MIDI/ Voice/Key)	This sets the sync mode for the LFO. See below for a description.

About the sync modes

The Sync modes determine how the LFO cycle affects the notes you play:

Parameter	Description
Part	In this mode, the LFO cycle is free running and will affect all the voices in sync. “Free running” means that the LFO cycles continuously, and doesn’t reset when a note is played.
MIDI	In this mode the LFO rate is synced in various beat increments to MIDI clock.
Voice	In this mode each voice in the Part has its own independent LFO cycle (the LFO is polyphonic). These cycles are also free running – each key down starts anywhere in the LFO cycle phase.
Key	Same as Voice except that it is not free running – for each key down the LFO cycle starts over.

About the waveforms

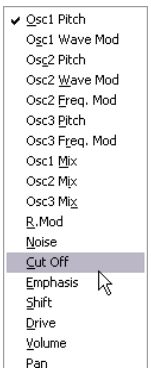
Most standard LFO waveforms are available for LFO modulation. You use Sine and Triangle waveforms for smooth modulation cycles, Square and Ramp up/down for different types of stepped modulation cycles and Random or Sample for random modulation. The Sample waveform is different. In this mode, one LFO actually samples and holds the values of the other LFO at the chosen frequency.

Assigning LFO modulation destinations

To assign a modulation destination for a LFO, proceed as follows:

1. Click in the “Mod Dest” box for one of the LFOs.
A pop-up appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

Click here…



…to open the modulation destination pop-up.

The modulation destination pop-up.

2. Select a destination, e.g. Filter Cut Off.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Select a suitable LFO Waveform, Speed, Depth and Sync mode.
You should now hear the filter cut off being modulated by the LFO.

4. Using the same basic method, you can add any number of modulation destinations for the LFO.
They will all be listed in the “Mod Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

Assigning LFO velocity destinations

You can also assign LFO modulation that is velocity controlled (i.e. governed by how hard or soft you strike a key). This is done as follows:

1. Click in the “Vel Dest” box for one of the LFOs.
A pop-up appears with all possible velocity destinations are shown.

2. Select a destination.
The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

- You can set positive and negative values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the LFO.
They will all be listed in the "Vel Dest" box.

- To remove a modulation destination click on its name in the list and select "Off" from the pop-up.

LFO modulation velocity control – an example:

If you follow the steps above and select the filter cut off parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the filter cut off parameter will be modulated by the LFO.

- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the filter cut off is modulated by the LFO.

Envelope page

The Envelope page is opened by clicking the ENV button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for the four independent envelope generators.

Envelope generators govern how a parameter value will change when a key is pressed, when a key is held and finally when a key is released.

The Envelope page

On the Envelope page, the parameters for one of the four envelope generators is shown at a time.

- You switch between the four envelopes in the section to the left.
Clicking on either of the four mini curve displays 1 to 4 will select it and display the corresponding envelope parameters to the right. The mini curve displays also reflect the envelope settings for each corresponding envelope.

- Envelope generators have four parameters; Attack, Decay, Sustain and Release (ADSR).
See below for a description of these.

- You can set envelope parameters in two ways; either by using the sliders or by click-dragging the curve in the Envelope curve display.
You can also do this in the mini curve displays.

- By default Envelope 1 is assigned to the master volume, and therefore acts as an amplitude envelope. The amplitude envelope is used to adjust how the volume of the sound should change from the time you press a key until the key is released.
If no amplitude envelope were assigned, there would be no output.

The Envelope parameters are as follows:

Attack

The attack phase is the time it takes from zero to the maximum value. How long this should take, depends on the Attack setting. If the Attack is set to "0", the maximum value is reached instantly. If this value is raised, it will take time before the maximum value is reached. Range is from 0.0 milliseconds to 91.1 seconds.

Decay

After the maximum value has been reached, the value starts to drop. How long this should take is governed by the Decay time parameter. The Decay time has no effect if the Sustain parameter is set to maximum. Range is from 0.0 milliseconds to 91.1 seconds.

Sustain

The Sustain parameter determines the level the envelope should rest at, after the Decay phase. Note that Sustain represents a level, whereas the other envelope parameters represent times. Range is from 0 to 100.

Release

Release determines the time it takes for the value to fall back to zero after releasing the key. Range is from 0.0 milliseconds to 91.1 seconds.

Punch

When Punch is activated, the start of the decay phase is delayed by a few milliseconds (i.e. the envelope remains at the top level for a moment before moving on to the decay phase). The result is a punchier attack similar to a compressor effect. This effect is more pronounced with short attack and decay times.

Retrigger

When Retrigger is activated, the envelope will re-trigger each time you play a new note. However, with certain textures/pad sounds and a limited number of voices it is recommended to leave the button deactivated, due to click noises that might occur, when the envelope is ended up abruptly. This is caused by the incoming re-trigger that forces the envelope to start over again.

Assigning Envelope modulation destinations

To assign a modulation destination for an Envelope, proceed as follows:

1. Click in the "Mod Dest" box for one of the Envelopes.
A pop-up appears with all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, e.g. Filter Cut Off.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Select a suitable envelope curve for the modulation.
You should now hear the filter cut off being modulated by the envelope as you play.

4. Using the same basic method, you can add any number of modulation destinations for the envelope.
They will all be listed in the "Mod Dest" box.

- To remove a modulation destination click on its name in the list and select "Off" from the pop-up.

Assigning Envelope velocity destinations

You can also assign Envelope modulation that is velocity controlled (i.e. governed by how hard or soft you strike a key). This is done as follows:

1. Click in the "Vel Dest" box for one of the envelopes.
A pop-up appears with all possible velocity destinations are shown.

2. Select a destination.
The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

- You can set positive and negative values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the Envelope.
They will all be listed in the "Vel Dest" box.

- To remove a modulation destination click on its name in the list and select "Off" from the pop-up.

Envelope modulation velocity control – an example:

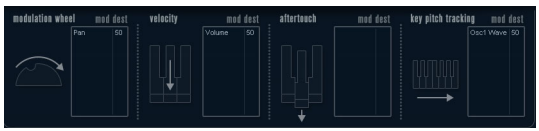
If you follow the steps above and select the filter cut off parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the filter cut off parameter will be modulated by the Envelope.

- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the filter cut off is modulated by the Envelope.

Event page

The Event page is opened by clicking the EVENT button at the top of the lower half of the control panel. This page contains the most common MIDI controllers and their respective assignments.



The Event page

The following controllers are available:

Controller	Description
Modulation Wheel	The modulation wheel on your keyboard can be used to modulate parameters.
Velocity	Velocity is used to control parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder.
Aftertouch	Aftertouch, or channel pressure, is MIDI data sent when pressure is applied to a keyboard after the key has been struck, and while it is being held down or sustained. Aftertouch is often routed to control filter cutoff, volume, and other parameters to add expression. Most (but not all) MIDI keyboards send Aftertouch.
Key Pitch Tracking	This can change parameter values linearly according to where on the keyboard you play.

To assign any of these controllers to one or several parameters, proceed as follows:

1. Click in the “Mod Dest” box for one of the controllers.

A pop-up appears with all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination.

The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount when the controller is at its full range.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.

To enter negative values type a minus sign followed by the value.

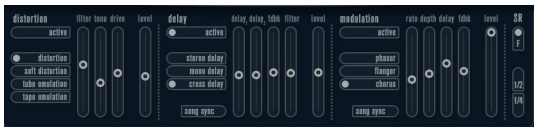
3. Using the same basic method, you can add any number of modulation destinations for the controllers.

They will all be listed in the “Mod Dest” box for the respective controller.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

Effects (EFX) page

This page features three separate effect units: Distortion, Delay and Modulation (Phaser/Flanger/Chorus). The Effect page is opened by clicking the EFX button at the top of the lower half of the control panel.



The Effects page

- Each separate effect section is laid out with a row of buttons that determine the effect type or characteristic and a row of sliders for making parameter settings.

- To activate an effect, click the “Active” button so that a dot appears.

Clicking again deactivates the effect.

Distortion

You can select between 4 basic distortion characteristics:

- Distortion provides hard clipping distortion.
- Soft Distortion provides soft clipping distortion.
- Tape Emulation produces distortion similar to magnetic tape saturation.
- Tube Emulation produces distortion similar to valve amplifiers.

The parameters are as follows:

Parameter	Description
Filter	This parameter sets the crossover frequency of the distortion filter. The distortion filter consists of a lowpass filter and a highpass filter with a cutoff frequency equal to the crossover frequency.
Tone	This parameter controls the relative amount of lowpass and high-pass filtered signal.
Drive	Sets the amount of distortion by amplifying the input signal.
Level	This controls the output level of the effect.

Delay

You can select between 3 basic delay characteristics:

- Stereo Delay has two separate delay lines panned left and right.
- In Mono Delay the two delay lines are connected in series for monophonic dual tap delay effects.

- In Cross Delay the delayed sound bounces between the stereo channels.

The parameters are as follows:

Parameter	Description
Song Sync	This switches tempo sync of the delay times on or off.
Delay 1	Sets the delay time ranging from 0ms to 728ms. If MIDI sync is activated the range is from 1/32 to 1/1; straight, triplet or dotted.
Delay 2	Same as Delay 1.
Feedback	This controls the decay of the delays. With higher settings the echoes repeat longer.
Filter	A lowpass filter is built into the feedback loop of the delay. This parameter controls the cutoff frequency of this feedback filter. Low settings result in successive echoes sounding darker.
Level	This controls the output level of the effect.

Modulation

You can select between 3 basic modulation characteristics:

- The Phaser uses an 8-pole allpass filter to produce the classic phasing effect.
- The Flanger is composed of two independent delay lines with feedback for the left and the right channel respectively. The delay time of both delays is modulated by one LFO with adjustable frequency.
- Chorus produces a rich chorus effect with 4 delays modulated by four independent LFOs.

The parameters are as follows:

Parameter	Description
Song Sync	This switches tempo sync of the Rate parameter on or off.
Rate	Sets the rate of the LFOs modulating the delay time. If Song Sync is activated the rate will be synced to various beat increments.
Depth	This parameter controls the depth of the delay time modulation.
Delay	This parameter sets the delay time of the four delay lines.
Feedback	The feedback parameter controls the amount of positive or negative feedback for all four delay lines. The adjustable range is from -1 to 1.
Level	This controls the output level of the effect.

SR parameters

With these buttons you can change the sample rate. Lower sample rates basically reduces the high frequency content and sound quality, but the pitch isn't altered. This is a great way to emulate the "lo-fi" sounds of older digital synths!

- If button "F" is active, the selected Part's program will play back with the sample rate set in the host application.
- If button "1/2" is active, the selected Part's program will play back with half the original sample rate.
- If button "1/4" is active, the selected Part's program will play back with a quarter of the original sample rate.
- A bonus effect of using lower sample rates is that it reduces the load on the computer CPU, allowing more simultaneous voices to be played etc.

Spector (Cubase only)

The synthesis in this synthesizer is based around a “spectrum filter”, which allows you to specify the frequency response by drawing a filter contour in the spectrum display. Slightly simplified, the signal path is the following:

- The starting point is the sound generated by up to 6 oscillators.
  You can choose between different numbers of oscillators in different configurations (in octaves, in unison, etc.). The oscillators can also be detuned for fat sounds or extreme special effects.
- Each oscillator produces two basic waveforms, labelled A and B.
  You can choose between six different waveforms, independently selected for A and B.
- The two waveforms pass through separate spectrum filters (A and B).
  You can draw different spectrum contours for the two filters, or select a contour from the included presets.
- The Cut 1 & 2 parameters allow you to shift the frequency range of the spectrum filter.
  This makes it easy to create unique-sounding filter sweeps.
- Finally, a Morph control lets you mix the output of spectrum filters A and B.
  Since this can be controlled with envelopes, LFOs etc. you can create morphing effects.
- You also have controllers and modulation parameters (two LFOs, four envelopes and three effects). See “Modulation and controllers” on page 59.

⇨ The signal flow of the Spector synth is illustrated in the section “Diagrams” on page 86.

Sound parameters

Oscillator section

A/B waveform pop-ups

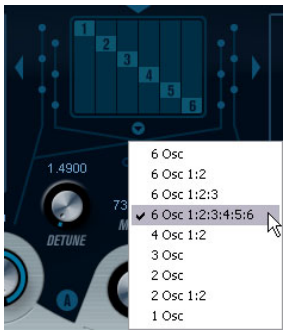
This is where you select basic waveforms for the A and B output of the oscillators. The options are especially suited for use with the spectrum filter.

Coarse and Fine

These parameters provide overall transposition and tuning of the oscillators (common for all oscillators, A and B waveforms).

Oscillator pop-up

This pop-up menu is opened by clicking on the text below the centrally placed section (which illustrates the currently selected oscillator configuration).



Click here to open the Oscillator pop-up.

The pop-up has the following oscillator configurations to choose between:

Option	Description
6 Osc	6 oscillators with the same pitch.
6 Osc 1:2	3 oscillators with base pitch and 3 pitched one octave down.
6 Osc 1:2:3	Three groups of two oscillators with the pitch ratio 1:2:3 (2 oscillators with base pitch, 2 oscillators at half the frequency of the base pitch and 2 oscillators at a third of the frequency).
6 Osc 1:2:3:4:5:6	6 oscillators tuned with the pitch ratio 1:2:3:4:5:6 (known as the “subharmonic series”).
4 Osc 1:2	2 oscillators with base pitch and 2 pitched one octave down.
3 Osc	3 oscillators with the same pitch.
2 Osc	2 oscillators with the same pitch.
2 Osc 1:2	One oscillator with base pitch and one pitched one octave down.
1 Osc	A single oscillator. In this mode, the Detune and Cut II parameters are not active.

Detune

Detunes the oscillators (in all oscillator modes except “1Osc”). Low values will give gentle chorus-like detuning; raising the control will detune the oscillators by several semitones for clangorous special effects.

Raster

This parameter reduces the number of harmonics present in the oscillator waveforms in the following manner:

Setting	Description
0	All harmonics present.
1	Only every second harmonic present.
2	Only every third harmonic present.
…	…and so on.

Portamento

This parameter makes the pitch glide between the notes you play. The parameter setting determines the time it takes for the pitch to glide from one note to the next. Turn the knob clockwise for longer glide time.

The “Mode” switch allows you to apply glide only when you play a legato note (when switch is set to Legato). Legato is when you play a note without releasing the previously played note. Note that Legato mode only works with monophonic Parts.

Spectrum filter section

This is where you create the contours (frequency response characteristics) for the two 128 pole resonant spectrum filters “A” and “B”.

- You can use the Preset pop-up menu to select a preset contour if you like.
- To change the contour, click and “draw” with the mouse.

Once you change the selected contour, it will be labeled as “Custom” in the Preset field above the display, indicating that you’re no longer using one of the presets.

- If you want to random calculate a spectrum filter curve, you can choose the Randomize function from the Preset pop-up.

Each time you choose this function, a new randomized spectrum will appear.

Cut I and II

These work much like cutoff frequency controls on a conventional filter: With the Cut controls at the maximum setting, the full frequency range will be used for the spectrum filter; lowering the Cut controls will gradually move the entire contour down in frequency, “closing” the filter. Please note the following:

- If a 2 oscillator configuration is used, you can set different “cutoffs” for the two oscillators with Cut I and Cut II, respectively. Similarly, if more than two oscillators are used, they are internally divided in two groups, for which you can set independent “cutoffs” with Cut I and II.

For example, in the “6 Osc” modes Cut I affects the sound of oscillators 1, 3 and 5 while Cut II affects the sound of oscillators 2, 4 and 6. In the “1 Osc” mode, the Cut II control is not used.

- If the Spectrum Sync (link symbol) button between the Cut controls is activated, the two knobs are synced and will follow each other and be set to the same value.

Morph

This controls the mix between the sound of spectrum filters A and B. When the Morph knob is turned fully left, only the “A” sound will be heard; when it’s turned right only the “B” sound will be heard. This allows you to seamlessly morph (manually or using an LFO or an envelope) between two totally different sounds.

Master Volume and Pan

The master Volume controls the master volume (amplitude) of the instrument. By default this parameter is controlled by Envelope 1, to generate an amplitude envelope for the oscillators.

The Pan dial controls the position in the stereo spectrum for the instrument. You can use Pan as a modulation destination.

Modulation and controllers

The lower half of the control panel displays the various modulation and controller assignment pages available as well as the effect page. You switch between these pages using the buttons below the Morph section.

The following pages are available:

- The LFO page has two low frequency oscillators (LFOs) for modulating parameters – see below.
- The Envelope page contains the four Envelope generators which can be assigned to control parameters – see “Envelope page” on page 61.
- The Event page contains the common MIDI controllers (Mod wheel, Aftertouch etc. and their assignments – see “Event page” on page 62.
- The Effect page has three separate effect types available; Distortion, Delay and Modulation – see “Effects (EFX) page” on page 63.

LFO page

This is opened by clicking the LFO button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for two independent LFOs. Depending on the currently selected Preset, there may already be modulation destinations assigned, in which case these are listed in the “Mod Dest” box for each LFO – see “Assigning LFO modulation destinations” on page 60. A low frequency oscillator (LFO) is used for modulating parameters, for example the pitch of an oscillator (to produce vibrato), or for any parameter where cyclic modulation is desired.

The two LFOs have identical parameters:

Parameter	Description
Speed	This governs the rate of the LFO. If MIDI Sync is activated (see below), the available rate values are selectable as note values, so the rate will synced to the sequencer tempo in Cubase in various beat increments.
Depth	This controls the amount of modulation applied by the LFO. If set to zero, no modulation is applied.
Waveform	This sets the LFO waveform.
Sync mode (Part/MIDI/ Voice/Key)	This sets the sync mode for the LFO. See below for a description.

About the sync modes

The Sync modes determine how the LFO cycle affects the notes you play:

Parameter	Description
Part	In this mode, the LFO cycle is free running and will affect all the voices in sync. “Free running” means that the LFO cycles continuously, and doesn’t reset when a note is played.
MIDI	In this mode the LFO rate is synced in various beat increments to MIDI clock.
Voice	In this mode each voice in the Part has its own independent LFO cycle (the LFO is polyphonic). These cycles are also free running – each key down starts anywhere in the LFO cycle phase.
Key	Same as Voice except that it is not free running – for each key down the LFO cycle starts over.

About the waveforms

Most standard LFO waveforms are available for LFO modulation. You use Sine and Triangle waveforms for smooth modulation cycles, Square and Ramp up/down for different types of stepped modulation cycles and Random or Sample for random modulation. The Sample waveform is different:

- In this mode, the LFO actually makes use of the other LFO as well.

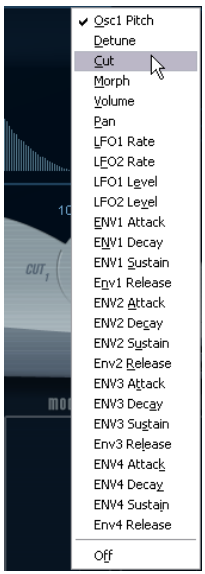
For example, if LFO 2 is set to use Sample the resulting effect will also depend on the speed and waveform of LFO 1.

Assigning LFO modulation destinations

To assign a modulation destination for a LFO, proceed as follows:

1. Click in the “Mod Dest” box for one of the LFOs.

A pop-up appears with all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.



The modulation destination pop-up.

2. Select a destination, e.g. Cut.

The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.

To enter negative values type a minus sign followed by the value.

3. Select a suitable LFO Waveform, Speed, Depth and Sync mode.

You should now hear the Cut parameter being modulated by the LFO.

4. Using the same basic method, you can add any number of modulation destinations for the LFO.

They will all be listed in the “Mod Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

Assigning LFO velocity destinations

You can also assign LFO modulation that is velocity controlled (i.e. governed by how hard or soft you strike a key). This is done as follows:

1. Click in the “Vel Dest” box for one of the LFOs.

A pop-up appears with all possible velocity destinations are shown.

2. Select a destination.

The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

- You can set positive and negative values by clicking on the value in the list, typing in a new value and pressing the Enter key.

To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the LFO.

They will all be listed in the “Vel Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

LFO modulation velocity control – an example:

If you follow the steps above and select the Cut parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the Cut parameter will be modulated by the LFO.
- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the Cut parameter is modulated by the LFO.

Envelope page

The Envelope page is opened by clicking the ENV button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for the four independent envelope generators.

Envelope generators govern how a parameter value will change when a key is pressed, when a key is held and finally when a key is released.

The Envelope page

On the Envelope page, the parameters for one of the four envelope generators is shown at a time.

- You switch between the four envelopes in the section to the left.

Clicking on either of the four mini curve displays 1 to 4 will select it and display the corresponding envelope parameters to the right. The mini curve displays also reflect the envelope settings for each corresponding envelope.

- Envelope generators have four parameters; Attack, Decay, Sustain and Release (ADSR).

See below for a description of these.

- You can set envelope parameters in two ways; either by using the sliders or by click-dragging the curve in the Envelope curve display.

You can also do this in the mini curve displays.

- By default Envelope 1 is assigned to the master volume, and therefore acts as an amplitude envelope. The amplitude envelope is used to adjust how the volume of the sound should change from the time you press a key until the key is released.

If no amplitude envelope were assigned, there would be no output.

The Envelope parameters are as follows:

Attack

The attack phase is the time it takes from zero to the maximum value. How long this should take, depends on the Attack setting. If the Attack is set to “0”, the maximum value is reached instantly. If this value is raised, it will take time before the maximum value is reached. Range is from 0.0 milliseconds to 91.1 seconds.

Decay

After the maximum value has been reached, the value starts to drop. How long this should take is governed by the Decay time parameter. The Decay time has no effect if the Sustain parameter is set to maximum. Range is from 0.0 milliseconds to 91.1 seconds.

Sustain

The Sustain parameter determines the level the envelope should rest at, after the Decay phase. Note that Sustain represents a level, whereas the other envelope parameters represent times. Range is from 0 to 100.

Release

Release determines the time it takes for the value to fall back to zero after releasing the key. Range is from 0.0 milliseconds to 91.1 seconds.

Punch

When Punch is activated, the start of the decay phase is delayed a few milliseconds (the envelope “stays” at top level for a moment before moving on to the decay phase). The result is a punchier attack similar to a compressor effect. This effect is more pronounced with short attack and decay times.

Retrigger

When Retrigger is activated, the envelope will re-trigger each time you play a new note. However, with certain textures/pad sounds and a limited number of voices it is recommended to leave the button deactivated, due to click noises that might occur, when the envelope is ended up abruptly. This is caused by the incoming re-trigger that forces the envelope to start over again.

Assigning Envelope modulation destinations

To assign a modulation destination for an Envelope, proceed as follows:

1. Click in the “Mod Dest” box for one of the Envelopes.
A pop-up appears with all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, e.g. Cut.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Select a suitable envelope curve for the modulation.
You should now hear the Cut parameter being modulated by the envelope as you play.

4. Using the same basic method, you can add any number of modulation destinations for the envelope.
They will all be listed in the “Mod Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

Assigning Envelope velocity destinations

You can also assign Envelope modulation that is velocity controlled (i.e. governed by how hard or soft you strike a key). This is done as follows:

1. Click in the “Vel Dest” box for one of the envelopes.
A pop-up appears with all possible velocity destinations are shown.

2. Select a destination.
The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

- You can set positive and negative values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the Envelope.
They will all be listed in the “Vel Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

Envelope modulation velocity control – an example:

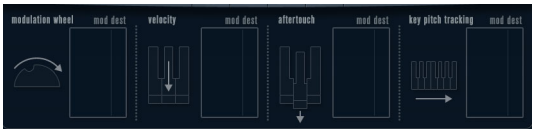
If you follow the steps above and select the Cut parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the parameter will be modulated by the Envelope.

- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the Cut parameter will be modulated by the Envelope.

Event page

The Event page is opened by clicking the EVENT button at the top of the lower half of the control panel. This page contains the most common MIDI controllers and their respective assignments.



The Event page

The following controllers are available:

Controller	Description
Modulation Wheel	The modulation wheel on your keyboard can be used to modulate parameters.
Velocity	Velocity is used to control parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder.
Aftertouch	Aftertouch, or channel pressure, is MIDI data sent when pressure is applied to a keyboard after the key has been struck, and while it is being held down or sustained. Aftertouch is often routed to control filter cutoff, volume, and other parameters to add expression. Most (but not all) MIDI keyboards send Aftertouch.
Key Pitch Tracking	This can change parameter values linearly according to where on the keyboard you play.

To assign any of these controllers to one or several parameters, proceed as follows:

1. Click in the “Mod Dest” box for one of the controllers.
A pop-up appears with all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount when the controller is at its full range.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of modulation destinations for the controllers.
They will all be listed in the "Mod Dest" box for the respective controller.

- To remove a modulation destination click on its name in the list and select "Off" from the pop-up.

Effects (EFX) page

This page features three separate effect units: Distortion, Delay and Modulation (Phaser/Flanger/Chorus). The Effect page is opened by clicking the EFX button at the top of the lower half of the control panel.

- Each separate effect section is laid out with a row of buttons that determine the effect type or characteristic and a row of sliders for making parameter settings.

- To activate an effect, click the "Active" button so that a dot appears.
Clicking again deactivates the effect.

Distortion

You can select between 4 basic distortion characteristics:

- Distortion provides hard clipping distortion.
- Soft Distortion provides soft clipping distortion.
- Tape Emulation produces distortion similar to magnetic tape saturation.
- Tube Emulation produces distortion similar to valve amplifiers.

The parameters are as follows:

Parameter	Description
Filter	This parameter sets the crossover frequency of the distortion filter. The distortion filter consists of a lowpass filter and a highpass filter with a cutoff frequency equal to the crossover frequency.
Tone	This parameter controls the relative amount of lowpass and high-pass filtered signal.
Drive	Sets the amount of distortion by amplifying the input signal.
Level	This controls the output level of the effect.

Delay

You can select between 3 basic delay characteristics:

- Stereo Delay has two separate delay lines panned left and right.
- In Mono Delay the two delay lines are connected in series for monophonic dual tap delay effects.
- In Cross Delay the delayed sound bounces between the stereo channels.

The parameters are as follows:

Parameter	Description
Song Sync	This switches tempo sync of the delay times on or off.
Delay 1	Sets the delay time ranging from 0 ms to 728 ms. If MIDI sync is activated the range is from 1/32 to 1/1; straight, triplet or dotted.
Delay 2	Same as Delay 1.
Feedback	This controls the decay of the delays. With higher settings the echoes repeat longer.
Filter	A lowpass filter is built into the feedback loop of the delay. This parameter controls the cutoff frequency of this feedback filter. Low settings result in successive echoes sounding darker.
Level	This controls the output level of the effect.

Modulation

You can select between 3 basic modulation characteristics:

- The Phaser uses an 8-pole allpass filter to produce the classic phasing effect.
- The Flanger is composed of two independent delay lines with feedback for the left and the right channel respectively. The delay time of both delays is modulated by one LFO with adjustable frequency.
- Chorus produces a rich chorus effect with 4 delays modulated by four independent LFOs.

The parameters are as follows:

Parameter	Description
Song Sync	This switches tempo sync of the Rate parameter on or off.
Rate	Sets the rate of the LFOs modulating the delay time. If Song Sync is activated the rate will be synced to various beat increments.
Depth	This parameter controls the depth of the delay time modulation.
Delay	This parameter sets the delay time of the four delay lines.

Parameter	Description
Feedback	The feedback parameter controls the amount of positive or negative feedback for all four delay lines. The adjustable range is from -1 to 1.
Level	This controls the output level of the effect.

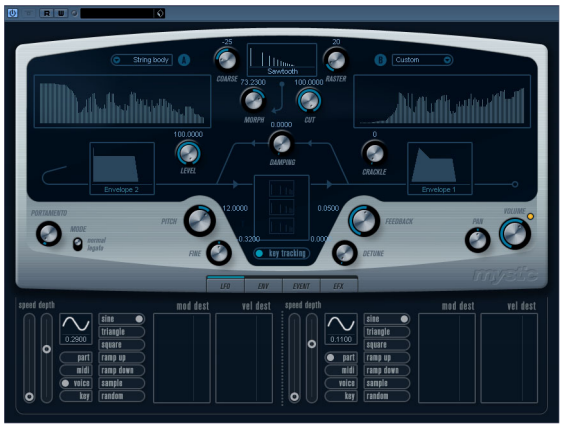
SR parameters

With these buttons you can change the sample rate. Lower sample rates basically reduces the high frequency content and sound quality, but the pitch isn't altered. This is a great way to emulate the "lo-fi" sounds of older digital synths!

- If button "F" is active, the selected Part's program will play back with the sample rate set in the host application.
- If button "1/2" is active, the selected Part's program will play back with half the original sample rate.
- If button "1/4" is active, the selected Part's program will play back with a quarter of the original sample rate.

- A bonus effect of using lower sample rates is that it reduces the load on the computer CPU, allowing more simultaneous voices to be played etc.

Mystic (Cubase only)



The synthesis method used by Mystic is based on three parallel comb filters with feedback. A comb filter is a filter with a number of "notches" in its frequency response, with the notch frequencies harmonically related to the frequency of the fundamental (lowest) notch.

A typical example of comb filtering occurs if you are using a flanger effect or a delay effect with very short delay time. As you probably know, raising the feedback (the amount of signal sent back into the delay or flanger) will cause a resonating tone – this tone is basically what the Mystic produces. As you will see, this astonishingly simple synthesis method is capable of generating a wide range of sounds, from gentle plucked-string tones to weird, non-harmonic timbres.

The basic principle is the following:

- You start with an "impulse sound", typically with a very short decay.

  The spectrum of the impulse sound will largely affect the tonal quality of the final sound. To set up an impulse sound on the Mystic you use a slightly simplified version of the synthesis found on the Spector synth.

- The impulse sound is fed into the three comb filters, in parallel. Each of these has a feedback loop.

  This means the output of each comb filter is fed back into the filter. This will result in a resonating feedback tone.

- When the signal is fed back into the comb filter, it goes via a separate, variable lowpass filter.
This filter corresponds to the damping of high frequencies in a physical instrument – when set to a low cutoff frequency it will cause high harmonics to decay faster than the lower harmonics (as when plucking a string on a guitar, for example).

- The level of the feedback signal is governed by a feedback control.
This determines the decay of the feedback tone. Setting this to a negative value will simulate the traveling wave in a tube with one open end and one closed end. The result is a more hollow, square wave-like sound, pitched one octave lower.

- A detune control offsets the fundamental frequencies of the three comb filters, for chorus-like sounds or drastic special effects.

Finally you have access to the common synth parameters – two LFOs, four envelopes and an effect section.

- By default, envelope 2 controls the level of the impulse sound – this is where you set up the short impulse decay when emulating string sounds etc.

⇨ The signal flow of the Mystic synth is illustrated in the section "Diagrams" on page 86.

Sound parameters

The Impulse Control section

This is where you set up the impulse sound – the sound fed into the comb filters, serving as a starting point for the sound. The Impulse Control has two basic waveforms that are filtered through separate spectrum filters with adjustable base frequency; the output is an adjustable mix between the two waveform/spectrum filter signals.

Spectrum displays

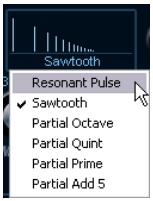
Allows you to draw a filter contour with your mouse for spectrum filters A & B.

- To set up the contour, click in one of the displays and drag the mouse to draw the desired curve. Note that this will produce the inverse contour in the other display, for maximum sonic versatility.
To set up the contour independently for the two filters, hold down [Shift] and click and drag the mouse in either display.

- Use the Preset pop-up menu to select a preset contour if you like.

- If you want to random calculate a spectrum filter curve, you can choose the Randomize function from the Preset pop-up.
Each time you choose this function, a new randomized spectrum will appear.

Waveform pop-up



The pop-up at the bottom of the waveform section (the central box at the top of the panel) allows you to select a basic waveform to be sent through filter contour A. The options are especially suited for use with the spectrum filter.

Cut

This offsets the frequency of the filter contour, working somewhat like a cutoff control on a standard synth filter. To use the filter contour in its full frequency range, set Cut to its maximum value.

Morph

Adjusts the mix between the two signal paths: waveform A spectrum contour A and waveform B spectrum contour B.

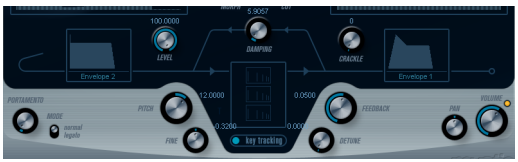
Coarse

This offsets the pitch for the impulse sound. In a typical “string setup”, when the impulse sound is very short, this will not change the pitch of the final sound, but the tonal color.

Raster

This removes harmonics from the impulse sound. As the harmonic content of the impulse sound is reflected in the comb filter sound, this will change the final timbre.

Comb filter sound parameters



Damping

This is a 6 dB/oct lowpass filter that affects the sound being fed back into the comb filters. This means the sound will become gradually softer when decaying, i.e. high harmonics to decay faster than the lower harmonics (as when plucking a string on a guitar, for example).

- The lower the Damping, the more pronounced this effect.

If you open the filter completely (turn Damping up to max) the harmonic content will be static – i.e. the sound will not get softer when decaying.

Level

This determines the level of the impulse sound being fed into the comb filters. By default, this parameter is modulated by envelope 2. That is, you use envelope 2 as a level envelope for the impulse sound.

- For a string-type sound, you want an envelope with a quick attack, a very short decay and no sustain (an “impulse” in other words), but you can also use other envelopes for other types of sounds.

Try raising the attack for example, or raising the sustain to allow the impulse sound to be heard together with the comb filter sound.

Crackle

This allows you to send noise directly into the comb filters. Small amounts of noise will produce a “crackling”, erratic effect; higher amounts will give a more pronounced noise sound.

Feedback

This determines the amount of signal sent back into the comb filters (the feedback level).

- Setting Feedback to zero (twelve o’clock) will effectively turn off the comb filter sound, as no feedback tone is produced.
- Setting Feedback to a positive value will create a feedback tone, with higher settings generating longer decays.
- Setting Feedback to a negative value will create a feedback tone with a more hollow sound, pitched one octave lower. Lower settings generate longer decays.

Detune

This offsets the notch frequencies of the three parallel comb filters, effectively changing the pitches of their feedback tones. At low settings, this creates a chorus-like detune effect. Higher settings detunes the three tones in wider intervals.

Pitch and Fine

Overall pitch adjustment of the final sound. This changes the pitch of both the impulse sound and the final comb filter sound.

Key Tracking

This button determines whether the impulse sound should track the keyboard or not. This will affect the sound of the comb filters in a way similar to a key track switch on a regular subtractive synth filter.

Portamento

This parameter makes the pitch glide between the notes you play. The parameter setting determines the time it takes for the pitch to glide from one note to the next. Turn the knob clockwise for longer glide time.

The “Mode” switch allows you to apply glide only when you play a legato note (when switch is set to Legato). Legato is when you play a note without releasing the previously played note. Note that Legato mode only works with monophonic Parts.

Master Volume and Pan

The master Volume controls the master volume (amplitude) of the instrument. By default this parameter is controlled by Envelope 1, to generate an amplitude envelope for the oscillators.

The Pan dial controls the position in the stereo spectrum for the instrument. You can use Pan as a modulation destination.

Modulation and controllers

The lower half of the control panel displays the various modulation and controller assignment pages available as well as the effect page. You switch between these pages using the buttons above this section.

The following pages are available:

- The LFO page has two low frequency oscillators (LFOs) for modulating parameters – see below.
- The Envelope page contains the four Envelope generators which can be assigned to control parameters – see “Envelope page” on page 69.
- The Event page contains the common MIDI controllers (Mod wheel, Aftertouch etc. and their assignments – see “Event page” on page 70.
- The Effect page has three separate effect types available; Distortion, Delay and Modulation – see “Effects (EFX) page” on page 71.

LFO page

This is opened by clicking the LFO button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for two independent LFOs. Depending on the currently selected Preset, there may already be modulation destinations assigned, in which case these are listed in the “Mod Dest” box for each LFO – see “Assigning LFO modulation destinations” on page 68.

A low frequency oscillator (LFO) is used for modulating parameters, for example the pitch of an oscillator (to produce vibrato), or for any parameter where cyclic modulation is desired.

The two LFOs have identical parameters:

Parameter	Description
Speed	This governs the rate of the LFO. If MIDI Sync is activated (see below), the available rate values are selectable as note values, so the rate will synced to the sequencer tempo in Cubase in various beat increments.
Depth	This controls the amount of modulation applied by the LFO. If set to zero, no modulation is applied.
Waveform	This sets the LFO waveform.
Sync mode (Part/MIDI/ Voice/Key)	This sets the sync mode for the LFO. See below for a description.

About the sync modes

The Sync modes determine how the LFO cycle affects the notes you play:

Parameter	Description
Part	In this mode, the LFO cycle is free running and will affect all the voices in sync. “Free running” means that the LFO cycles continuously, and doesn’t reset when a note is played.
MIDI	In this mode the LFO rate is synced in various beat increments to MIDI clock.
Voice	In this mode each voice in the Part has its own independent LFO cycle (the LFO is polyphonic). These cycles are also free running – each key down starts anywhere in the LFO cycle phase.
Key	Same as Voice except that it is not free running – for each key down the LFO cycle starts over.

About the waveforms

Most standard LFO waveforms are available for LFO modulation. You use Sine and Triangle waveforms for smooth modulation cycles, Square and Ramp up/down for different types of stepped modulation cycles and Random or Sample for random modulation. The Sample waveform is different:

- In this mode, the LFO actually makes use of the other LFO as well.

  For example, if LFO 2 is set to use Sample the resulting effect will also depend on the speed and waveform of LFO 1.

Assigning LFO modulation destinations

To assign a modulation destination for a LFO, proceed as follows:

1. Click in the “Mod Dest” box for one of the LFOs.
A pop-up appears with all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

The modulation destination pop-up.

2. Select a destination, e.g. Cut.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Select a suitable LFO Waveform, Speed, Depth and Sync mode.
You should now hear the Cut parameter being modulated by the LFO.

4. Using the same basic method, you can add any number of modulation destinations for the LFO.
They will all be listed in the “Mod Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

Assigning LFO velocity destinations

You can also assign LFO modulation that is velocity controlled (i.e. governed by how hard or soft you strike a key). This is done as follows:

1. Click in the “Vel Dest” box for one of the LFOs.
A pop-up appears with all possible velocity destinations are shown.

2. Select a destination.
The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

- You can set positive and negative values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the LFO.
They will all be listed in the “Vel Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

LFO modulation velocity control – an example:

If you follow the steps above and select the Cut parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the Cut parameter will be modulated by the LFO.

- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the Cut parameter is modulated by the LFO.

Envelope page

The Envelope page is opened by clicking the ENV button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for the four independent envelope generators.

Envelope generators govern how a parameter value will change when a key is pressed, when a key is held and finally when a key is released.

The Envelope page

On the Envelope page, the parameters for one of the four envelope generators is shown at a time.

- You switch between the four envelopes in the section to the left.

Clicking on either of the four mini curve displays 1 to 4 will select it and display the corresponding envelope parameters to the right. The mini curve displays also reflect the envelope settings for each corresponding envelope.

- Envelope generators have four parameters; Attack, Decay, Sustain and Release (ADSR).

See below for a description of these.

- You can set envelope parameters in two ways; either by using the sliders or by click-dragging the curve in the Envelope curve display.

You can also do this in the mini curve displays.

- By default Envelope 1 is assigned to the master volume, and therefore acts as an amplitude envelope. The amplitude envelope is used to adjust how the volume of the sound should change from the time you press a key until the key is released.

If no amplitude envelope were assigned, there would be no output.

- Envelope 2 is by default assigned to the Level parameter.

See “Level” on page 66.

The Envelope parameters are as follows:

Attack

The attack phase is the time it takes from zero to the maximum value. How long this should take, depends on the Attack setting. If the Attack is set to “0”, the maximum value is reached instantly. If this value is raised, it will take time before the maximum value is reached. Range is from 0.0 milliseconds to 91.1 seconds.

Decay

After the maximum value has been reached, the value starts to drop. How long this should take is governed by the Decay time parameter. The Decay time has no effect if the Sustain parameter is set to maximum. Range is from 0.0 milliseconds to 91.1 seconds.

Sustain

The Sustain parameter determines the level the envelope should rest at, after the Decay phase. Note that Sustain represents a level, whereas the other envelope parameters represent times. Range is from 0 to 100.

Release

Release determines the time it takes for the value to fall back to zero after releasing the key. Range is from 0.0 milliseconds to 91.1 seconds.

Punch

When Punch is activated, the start of the decay phase is delayed a few milliseconds (the envelope “stays” at top level for a moment before moving on to the decay phase). The result is a punchier attack similar to a compressor effect. This effect is more pronounced with short attack and decay times.

Retrigger

When Retrigger is activated, the envelope will re-trigger each time you play a new note. However, with certain textures/pad sounds and a limited number of voices it is recommended to leave the button deactivated, due to click noises that might occur, when the envelope is ended up abruptly. This is caused by the incoming re-trigger that forces the envelope to start over again.

Assigning Envelope modulation destinations

To assign a modulation destination for an Envelope, proceed as follows:

1. Click in the “Mod Dest” box for one of the Envelopes.
A pop-up appears with all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, e.g. Cut.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Select a suitable envelope curve for the modulation.
You should now hear the Cut parameter being modulated by the envelope as you play.

4. Using the same basic method, you can add any number of modulation destinations for the envelope.
They will all be listed in the “Mod Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

Assigning Envelope velocity destinations

You can also assign Envelope modulation that is velocity controlled (i.e. governed by how hard or soft you strike a key). This is done as follows:

1. Click in the “Vel Dest” box for one of the envelopes.
A pop-up appears with all possible velocity destinations are shown.

2. Select a destination.
The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

- You can set positive and negative values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the Envelope.
They will all be listed in the “Vel Dest” box.

- To remove a modulation destination click on its name in the list and select “Off” from the pop-up.

Envelope modulation velocity control – an example:

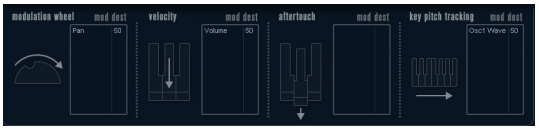
If you follow the steps above and select the Cut parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the parameter will be modulated by the Envelope.

- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the Cut parameter will be modulated by the Envelope.

Event page

The Event page is opened by clicking the EVENT button at the top of the lower half of the control panel. This page contains the most common MIDI controllers and their respective assignments.



The Event page

The following controllers are available:

Controller	Description
Modulation Wheel	The modulation wheel on your keyboard can be used to modulate parameters.
Velocity	Velocity is used to control parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder.
Aftertouch	Aftertouch, or channel pressure, is MIDI data sent when pressure is applied to a keyboard after the key has been struck, and while it is being held down or sustained. Aftertouch is often routed to control filter cutoff, volume, and other parameters to add expression. Most (but not all) MIDI keyboards send Aftertouch.
Key Pitch Tracking	This can change parameter values linearly according to where on the keyboard you play.

To assign any of these controllers to one or several parameters, proceed as follows:

1. Click in the “Mod Dest” box for one of the controllers.
A pop-up appears with all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount when the controller is at its full range.

- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the Enter key.
To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of modulation destinations for the controllers.
They will all be listed in the "Mod Dest" box for the respective controller.

- To remove a modulation destination click on its name in the list and select "Off" from the pop-up.

Effects (EFX) page

This page features three separate effect units: Distortion, Delay and Modulation (Phaser/Flanger/Chorus). The Effect page is opened by clicking the EFX button at the top of the lower half of the control panel.

- Each separate effect section is laid out with a row of buttons that determine the effect type or characteristic and a row of sliders for making parameter settings.

- To activate an effect, click the "Active" button so that a dot appears.
Clicking again deactivates the effect.

Distortion

You can select between 4 basic distortion characteristics:

- Distortion provides hard clipping distortion.
- Soft Distortion provides soft clipping distortion.
- Tape Emulation produces distortion similar to magnetic tape saturation.
- Tube Emulation produces distortion similar to valve amplifiers.

The parameters are as follows:

Parameter	Description
Drive	Sets the amount of distortion by amplifying the input signal.
Filter	This parameter sets the crossover frequency of the distortion filter. The distortion filter consists of a lowpass filter and a highpass filter with a cutoff frequency equal to the crossover frequency.
Tone	This parameter controls the relative amount of lowpass and high-pass filtered signal.
Level	This controls the output level of the effect.

Delay

You can select between 3 basic delay characteristics:

- Stereo Delay has two separate delay lines panned left and right.
- In Mono Delay the two delay lines are connected in series for monophonic dual tap delay effects.
- In Cross Delay the delayed sound bounces between the stereo channels.

The parameters are as follows:

Parameter	Description
Song Sync	This switches tempo sync of the delay times on or off.
Delay 1	Sets the delay time ranging from 0ms to 728ms. If MIDI sync is activated the range is from 1/32 to 1/1; straight, triplet or dotted.
Delay 2	Same as Delay 1.
Feedback	This controls the decay of the delays. With higher settings the echoes repeat longer.
Filter	A lowpass filter is built into the feedback loop of the delay. This parameter controls the cutoff frequency of this feedback filter. Low settings result in successive echoes sounding darker.
Level	This controls the output level of the effect.

Modulation

You can select between 3 basic modulation characteristics:

- The Phaser uses an 8-pole allpass filter to produce the classic phasing effect.
- The Flanger is composed of two independent delay lines with feedback for the left and the right channel respectively. The delay time of both delays is modulated by one LFO with adjustable frequency.
- Chorus produces a rich chorus effect with 4 delays modulated by four independent LFOs.

The parameters are as follows:

Parameter	Description
Song Sync	This switches tempo sync of the Rate parameter on or off.
Rate	Sets the rate of the LFOs modulating the delay time. If Song Sync is activated the rate will be synced to various beat increments.
Depth	This parameter controls the depth of the delay time modulation.
Delay	This parameter sets the delay time of the four delay lines.

Parameter	Description
Feedback	The feedback parameter controls the amount of positive or negative feedback for all four delay lines. The adjustable range is from -1 to 1.
Level	This controls the output level of the effect.

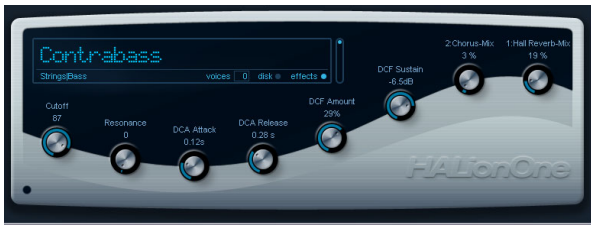
SR parameters

With these buttons you can change the sample rate. Lower sample rates basically reduces the high frequency content and sound quality, but the pitch isn't altered. This is a great way to emulate the "lo-fi" sounds of older digital synths!

- If button "F" is active, the selected Part's program will play back with the sample rate set in the host application.
- If button "1/2" is active, the selected Part's program will play back with half the original sample rate.
- If button "1/4" is active, the selected Part's program will play back with a quarter of the original sample rate.

- A bonus effect of using lower sample rates is that it reduces the load on the computer CPU, allowing more simultaneous voices to be played etc.

HALionOne



HALionOne is a sample player that can play sound content in the *.hsb (HALion Sound Bank) format. These samples have associated preset files that store the panel settings and reference the HSB samples. Included are several presets (as *.vstpreset and *.trackpreset files).

The operation of HALionOne is very simple; load a preset (a *.vstpreset or a *.trackpreset file for an Instrument Track) and start playing! You do, however, have the option to tweak the basic parameters to tailor the sound to your liking.

HALionOne parameters

The HALionOne differs from other VST Instruments in that the panel parameters shown can vary according to which parameters are stored in the HSB file. HSB files cannot be created with HALionOne – you need the full version of HALion to do this – but when created, certain parameters are assigned as part of the file and the associated program (or preset). This means that for each preset, only these assigned parameters are shown on the instrument panel. Typically, these are filter cutoff, DCA and DCF parameters and any assigned effect parameters (the effects are "built in").

If you load HALionOne for an Instrument track and select, for example, the "Draw Organ" preset, the following parameters are shown:

Parameter	Description
Cutoff	This allows you to adjust filter frequency or cutoff. The filter used is a Waldorf Low Pass filter with a 24 dB slope.
Resonance	Raising the filter resonance value will emphasize the frequencies around the set filter frequency.
DCF Amount	Controls the amount of the DCF (filter) envelope.
DCA Attack	Controls the time it takes for the DCA signal to reach its highest level.
DCA Decay	Controls the time it takes the DCA signal to decay to the sustain level.
DCA Sustain	Controls the DCA signal level after the Decay phase, as long as you press the key on your MIDI keyboard.
DCA Release	Controls the DCA signal after a key is released.
DCA Amount	Controls the amount of the DCA (amplifier) envelope.

These parameter assignments are used for many of the HALionOne presets, but not for all. As stated above, other parameters may be shown; these will be clearly labelled on the panel. For most of the presets there are also associated effects – the effect parameters are usually assigned to the quick controls on the right and typically control the dry/wet mix of the effect.

Effect Bypass

- This button, located at the bottom right in the box displaying the preset name, allows you to bypass any effects. The blue LED beside the button is lit if any effects are used in the preset.

Efficiency slider

The Efficiency slider provides a way of balancing audio quality vs. conservation of computer power. The lower the setting, the more voices are available. As a trade-off, sound quality is reduced.

Voices allocated

- The Voices field dynamically displays the number of voices currently used.

MIDI and Disk activity LEDs

The MIDI activity LED indicates received MIDI input. The Disk LED will light up green when samples are streamed from disk, and red when samples cannot be loaded from disk in time. In such a case you should consider lowering the Efficiency slider. When the disk LED doesn't light up, samples are read from memory.

Locate Contents

If you have moved the HALionOne content files to a different location (i.e. any other location than the folder in which it was stored at installation time), you need to use the Locate Contents function to inform HALion One about where to find its files. This is done as follows:

- Right-click anywhere on the control panel and select "Locate contents".

A file dialog opens where you can navigate to the folder location.

HALionOne and MIDI files

When the Preferences option "Import to Instrument Tracks" is activated (on the MIDI–MIDI File page), importing a MIDI file into Cubase will automatically set up instrument tracks, with HALionOne as the associated instrument. This allows you to quickly audition any imported MIDI files, to change parameter settings or to add effects, etc.

Groove Agent ONE

Groove Agent ONE is an easy-to-use sample-based MPC-style virtual drum machine for creating beats and reconstructing loops.

Audio samples can be associated with the Groove Agent ONE pads. Each pad is associated with a MIDI pitch, allowing you to trigger individual pads via MIDI notes.

To facilitate the creation of your own drum patterns, Groove Agent ONE provides a number of advanced functions.

Groups and pads

The pads and all functions related to the associating and auditioning of sounds can be found in the right half of the Groove Agent ONE panel.

Groove Agent ONE provides up to 128 pads, organized in eight groups of 16 pads. You can switch between the different groups by clicking on the corresponding group buttons (labeled 1 to 8) above the pads. Each pad is mapped to a particular MIDI note (C-2 to G8, which equals 128 notes).

- The button of the active group is highlighted. If one or more pads of a group have samples mapped to them, an additional red frame is displayed around group buttons.

By default, group 3 is active when you open Groove Agent ONE.

Pad functions

- The pads show the associated MIDI note in the top right corner.
  You can change the MIDI note by right-clicking it and selecting a different note from the pop-up menu.
- You can assign up to eight samples to a pad.
  See "Drag&drop of audio material" on page 74.
- If one or more samples have been assigned to a pad, the name of the first of these samples is displayed at the bottom of the pad.
  To change the name, right-click it, enter a new name and press [Enter]. This allows you, e.g., to indicate that more than one sample is mapped to this pad.
- To remove a sample assignment, click on the pad and drag the associated sample(s) to the trash icon in the LCD display to the left (see "Editing sounds" on page 75).
  Note that the trash icon is found only on either the Voice, Filter or Amplifier pages.
- The pad status is indicated by different colors.
  During playback, a pad will light up yellow for as long as a sample mapped to this pad is played back. When either the Voice, Filter or Amplifier button is activated in the Pad Edit section and you click on a pad, it will turn green to indicate that it is selected for editing. Unselected pads not playing back any samples are gray.
- You can mute a pad by [Shift]-clicking it.
  A prohibition symbol is displayed on the muted pad. To unmute, [Shift]-click once more.
- You can drag a sample from one pad to another pad.
  If the second pad already has a sample mapped to it, the sample assignment is swapped. Note that you can also swap the MIDI notes of the two pads by pressing [Shift] when dropping the sample.
- You can drag and drop samples between groups.
  Click on a pad that has a sample mapped to it, keep the mouse button pressed and move the mouse pointer over the button of another group. When the pad display now changes to display the pads of the other group, drag and drop the sample on the desired pad.

Velocity

- The velocity is determined by where on the pad you click: velocity is lowest at the bottom of the pad and highest at the top.
- You can force all pads to a velocity value of 127 by activating the V-Max button in the Global section in the top right corner of the Groove Agent ONE panel.

Resetting pads

You will find a Reset button in the Global section in the top right corner of the Groove Agent ONE panel. It allows you to clear all pad assignments of the current instance of Groove Agent ONE.

As a safety precaution, the Reset button is locked by default. Clicking the Reset button when it is locked will have no effect.

To unlock the Reset button, hold down the [Shift] key while clicking. The button color changes to red. When you click Reset now, all pad assignments are reset.

⚠ The Reset button is re-locked automatically five seconds after unlocking it.

Drag&drop of audio material

Groove Agent ONE provides advanced drag&drop support. You can drag one or more samples at the same time from Cubase onto Groove Agent ONE. Samples will either be mapped to the same pad, or to different pads.

You can drag files to Groove Agent ONE from the following Cubase locations:

- MediaBay
- Sample Editor (regions)
- Audio Part Editor

Layering samples on the same pad

When you select between one and eight samples and drag them to Groove Agent ONE, dropping them onto a pad (or onto the Layer indicator – see below) will automatically create a corresponding number of layers for this pad.

Drag&drop to several pads

Rather than dropping several samples to the same pad, you can also let Groove Agent ONE distribute samples across the available pads in one or several groups. To do so, press [Shift] and drop the samples onto a pad. The samples are mapped to the available pads, starting with the pad on which you initially dropped the samples, and then upwards according to the pad's MIDI pitch.

How many samples can be dropped to several pads depends on the number of pads available in your current instance of Groove Agent ONE. If Groove Agent ONE cannot supply a sufficient number of free pads for the number of dropped samples, a dialog is displayed in which you can confirm or cancel the operation.

Slicing a loop and triggering individual sounds via MIDI

Drag&drop to several pads has a number of uses. For example, it allows you to trigger individual sounds from an audio loop via MIDI. Proceed as follows:

1. Slice up a drum loop using the Sample Editor. Open the resulting audio part in the Audio Part Editor and press [Ctrl]/[Command]+[A] to select all audio events.
See the Operation Manual for details about slicing.

2. In the Audio Part Editor, click on one of the selected events and drag it to the Groove Agent ONE window.

3. Press the [Shift] key.

4. Point the mouse cursor at an empty pad and let go of the mouse key.
The individual samples from the audio part are now mapped to the available pads of Groove Agent ONE.

Now look at the Exchange section (to the left of the pads): the MIDI Export pad (the field displaying a double arrow) at the bottom of the section is lit. When mapping several samples to several pads, Groove Agent ONE creates a MIDI file containing all MIDI information to trigger these pads, and maps this file to the MIDI Export pad.

5. Drag this MIDI file from the MIDI Export pad to the Cubase Project window.
Dropping the file onto the Project window will create a new MIDI track. You can also drop the MIDI file to an existing MIDI track.

6. Play back the MIDI file.
The unedited MIDI file will play the same groove as the original audio loop. By editing the MIDI file you can change the original groove.

Saving and loading VST presets

You can save your current Groove Agent ONE configuration, including all settings for samples, pads and groups, as a VST preset.

1. At the top of the Groove Agent ONE window, click the VST Sound button and select “Save Preset” from the pop-up menu.
The Save Preset dialog is opened.

2. Enter a name for the new preset and click OK.
The preset is saved in the User Content folder on your system.

Proceed as follows to load an existing VST preset:

1. At the top of the Groove Agent ONE window, click the VST Sound button and select “Load Preset” from the pop-up menu.
The Presets browser is opened.

2. The Presets browser shows all presets it finds in the VST 3 Presets folder for Groove Agent ONE. Double-click the desired preset.
The Presets browser is closed and the preset is loaded into Groove Agent ONE.

- When a sample belonging to a preset cannot be found, Groove Agent ONE will display a standard file dialog in which you can navigate to the file.

Editing sounds

All sound editing functions can be found in and below the LCD display in the left half of the panel.

The LCD display can show four different sound editing pages, selected by clicking one of the four buttons in the Pad Edit section.

The information on the Play page refers to this instance of Groove Agent ONE as a whole. When the Play button is activated, the LCD display shows the name of the loaded VST preset and information on the number of samples and pads used by this instance of Groove Agent ONE. The Size parameter indicates the amount of RAM occupied by the currently loaded samples.

On the Voice, Filter and Amplifier pages, sample-specific data is displayed:

Parameter	Description
Brightness	Use the little slider at the very top of the LCD display to set the display brightness.
VST Preset	The name of a loaded VST Preset is displayed in the top left of the LCD display.
Sample/Pad	The name of the sample (and the pad to which it is assigned).
Trash icon	You can remove the current sample assignment by clicking on a pad or on the Layer indicator (see below) and dragging it onto the trash icon.

Parameter	Description
MIDI input off	When the MIDI symbol button in the top right corner of the LCD display is activated, the LCD display will show the waveform and parameter values of the currently playing sample. When this button is deactivated, the display will show only the data for the currently edit selected sample.
Layer indicator	The long bar near the top of the LCD display shows the active layer for the current pad. If more than one layer exist for the selected pad, the bar is divided accordingly. You can drag the dividing line between layers to change the velocity ranges of the layers. You can drag a new sample from the MediaBay and drop it directly onto the Layer indicator bar (this is the same as dropping a sample on a pad). You can drag layers to a different position on the bar.
Layer number	The layer number indicates which is the active layer of the current pad.
Sample	This is the name of the sample file.
Velocity	Here you can specify a velocity range for the current layer.
Coarse	Here you can tune the sample by up to ±12 semitones.
Fine	This parameter lets you finetune the sample by up to ±100 cents.
Volume	Sets the sample volume.
Waveform	The waveform of the current sample.

Depending on the selected page (Play, Voice, Filter, Amplifier), up to six quick controls with different pad-specific parameter assignments are displayed.

Play parameters

⇨ The parameter controls on the Play page are copies of the same parameters on the Voice, Filter and Amplifier pages.

The row of parameter controls below the LCD display shows six parameters:

Parameter	Description
Volume	The volume of the pad currently selected for editing.
Pan	The panorama setting of the pad currently selected for editing.
Coarse	Use this control to tune the pad by up to ±12 semitones.
Cutoff	Sets the filter cutoff frequency.
Q	Sets the filter resonance.
Output	Groove Agent ONE provides up to 16 stereo outputs. You can route pads to individual outputs using this control.

Voice parameters

The row of parameter controls below the LCD display shows six parameters:

Parameter	Description
Mode	Here you can reverse the currently selected sample so that you hear it backwards.
Coarse	Use this control to tune the pad by up to ±12 semitones.
Fine	Use this control to finetune the pad by up to ±100 cents.
Mute Gr.	With this control you can assign a pad to one of eight mute groups. Pads within a mute group will never play back simultaneously. New notes will cancel previous notes.
Tr. Mode	The sample of the currently selected pad is played either from start to finish (One Shot) or only for as long as you hold the mouse button/key (Key Hold). Key Hold can also be determined by the length of the corresponding MIDI note on your track.
Output	Groove Agent ONE provides up to 16 stereo outputs. You can route pads to individual outputs using this control. See the Operation Manual for information on how to use multitimbral instruments in Cubase.

Filter parameters

The row of parameter controls below the LCD display shows four parameters used to edit the Groove Agent ONE filter:

Parameter	Description
Type	Sets the filter type: low-pass (LP), high-pass (HP) or band-pass (BP). When you set this knob to OFF, the settings on this editing page have no effect.
Cutoff	Sets the filter cutoff frequency.
Q	Sets the filter resonance.
Mod	This parameter determines the influence that velocity has on the cutoff frequency. When set to 0%, the setting has no effect. When set to any other value, the cutoff frequency changes depending on the velocity.

Amplifier parameters

The row of parameter controls below the LCD display shows six parameters:

Parameter	Description
Volume	The volume of the pad currently selected for editing.
Pan	The panorama setting of the pad currently selected for editing.
Attack	Controls the amplifier envelope attack time.

Parameter	Description
Release	Controls the amplifier envelope release time. Reduce the release time to shorten the decay of sounds played in one-shot mode.
Amp Mod	This parameter determines the influence that velocity has on the pad volume setting. When set to 100%, the pad will sound louder the higher the velocity. When set to 0%, velocity will have no effect on the pad volume.
Attack Mod	This parameter determines the influence that velocity has on the Attack setting. When set to 0%, velocity will have no effect on the attack. When set to 100% and playing a pad with high velocity, the Attack time is increased by 50%. The higher the Attack Mod setting, the longer the additional attack time for a pad.

Master volume

In the Master section in the lower left of the Groove Agent ONE panel you can find a master volume slider that sets the output volume of the instrument.

The Exchange section

This section is used to import or export data to/from Groove Agent ONE.

The MIDI Export pad is described in detail in the section “Slicing a loop and triggering individual sounds via MIDI” on page 75.

Importing MPC files

Clicking the Import button opens a file dialog in which you can navigate to a .pgm file (.pgm is the AKAI MPC exchange format).

⇨ Note that Groove Agent ONE will import only the mapping data from the .pgm file. Any additional information (on MPC effects etc.) cannot be imported into Groove Agent ONE.

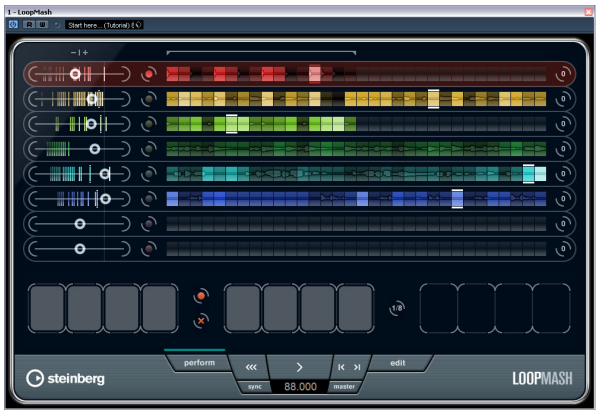
Automation of Groove Agent ONE parameters

When opening an automation subtrack for a track that uses Groove Agent ONE, you can select the following plug-in parameters from the Add Parameters dialog:

- Volume
- Pan
- Mute
- Cutoff
- Resonance

These parameters are available for the pads C1 to B4.

LoopMash (Cubase only)



LoopMash is one of a kind: a powerful tool for the slicing and instant re-assembling of any kind of rhythmic audio material. With LoopMash, you can preserve the rhythmic pattern of one audio loop, but you can replace all sounds of this loop with the sounds of up to seven other loops.

LoopMash is fully integrated into Cubase, which allows you to drag and drop audio loops from the MediaBay or Project window directly onto the LoopMash panel.

Getting started

To give you a first impression of what you can do with LoopMash, we have created a tutorial preset. Proceed as follows:

1. In Cubase, create an instrument track with LoopMash as the associated VST Instrument.
In the Inspector for the new track, click the Edit Instrument button to open the LoopMash panel. It has two main areas: the tracks section in the upper part of the panel, and the parameter section at the bottom.

2. At the top of the plug-in panel, click on the icon to the right of the Preset menu field and select Load Preset from the pop-up menu.

3. The Presets browser opens, showing presets found in the VST 3 Presets folder for LoopMash.

4. Select the preset called “A Good Start…(Tutorial)88”.
The Presets browser is closed and the preset is loaded into LoopMash.

5. At the bottom of the panel, make sure that the Sync button below the Transport controls is off, and start playback by clicking on the play button.

In the LoopMash panel, you can see a sliced loop waveform in the top (red) track. This track is selected (which is indicated by the track's background color and the lit button to the left of the waveform display).

The selected track holds the master loop. The rhythmic pattern of the LoopMash output is governed by the master loop – i.e. what you hear is the rhythmic pattern of this loop.

6. Look at the row of 12 pads below the track section: the first (leftmost) pad is selected. Select the third pad.
A new loop is displayed on the second track in the track display, and you will hear that the snare drum sound of the first loop has been replaced with a handclap sound.

7. Select the fifth, and then the seventh pad. Each time a new loop is added to the mash.
Note how the rhythmic pattern of the music stays the same, although an increasing number of sounds is taken from the other loops.

To the left in each track, you will find the similarity gain slider. These sliders are the most important control elements of LoopMash: the further to the right you move the similarity gain slider of a track, the more important the sounds of this particular loop become for the audible output of LoopMash.

How does LoopMash work?

Whenever you import a loop into LoopMash, the plug-in analyzes the audio material. It generates so-called "perceptual descriptors" (information on tempo, rhythm, spectrum, timbre etc.) and then slices the loop into eighth-note segments.

This means that after you have imported several loops, LoopMash will know the rhythmic pattern of each loop and the location of various sounds that make up this pattern within each loop. During playback, LoopMash uses the perceptual descriptors to determine how similar each slice is to the current slice of the master track.

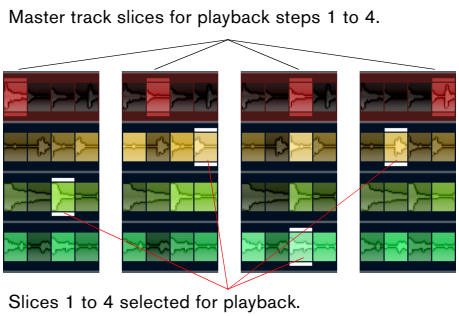
Please note that LoopMash does not categorize the sounds, but looks for overall similarity in the sound. For example, LoopMash might replace a low snare drum sound with a kick drum sound, even though a high snare sound is also available. LoopMash always tries to create a loop acoustically similar to the master loop, but using other sounds.

The similarity is shown by the brightness of each slice on each track, and also by the position of each slice on the similarity gain slider to the left of each track. The brighter a slice, the greater the similarity to the current master track slice, and the further to the right it is displayed on the similarity gain slider. Darker slices have smaller similarity and can be found further to the left on the slider.

The similarity gain settings of the various tracks determine which slice gets playback priority. This creates a new loop, over and over again, but with the rhythmic pattern of the original master loop.

In the following figure you can see four tracks. The track at the top is the master track. During playback, LoopMash moves through the master loop step-by-step (which is indicated by a horizontal line above and below the current slice) and automatically selects four different slices from these tracks to replace the slices of the master track. The currently playing slice is indicated by a white horizontal line above and below the slice:

The following shows the result of the selection process for each playback step:



Experiment with the provided LoopMash presets, and with your own loops of different lengths and with different rhythms, containing many different sounds – LoopMash is like an instrument, and we very much encourage you to play it!

LoopMash parameters

You can influence the process of constantly assembling a new loop with the various functions and parameter controls of LoopMash.

Track functions

- LoopMash provides advanced drag&drop support. You can drag single loop files from Cubase or the Explorer/Finder to the tracks on the LoopMash panel.

The quickest way to find the LoopMash content is to use the MediaBay: Open the VST Sound node and the LoopMash folder. Files can be dragged to LoopMash from the following Cubase locations: MediaBay, Project window, Pool, Sample Editor regions, Audio Part Editor. Dragging a loop to a track already occupied will replace the original loop.

- You can audition individual slices on each track by clicking on them.

You can also use the Step function in the transport controls (see below) to audition single slices.

- You can set a track transposition value.

Click on the button to the right of the waveform and select the desired transposition interval from the pop-up menu. The set value is displayed on the button. Note that this function is tied to the setting for the Slice Timestretch parameter (see below). When Slice Timestretch is deactivated, transposition is created by increasing/decreasing the playback speed of the slices (transposing a track up by one octave corresponds to playing the slices twice as fast). With Slice Timestretch on, you get true pitch shifting, i.e. there is no change in playback speed.

- To remove a loop from a LoopMash track, right-click the track and select “Remove from track”.

- One track is always selected. This is the master track: it provides the rhythmic pattern you hear, and it is the sounds of this loop that are replaced by slices selected from the other loops in the current LoopMash configuration.

Activate the button to the left of the waveform display to select this track and make it the master.

- A horizontal line above and below individual slices indicates the current playback position within the master loop (in the track color) and the slice currently selected for playback (in white).

- The Similarity Gain slider (to the left on each track) determines how important a particular track is for the “mashing up” of the master loop.

Move the slider to the right to select more slices from the current track for playback, and to the left to reduce the number of slices for playback (set to middle position by default). A thin white line intersects all similarity gain sliders – this is the “similarity threshold” (see below).

- The brightness of the slices changes when moving the similarity gain slider.

The further to the right, the lighter the color, and the higher playback priority for these slices. The currently playing slice is brightest.

- The vertical lines on the similarity gain slider correspond to the slices in this loop.

The changing pattern of slices indicates similarity of each slice, on all tracks, to the current master track slice. The further to the right a line is, the greater the similarity of this slice to the master slice. A slice must be to the right of the similarity threshold (see above) line to be considered for playback.

- A track can hold up to 32 slices.

Even if a long loop were to contain more than 32 slices, LoopMash will import only the first 32. Ideally, you would use a loop file cut at bar boundaries. When you import your file from the MediaBay, LoopMash will use the tempo information supplied by the MediaBay for the slicing of the loop.

- If you want to shorten the play length of the master loop, you can drag the bracket at the top of the track section.

You can drag the bracket handles, or you can drag the bracket as a whole. This allows you to select even a very small range within your master loop for playback – the rest of the loop is not taken into consideration. Note that short loop ranges (less than 1 bar) may conflict with the Jump interval setting (see below).

Transport controls

The transport controls can be found at the bottom of the LoopMash panel.

Button	Description
Play	Click the Play button to start or stop playback.
Locate	Click the Locate button to return to the beginning of the loop (bar 1/beat 1). Playback always starts automatically when clicking this button.
Step	Clicking in the left/right half of this button will step backwards/forwards through the timeline, playing one slice at a time.

Setting the LoopMash tempo

During playback, LoopMash can be synchronized to the tempo set in Cubase, or can follow its own tempo setting:

- Click the Sync button (to the left below the transport controls) to activate or deactivate synchronization to the project tempo set in Cubase.
When Sync is on, playback can be started using the Cubase transport controls. With Sync off, LoopMash will start playing only when you click the Play button in LoopMash.

- When the Sync button is deactivated, the current LoopMash tempo (in BPM) is displayed in the tempo field below the Play button.
To change the "local" tempo, click in the tempo field, enter a new value and press [Enter].

- When the Sync button is deactivated, you can click the Master button (to the right of the Sync button) to copy the tempo of the current master loop into the Tempo field.

The Edit page

Click the Edit button (to the right of the transport controls) to open the Edit page. These controls allow you to influence the way in which LoopMash plays back.

The following parameters are available:

Option	Description
Number of Voices	Here you can set the total number of slices from all tracks that will be used to replace the master slice (according to the current similarity gain settings). The range is from one (left) to four (right) voices, i.e. sounds from up to four loops can play simultaneously. Increasing the number of voices will increase the CPU load.
Voices per Track	This is the maximum number of slices that can be selected from a single track. The range is from one to four. The less slices can be picked from the same track, the more variety you will get in the LoopMash output.
Slice Selection Offset	Move this slider to the right to allow slices of smaller similarity to be selected for playback. This setting affects all tracks of this scene (see below).
Random Slice Selection	Move this slider to the right to allow more variation when selecting slices for playback, adding a more "random" feel to the selection process. This setting affects all tracks of this scene (see below).
Slice Quantize	Move this slider to the right to apply quantizing to the slices, i.e. the slices are aligned to a eighth-note grid. When the slider is all the way to the left, the slices will follow the rhythmic pattern defined by the original master loop.
Staccato Amount	When you move this slider to the right, the length of the slices is gradually reduced, giving the output a staccato feel.
Slice Timestretch	Use this option to apply realtime timestretching to the slices, filling gaps or avoiding overlaps between slices not played back at their original tempo, or when combining slices with different original tempos. Applying timestretch will increase the CPU load and may affect the sound quality. Reduce the need for timestretching by using loops with similar original tempos. See also the description of the track transposition value above.
Dry/Wet Mix	This sets the balance between the volumes of the master loop and the selected slices from the other tracks.

Scenes and the Performance page

Click the Perform button (to the left of the transport controls) to open the Performance page.

The settings you make on this page allow you to store LoopMash configurations so that you can recall them later.

Below the tracks, a row of 12 pads is displayed. You can save one "scene", a combination of up to eight tracks with all parameter settings, to each of these pads. This means that you can create a LoopMash configuration with up to 96 loops – 12 scenes with eight tracks each.

The following parameters are available:

Option	Description
Scene pad 1–12	Empty scene pads are black, pads with associated scenes are gray. The currently selected scene is white. Click on a pad to recall the corresponding scene.
Store Scene	To store a scene that you have set up, first click the round red button (between pads 4 and 5, at the top) and then a pad. This will save your setup to that pad.
Empty Scene	To remove a scene from a pad, first click the red x button (between pads 4 and 5, at the bottom) and then the desired pad.
Jump interval (1/8: Now; 1/4: Next beat; 1/2: Next half bar; 1: Next bar; e: End)	To set behavior when changing from one scene to the next during playback, click the button between pads 8 and 9. A pop-up menu opens, in which you can set at which point the change to the next scene will occur. End means that the current loop is played to the end before switching scenes. When you set up a short loop range (see above), you may need to set the interval to e to ensure that the jump point is reached.
MIDI control	If you have a MIDI keyboard connected to your computer, you can change between scenes by pressing keys on your keyboard. Pads 1–12 are mapped to the C–B keys (on all octaves).

⚠ Once you have set up a LoopMash configuration, you should save it to a scene pad. Changing scenes without saving means discarding any unsaved changes.

Saving and loading VST presets

You can save all current scenes as a VST preset. Proceed as follows:

1. At the top of the LoopMash window, click the icon to the right of the Preset field and select “Save Preset” from the pop-up menu.
The Save Preset dialog is opened.

2. Enter a name for the new preset and click OK.
The preset is saved in the User Content folder on your system. Make sure you tag your presets for better handling in the MediaBay.

To load an existing VST preset, proceed as follows:

1. At the top of the LoopMash window, click the icon to the right of the Preset field and select “Load Preset” from the pop-up menu.
The Presets browser is opened.

2. The Presets browser shows all presets it finds in the VST 3 Presets folder for LoopMash. Double-click the desired preset.
The Presets browser is closed and the preset is loaded into LoopMash.

- When a loop belonging to a preset cannot be found, LoopMash will display a standard file dialog in which you can navigate to the file.

Embracer – Surround Pad Synthesizer (Cubase only)

Embracer is a simple but powerful polyphonic synthesizer designed entirely for producing pads and accompaniment sounds. With its easy-to-use envelope and tone controls, it gives you fast access to the sounds you need without having to search through thousands of presets. However, the most powerful feature of Embracer is its surround output. With a single switch, you can turn the instrument from stereo to surround and the width control allows you to spread your pad sound anywhere from mono to stereo to full 360° surround. The unique “eye” controller gives you an exact idea of how the sound will be placed in a mix.

If you’ve never worked with a surround system before, now is the time to start exploring these possibilities.

The Embracer Surround Pad Synthesizer has the following properties:

- Embracer is a Polyphonic surround pad synthesizer.
- 2 oscillators with 12 waveforms.
- Independent envelope and tone controls.
- Stereo and surround outputs.
- Up to 32 voices of polyphony per instance.
- Dynamic width control for exciting 3D sounds.
- “Eye” controller for simultaneous tone and width control.
- Full MIDI control implementation.

Osc 1 and 2

Parameter	Description
Wave	Selects the waveform for each oscillator. Available waveforms are: Carpet, DigiPad, Choir, Ensemble, Metal Phaze, Phase Strings, Sing Sing, Soft Wave, Spit Strynx, Stepfloor, Submerged, Wave Bellz. Note: If you want to use only one oscillator, set the waveform to OFF. In this case only one voice per key will be used.
Tone	Embracer offers a high pass and low pass filter for each oscillator. Both filters are controlled via a single Tone knob. In the 50% center position, the signal will not be filtered. Reducing the tone value adds low pass filtering. Values above 50% add high pass filtering. This parameter can also be controlled by the "eye" controller.
Width	Controls the spatial spread of the signal. A value of 0% puts the signal mono into the center position. In stereo mode, a value of 100% results in a maximum stereo width. In surround mode, a value of 100% creates a full 360° surround image. The width parameter can be controlled by a variety of modulation sources, as well as by the "eye" controller.
Coarse (Oscillator 2 only)	Changes the pitch in semitones. Maximum range is +1/24 semitones = 2 octaves.
Fine (Oscillator 2 only)	Changes the pitch in fine steps with a range of up to +/- 50 cents. Note: If you want to create a slight detune effect between the oscillators, make sure to set the master tune parameter to a negative value of the same amount to keep the instrument in tune.

Envelope and Level

Parameter	Description
Attack	Controls the attack time of each oscillator. Higher values create slower attacks.
Attack Vel	Sets the amount of velocity control of the attack time. Higher values increase the velocity sensitivity.
Level	Controls the oscillator output level.
Level Vel	Sets the amount of velocity control of the oscillator level. Higher values increase the velocity sensitivity.

Master

Parameter	Description
Release	Controls the overall release time of the volume envelope. Higher values result in longer release times.
Mode	Sets the output mode of Embracer. You can choose between "Stereo" and "Surround". In Stereo Mode, Embracer has one stereo output in the VST Mixer. In Surround Mode, Embracer has either a quadraphonic 4-channel output or two independent stereo outputs in the Mixer. See below for more details on using Embracer in a surround mixer setup.
Width Ctr	Use this parameter to select a modulation source for the width parameter. Available sources are: Mod Wheel, Aftertouch, Velocity and Envelope. Both oscillators are controlled simultaneously. However, modulation depth is controlled independently by the respective width parameter of each oscillator.
Max Poly	Sets the total number of voices available. Each oscillator uses one voice per note played. Thus, a two-oscillator sound with 8 voices results in 4-voice polyphony. The default value for Max Poly is 16.
Fine Tune	Use this to adjust the pitch of the whole instrument. Range is +/- 50 cents. Use Fine Tune in combination with the Fine Tune parameter of OSC 2 to create smooth detune effects.
Master Out	Sets the overall output volume of the instrument.

The "Eye"

The Embracer's unique "Eye" controller offers a creative new way of controlling the sound's overall character and shape. This controller gives you access to several parameters at the same time.

For each oscillator, there is a circle representing the tone and width of the sound. Click and drag the corresponding circle to change its shape. There are also two (numbered) oscillator handles. You can drag these vertically to change the tone or horizontally to change the width of the respective oscillator. When you drag a handle, the respective Tone and Width knobs of the oscillator are adjusted accordingly. Play a note while editing to hear the effect.

The "eye" cannot only be used as a controller for the tone and width parameters, but also works as a surround scope for monitoring the spatial integration of the current sound. The display represents the sound's position in the stereo or surround sound field. In stereo mode, the sound position is shown only in the upper half of the display and represents the front part of the sound field. In surround mode, the sound position is shown in the upper and lower half of the display and represents the front and rear part of the sound field.

- You can use Embracer's automation feature to record the movements of the mouse within the "eye" controller!

Using Embracer in Surround Mode

When you want to enjoy Embracer in 3D, set it up in surround mode and listen to it on a surround system. Let's assume you have a surround monitoring system set up with your VST mixer and your VST output connections are properly set up.

1. Open an instance of Embracer in the VST instruments rack and set it to surround mode.

2. When you open the mixer you will see two separate stereo channels for the Embracer. The first is titled "Embracer" and the second "Embracer rear".

3. Assign both channel outputs to the surround output bus.

The two channel strips will now show independent surround panners. By default, the first output pair is assigned to the front left and right channels and the second output pair to the rear left and right channels. The surround width can be controlled with the "width" parameter.

4. Double-click on the surround panner to open its control panel. Set the "Mono/Stereo" parameter to either "Y-Mirror", "X-mirror" or "XY-mirror". You can now freely adjust the surround panning to your taste.

5. If your surround configuration includes a center or LFE channel, you can also add some of Embracer's signal to the center or LFE channels. Feel free to experiment to find out what works best in a given project and mix.

Monologue – Monophonic Analog Modeling Synthesizer (Cubase only)

Monologue is a monophonic analog synthesizer based on physical modeling technology. It offers full, rich and colorful sounds without consuming a lot of CPU power. The Monologue synthesizer is the perfect tool for bass, lead and sequenced sounds.

The Monophonic Analog Modeling Synthesizer has the following properties:

- 2 oscillators with sawtooth, square and triangle waveforms.
- An additional noise generator for white noise.
- Monologue has two filters: a high pass filter and a versatile multimode filter.
- Monologue has a single LFO.
- Monologue has 4-stage ADSR mod and amp envelopes.
- Monologue has an effects section with chorus, phaser, and flanger effects, plus separate delay and overdrive units.
- Monologue has a X/Y matrix pad for additional realtime modulation with access to all Monologue parameters.

Osc 1 and 2

Parameter	Description
Waveform (pop-up menu)	This is where you select the waveform: Saw, Square and Sub for oscillator 1 and Saw, Square and Triangle for Oscillator 2.
Coarse	Sets the coarse pitch in semitones. The available range is +/- one octave.
Fine	Allows you to fine-tune the pitch in cent increments. The available range is +/- 50 cents.

Parameter	Description
Depth	Controls the pitch modulation depth for the mod source defined in the “mod src” field. The available range is +/- one octave.
Mod Src	Defines the pitch modulation source. Available sources are: Modwheel, Aftertouch, Pitchbend, Velocity, LFO and Mod Env.
PWM (OSC2 only)	Controls the pulse width of the square wave. In the center position, pulse width is 50/50. Turning the PWM knob clockwise or counter clockwise creates a positive or negative pulse, respectively.
Sync (OSC2 only)	Activating the sync button synchronizes the pitch of oscillator 2 to the pitch of oscillator 1. When this is active, changing or modulating the pitch of oscillator 2 will change the tone and not the pitch. For the typical sync sound, turn osc 1 down in the mix and use osc 2 only.

Mix

Parameter	Description
Osc 1	Sets the pre-filter level for oscillator 1.
Noise	Sets the pre-filter noise level.
Osc 2	Sets the pre-filter level for oscillator 2.

Filter

Parameter	Description
Mode	Sets the filter type. Available filter types are 24 dB Low pass, 18 dB Low pass, 12 dB Low pass, 6 dB Low pass, 12 dB Band pass and 12 dB High pass.
Cutoff	Sets the filter cutoff frequency. How this parameter operates is governed by the filter type.
High Pass	Sets the cutoff frequency of the additional high-pass filter.
Res	Changes the resonance of the multi-mode filter. Full resonance puts the filter into self-oscillation.
Key Track	Determines the amount of key tracking applied to the filter cutoff frequency. The available range is 0 to 100%. A range of 100% tunes the filter cutoff frequency to the keyboards pitch 1:1.
Mod Src (A+B)	Defines the filter modulation source. The available sources are: Modwheel, Aftertouch, Pitchbend, Velocity, LFO, and Mod Env.
Depth (A+B)	Controls the filter modulation depth for the mod source set in the “mod src” field.

Envelope

Parameter	Description
A – (Attack)	Sets the attack time.
D – (Decay)	Sets the decay time.
S – (Sustain)	Sets the sustain level.
R – (Release)	Sets the release time.
Mod Src (A+B)	Defines the envelope modulation source. You can select: Modwheel, Aftertouch, Pitchbend, Velocity, LFO and Mod Env.
Depth (A+B)	Controls the envelope modulation depth for the mod source defined in the “mod src” field.

LFO

Parameter	Description
Waveform (pop-up menu)	Here, you can select the waveform for the low frequency oscillator. Available waveforms are: Triangle, Square, Sawtooth, Sample & Hold and Random.
Rate	Adjusts the frequency of the LFO, thus changing the rate of the modulation. Depending on the LFO sync parameter, you can edit the rate in Hertz or in note values.
Sync	When “Sync” is “on” the LFO speed will be synchronized to the sequencer’s tempo. This also affects the LFO rate format.
Mod Src	Defines the LFO modulation source. Available sources are: Modwheel, Aftertouch, Pitchbend, Velocity, LFO and Mod Env.
Depth	Controls the LFO modulation depth for the mod source defined in the “mod src” field.

X/Y Pad

Parameter	Description
X Par	Sets the parameter to be modulated on the x axis of the XY Pad. All of Monologue’s parameters are available as destinations.
Y Par	Sets the parameter to be modulated on the y axis of the XY Pad.
XY Pad	Use the mouse to control any two of Monologue’s parameters in combination. By moving the mouse horizontally, you can control the x parameter, by moving it vertically, you can control the y parameter. You can also record controller movements as automation data.

Effects

Parameter	Description
FX Type (pop-up menu)	Selects the effect type for Monologue’s pitch effects. The available types are Chorus, Flanger and Phaser.
Rate	Use this to adjust the rate of the effect modulation.
Depth	Use this to adjust the depth of the effect modulation.
FBK	Controls the feedback of the effect.
Mix	Controls the balance between dry and wet (effect) signal. Set to 0, the effect will be off. Set to 50, the balance between dry and wet signal is 50/50.

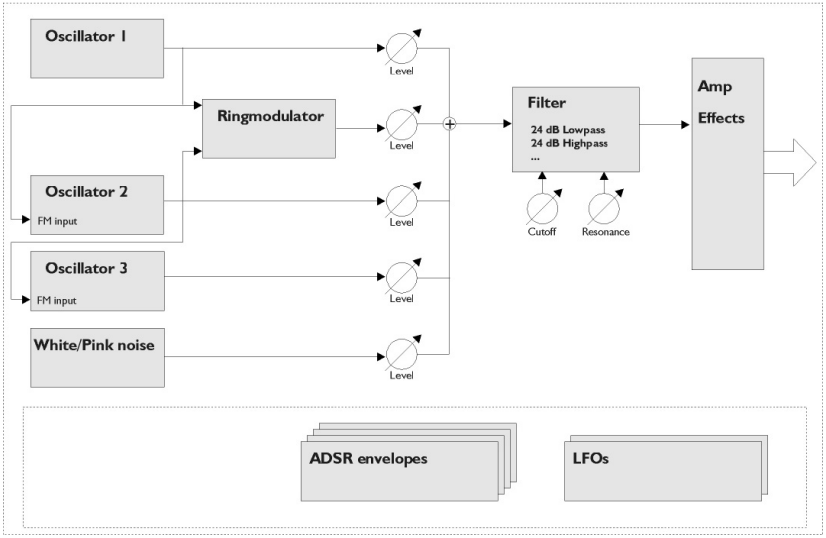
Parameter	Description
Overdrive	Controls the amount of overdrive (distortion) added to the signal. A slight amount of overdrive will create punch and bottom. Higher amounts will add distortion.
Delay	Sets the delay time in musical values. The delay effect is always in sync with the song tempo.
Spread	Controls the stereo spread of the delay signal. If you set this to 0, the delay will be centered mono. Higher amounts of spread will shift the left and right delay channels. If you set this to 100, the delays will "ping-pong" between the left and right channels at an even rate.
Tone	Adds a low pass filter to the delay. Increasing "tone" will make every delay repetition darker in tone.
FBK	Controls the amount of feedback of the delay. High feedback levels will create infinite delays. Use this parameter with caution.
Mix	Controls the balance between dry and wet (effect) signal. Set to 0, the effect will be off. Set to 50, the balance between dry and wet signal is 50/50.

Master

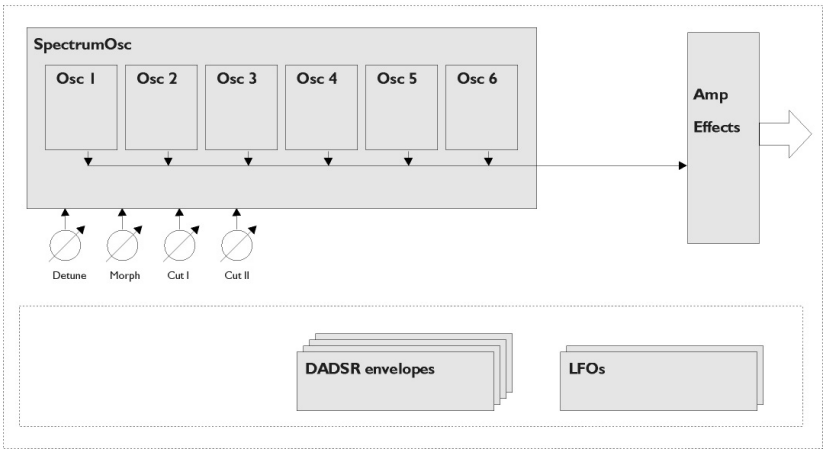
Parameter	Description
Glide Mode	The available modes are: "held", "on" and "off". With "held" selected, a glide effect only occurs for notes played legato.
Rate	Controls the glide rate – the time it takes for a note to reach its destination pitch.
PB Range	Controls the range of a pitch bend MIDI controller. Range can be set between 1 and 24 semitones for a total of two octaves.
Env Trigger	When set to "Multi", each keystroke will re-trigger the envelopes. When set to "single", legato notes will not retrigger the envelopes, effectively holding the envelopes on the sustain level until all keys are released before a new note is triggered.
Note Priority	Defines which note is played when multiple keys are held. Options are: First, Lowest, Highest, and Last.
Oct	Controls the master pitch of Monologue in octave steps. Range is +/- 4 octaves.
Master Out	Controls the master output level that is sent to the VST mixer. Use it to adjust the balance between different presets. Use the VST mixer channel volume to control or automate the Monologue master volume.
Keyboard	Pressing the "keyboard" button will reveal a six octave virtual keyboard. Pressing the "keyboard" button again will hide the keyboard and display the master section again.

Diagrams

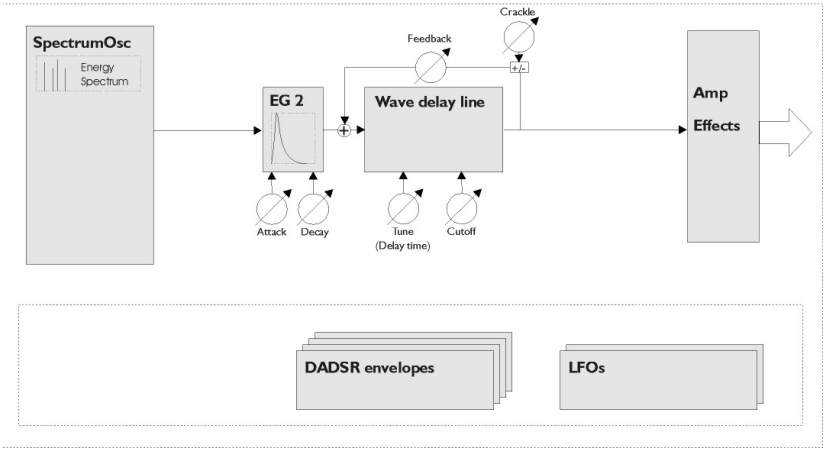
Prologue



Mystic



Spector


SpectrumOsc
Energy Spectrum
EG 2
Attack
Decay
Feedback
Crackle
+/-
Wave delay line
Tune
(Delay time)
Cutoff
Amp
Effects
DADSR envelopes
LFOs

3

MIDI effects

Introduction

This chapter describes the included MIDI realtime effects and their parameters.

How to apply and handle MIDI effects is described in the chapter “MIDI realtime parameters and effects” in the Operation Manual.

Arpache 5

A typical arpeggiator accepts a chord (a group of MIDI notes) as input, and plays back each note in the chord separately, with the playback order and speed set by the user. The Arpache 5 arpeggiator does just that, and more. Before describing the parameters, let’s look at how to create a simple, typical arpeggio:

1. Select a MIDI track and activate monitoring (or record enable it) so that you can play “thru” the track.
Make sure that the track is properly set up for playback to a suitable MIDI instrument.

2. Select and activate the arpeggiator.
For now, use it as an insert effect for the selected track.

3. In the arpeggiator panel, use the Quantize setting to set the arpeggio speed.
The speed is set as a note value, relative to the project tempo. For example, setting Quantize to “16” means the arpeggio will be a pattern of sixteenth notes.

4. Use the Gate setting to set the length of the arpeggio notes.
This allows you to create staccato arpeggios (Gate value smaller than the Quantize setting) or arpeggio notes that overlap each other (Gate value greater than Quantize).

5. Set the Key Range parameter to 12.
This will make the notes arpeggiate within an octave.

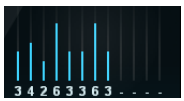
6. Play a chord on your MIDI instrument.
Now, instead of hearing the chord, you will hear the notes of the chord played one by one, in an arpeggio.

7. Try the different arpeggio modes by clicking the Play Order buttons.
The symbols on the buttons indicate the playback order for the notes (Invert, Up Only, etc.). The settings are described below.

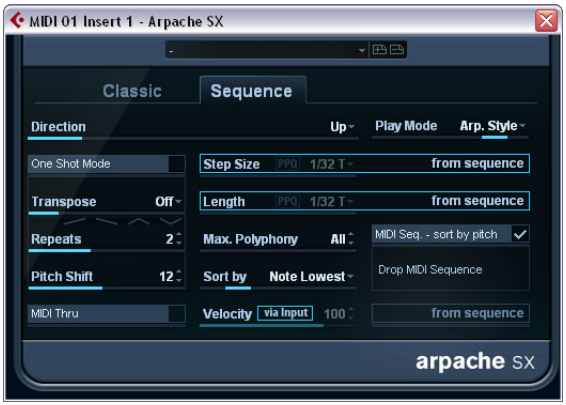
Parameters

The Arpache 5 has the following settings:

Setting	Description
Thru	If this is activated, the notes sent to the arpeggiator (i.e. the chord you play) will be passed through the plug-in (sent out together with the arpeggiated notes).
Play Order buttons	Allows you to select the playback order for the arpeggiated notes. The options are Normal, Invert, Up only, Down only, Random, User. If you select User, you can set the playback order manually using the 12 Play Order slots that are now shown at the bottom of the dialog.
Quantize	Determines the speed of the arpeggio, as a note value related to the project tempo. The range is 32T (1/32 note triplets) to “1.” (dotted note values).
Gate	Sets the length of the arpeggio notes, as a note value related to the project tempo. The range is the same as for the Quantize setting.
Key Range	Determines the arpeggiated note range, in semitones counted from the lowest key you play. This works as follows: – Any notes you play that are outside this range will be transposed in octave steps to fit within the range. – If the range is more than one octave, octave-transposed copies of the notes you play will be added to the arpeggio (as many octaves as fit within the range).

Setting	Description
Play Order slots	If the User play order is selected, you can use these “slots” to specify a custom playback order for the arpeggio notes: Each of the 12 slots corresponds to a position in the arpeggio pattern. For each slot, you specify which note should be played on that position by selecting a number. The numbers correspond to the keys you play, counted from the lowest pressed key. So, if you play the notes C3-E3-G3 (a C major chord), “1” would mean C3, “2” would mean E3, and “3” would mean G3. Note that you can use the same number in several slots, creating arpeggio patterns that are not possible using the standard play modes. Please note that you need to begin with the left-most slot and then fill the slots to the right. 
MIDI Thru	If this is activated, the notes sent to the arpeggiator (i.e. the chord you play) will pass through the plug-in (sent out together with the arpeggiated notes).

Arpache SX



This is an even more versatile and advanced arpeggiator, capable of creating anything from traditional arpeggios to complex, sequencer-like patterns. The Arpache SX has two different modes: Classic and Sequence.

Classic vs. Sequence mode

The Classic mode determines the basic behavior of the Arpache SX. When Sequence mode is selected, the Arpache SX uses the events of an additional MIDI part as a pattern. This pattern then forms the basis for the arpeggio, in conjunction with the MIDI input.

The following parameters are available:

Parameter	Description
Direction	This allows you to choose how the notes in the chord you play should be arpeggiated. In Classic mode you can choose a value from a pop-up menu, in Sequence mode you will find additional options, see below.
One Shot Mode	Activate this option if you want the phrase to be played only once. When this option is deactivated, the phrase will be looped.
Transpose	When a setting other than “Off” is selected, the arpeggio will be expanded upwards, downwards or both (depending on the mode). This is done by adding transposed repeats of the basic arpeggio pattern. and
Repeats	The “Repeats” setting sets the number of transposed repeats.
Pitch Shift	The “Pitch Shift” setting determines the transposition of each repeat.
MIDI Thru	If this is activated, the notes sent to the arpeggiator (i.e. the chord you play) will pass through the plug-in (sent out together with the arpeggiated notes).

Parameter	Description
Step Size	Determines the resolution of the arpeggio, i.e. its “speed” (in fixed note values or PPQ, if the PPQ button is activated). In Sequence mode you can also activate the “from sequence” option, see below.
Length	Determines the length of the arpeggio notes (in fixed note values or PPQ, if the PPQ button is activated). In Sequence mode you can also activate the “from sequence” option, see below.
Max. Polyphony	Determines how many notes should be accepted in the input chord. The “All” setting means there are no limitations.
Sort by	When you play a chord into the Arpache SX, the arpeggiator will sort the notes in the chord in the order specified here. For example, if you play a C-E-G chord, with “Note Lowest” selected, C will be the first note, E will be the second and G the third. This affects the result of the Arp Style setting.
Velocity	Determines the velocity of the notes in the arpeggio. Using the slider you can set a fixed velocity, or you can activate the “via Input” button to use the velocity values of the corresponding notes in the chord you play. In Sequence mode you can also activate the “from sequence” option, see below.

Sequence mode

In Sequence mode you can import a MIDI part into the Arpache SX by dragging it from the Project window and dropping it in the “Drop MIDI Sequence” field on the right of the Arpache SX panel.

Now, the notes in the dropped MIDI part will be sorted internally, either according to their pitch (“MIDI Seq. sort by pitch” checkbox activated) or according to their play order in the part. This results in a list of numbers. For example, if the notes in the MIDI part are C E G A E C and they are sorted according to pitch, the list of numbers will read 1 2 3 4 2 1. Here, there are 4 different notes/numbers and 6 trigger positions.

The MIDI input (the chord you send into the Arpache SX) will generate a list of numbers, with each note in the chord corresponding to a number depending on the “Sort by” setting.

Furthermore, the two lists of numbers will be matched – the Arpache SX tries to play back the pattern from the dropped MIDI part but using the notes from the MIDI input (chord). The result depends on the Play Mode setting:

Option	Description
Trigger	The whole pattern from the dropped MIDI file will be played back, but transposed according to one of the notes in the MIDI input. Which note is used for transposing depends on the Sort by setting.
Trigger Cnt.	As above, but even when all keys are released, the phrase continues playing from the last position (where it stopped), when a new key is pressed on the keyboard. This is typically used when playing “live” through the Arpache SX.
Sort Normal	Matches the notes in the MIDI input with the notes in the dropped MIDI part. If there are fewer notes (numbers) in the MIDI input, some steps in the resulting arpeggio will be empty.
Sort First	As above, but if there are fewer notes in the MIDI input, the missing notes will be replaced by the first note.
Sort Any	As above, but if there are fewer notes in the MIDI input, the missing notes will be replaced by any (random) note.
Arp. Style	As above, but if there are fewer notes in the MIDI input, the missing notes will be replaced by the last valid note in the arpeggio.
Repeat	In this mode, the chords played will not be separated into notes. Instead they will be used as is, and only the rhythm of the dropped MIDI part is used for playback.

Note also that you can choose to keep the original note timing, note length and note velocities from the dropped MIDI part, by selecting “from sequence” for the Step Size, Length and Velocity options.

Auto LFO

This plug-in works like an LFO in a synthesizer, allowing you to send out continuously changing MIDI controller messages. One typical use for this is automatic MIDI panning, but you can select any MIDI continuous controller event type. The Auto LFO effect has the following parameters:

Waveform

These settings determine the shape of the controller curves sent out. You can click on a waveform symbol, or choose a value from the pop-up menu.

Wavelength

This is where you set the speed of the Auto LFO, or rather the length of a single controller curve cycle. Using the slider or by choosing an entry from the pop-up menu, you can set this to rhythmically exact note values (or PPQ values if the PPQ button is activated). The lower the note value, the slower the speed. For example, if you set this to “1/8”, the waveform will be repeated every eighth note.

Controller Type

Determines which continuous controller type is sent out. Typical choices would include pan, volume and brightness, but your MIDI instrument may have controllers mapped to various settings, allowing you to modulate the synth parameter of your choice – check the MIDI implementation chart for your instrument for details!

Density

This determines the density of the controller curves sent out. The value can be set to “small”, “medium”, or “large”, or to rhythmically exact note values (by choosing from the pop-up menu). The higher the note value, the smoother the controller curve. For example, if you set this to “1/16”, a new controller event will be sent out at every 1/16 note position.

Value Range

These two sliders are used to determine the range of controller values sent out, i.e. the “bottom” and “top” of the controller curves.

Beat Designer

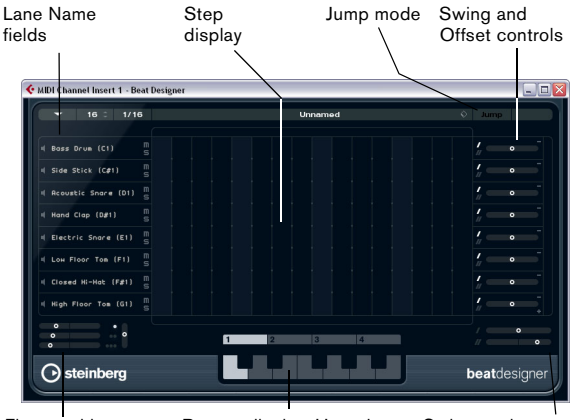
The Beat Designer is a MIDI pattern sequencer that allows you to create your own drum parts or “patterns” for a project. With the Beat Designer, you can quickly and easily set up the drums for a project, by experimenting and creating new drum sequences from scratch.

Normally, you will work on a short sequence, adjusting and modifying it while playing it back in a loop until you get the desired result. The drum patterns can then either be converted to MIDI parts on a track or triggered using MIDI notes during playback, see “Converting patterns into MIDI parts” on page 97 and “Triggering patterns” on page 97.

To use the Beat Designer, select it as MIDI insert effect for a MIDI track (routed to a VSTi or an external device) or an instrument track.

Overview

When you open the control panel for the Beat Designer for the first time, it shows a display with 8 empty lanes, each containing 16 steps.



Patterns and subbanks

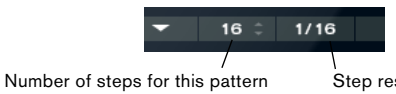
The Beat Designer patterns are saved as pattern banks. One pattern bank contains 4 subbanks which in turn contain 12 patterns each.

In the pattern display in the lower part of the Beat Designer, subbanks and patterns are displayed graphically. To select a subbank, click on a number (1 to 4) at the top of the display. To select a pattern within this subbank, click on a “key” in the keyboard display below.

Initial settings

The steps represent the beat positions in the pattern. You can specify the number of steps and the step resolution globally for a pattern:

- Click in the “Number of steps for this pattern” value field and enter the desired value.
  The maximum number of steps is 64. By default, 16 steps are shown.
- The playback length, i.e. the note value for the steps, can be specified in the Step resolution pop-up menu next to the Number of Steps setting.
  On this menu, you can also set triplet values. These also affect the Swing setting, see “The Swing setting” on page 95. The default setting is 1/16.



Selecting drum sounds

To specify a drum sound, click in the drum name field for a lane and select the desired drum sound from the pop-up menu. The available drum sounds depend on the selected drum map. If no drum map is selected for the track, the GM (General MIDI) drum names are used.

- To find the right sound, you can audition the selected drum sound by clicking the Preview Instrument button (the speaker icon).

Entering drum steps

To enter a drum step, click on the step field where you want to add a beat. You could e.g. add a snare drum on each downbeat for a lane and a bass drum on a second lane. When you click in an empty field, it becomes “filled”, indicating that you will hear a drum beat on this step.

You can also click and drag to enter a continuous range of drum steps.

⇨ When working on drum patterns, it is a good idea to play back a section of the project in a loop while inserting the drum sounds, as this allows you to hear the result immediately.

Removing steps

- To remove a drum step, simply click on the corresponding field again.
- To remove a range of drum steps, click and drag over them.

Setting the velocity

When entering a drum step, the velocity setting of this step is determined by where you click: Click in the upper part of a step for the highest velocity setting, in the middle section for a medium velocity and in the lower part for the lowest velocity setting. This is a quick way of roughly setting the velocity on the fly while entering drum sounds. In the display, the different velocity settings are indicated by different colors.

- You can fine-tune the velocity setting for an existing drum step by clicking on it and dragging up or down.
  The current velocity is indicated numerically while you drag, allowing you to find the desired setting easily. The available range is from 1 to 127.
- You can also fine-tune the velocity for a range of drum steps. Click on the first step, drag up or down to enter into velocity edit mode, and then drag sideways and up or down to modify the velocity for all the steps.
- If you hold down [Shift] while dragging up or down, you can change the velocity for all steps on a lane.

⇨ If you change the velocity for several steps at the same time, the relative velocity differences will be kept for as long as possible (until the minimum or maximum setting is reached).
The velocity for the steps will be increased or decreased by the same amount.

- You can also create a crescendo (or decrescendo) for an existing range of drum steps by holding down [Alt]/[Option], clicking on the first step, dragging up or down and then dragging to the left or right.

Editing operations

- You can move all drum steps on a lane by holding down [Shift], clicking on the lane and dragging to the left or right.
- You can also “invert” a lane, i.e. add drum sounds for all steps that were empty while removing all existing drum steps. This lets you create unusual rhythmic patterns. To do so, hold down [Alt]/[Option] and drag the mouse over the lane.
- You can copy the content of a lane onto another lane by holding down [Alt]/[Option], clicking in the section to the left of the lane you want to copy and dragging to the desired position.
  When you drag, a vertical line and a plus symbol will be displayed.

Lane handling

If you find that you have too many or too few lanes in the Beat Designer, you can add or remove them.

- To add a lane, click on the “Add Instrument Lane” button at the bottom right of the last lane shown.
- To remove a lane, click on the “Remove Instrument Lane” button in the controls section at the far right of the lane.
- You can change the order of the drum lanes by clicking in an empty area in the section to the left of a lane (i.e. not on a button) and dragging it to another position.
- You can mute or solo a lane by clicking the respective buttons to the left of the step display.

⚠ The lane operations always affect all patterns in the Beat Designer instance, not only the one you edit.

The Edit menu

This menu contains the following editing functions:

Option	Description
Shift Left	This moves all steps of the current pattern (all steps on all lanes) to the left.
Shift Right	This moves all steps of the current pattern (all steps on all lanes) to the right.
Reverse	Reverses the pattern, so that it plays backwards.
Copy Pattern	This copies the pattern to the clipboard. Copied patterns can be pasted into another pattern subbank (see below), and even directly into the project. The default key command for this is [Ctrl]/[Command]+[C].
Paste Pattern	Allows you to paste a complete pattern, e.g. into another pattern subbank, even into another instance of the Beat Designer. This is handy when you want to create variations based on existing patterns. The default key command for this is [Ctrl]/[Command]+[V].
Clear Pattern	This resets the current pattern.
Insert Pattern at Cursor	This creates a MIDI part for the current pattern and inserts it in the Project window, at the position of the project cursor. See also "Converting patterns into MIDI parts" on page 97.
Insert Subbank at Cursor	This creates a number of MIDI parts (one for each used pattern in the subbank) and inserts them one after the other, starting at the project cursor. See also "Converting patterns into MIDI parts" on page 97.
Insert Pattern at Left Locator	This creates a MIDI part for the current pattern and inserts it in the Project window, at the left locator. See also "Converting patterns into MIDI parts" on page 97.
Insert Subbank at Left Locator	This creates a number of MIDI parts (one for each used pattern in the subbank) and inserts them one after the other, starting at the left locator. See also "Converting patterns into MIDI parts" on page 97.
Fill Loop with Pattern	This creates a MIDI part for the current pattern and inserts it in the Project window as often as needed to fill the current loop area (the space between the left and right locators). See also "Converting patterns into MIDI parts" on page 97.

- You can set up key commands for the Insert options and the Fill Loop command in the Key Commands dialog.

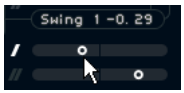
How to set up and use key commands is described in the chapter "Key Commands" in the Operation Manual.

The Swing setting

This parameter can be used to create a swing or shuffle rhythm, which allows you to add a more human feel to drum patterns that might otherwise be too static. This is done by offsetting every second drum step for a lane. If a triplet step resolution is used, every third drum step will be offset instead.

In the lower right section of the Beat Designer panel, you can find two Swing sliders. Dragging a slider to the right will delay every second (or third, see above) drum step in the pattern. Dragging to the left will make them play a little earlier.

You can set up two swing settings with these sliders and then quickly switch between these during playback. By default, the first swing setting is used (activated) in all lanes, but the slider is set to zero (middle position). Change the setting for this slider to hear how the pattern's feel changes.



Drag the upper fader to set swing setting I and the lower fader to set swing setting II.

You can switch between the two swing settings using the Swing buttons to the right of the step display.

Click on the buttons to select the respective swing setting or click on a selected button to deactivate swing for this lane.

Adding flams

The Flam parameter lets you add flams (short secondary drum hits just before or after the actual main drum beat).

You can add up to three flams for each pattern step:

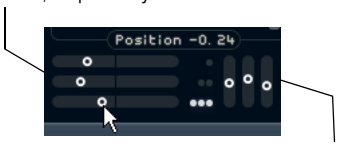
1. Click in the lower left corner of the step you want to add a flam to.
Little squares appear in the step when you point with the mouse at the step. After you clicked, the first square becomes filled to indicate that you added a flam.

Click here to add up to three flams to the step.

2. Click again to add the second and third flam, if needed.

3. In the lower left section of the Beat Designer panel you can make settings for the flams you created.

Here, you can specify the flam positions for all steps containing one, two and three flams, respectively.



With these sliders, you can specify the velocity for the separate flams.

- The first (topmost) Position slider specifies the flam position for all steps containing one single flam, the second slider the flam positions for all steps containing two flams, and the third slider the flam position for all steps containing three flams.
- Drag a Position slider to the left to add the flams before the drum step and to the right to add them after the step.
- When you add flams before the very first drum step in a pattern, this is indicated in the display by a small arrow in the top left corner of this step. This indicates that you have to treat this pattern with special care in playback and arranging. Starting playback at the normal pattern start would result in these flams not being played.
- Use the vertical sliders to the right of the flam sliders to set the velocity for the flams.

4. Start playback to hear the flams you created.

Offsetting lanes

To the right of the step display, you can find the Offset sliders for the lanes. These allow you to offset all drum steps on this lane. Drag a slider to the left to make the drum steps start a little earlier and to the right to let them start later.

Playing e.g. the bass drum or snare a little earlier allows you to add more “urgency” to the drums, delaying these drum sounds will result in a more relaxed drum pattern. Experiment with the settings to find out which fit best in your project.

Note that this function can also be used to correct faulty drum samples: If a drum sound has an attack that is slightly late, simply adjust the Offset slider for the lane.

Saving and loading presets

You can save all 48 Beat Designer patterns as a pattern bank. This can then be loaded in other projects. Pattern banks contain all the step and lane settings for a pattern (Mute and Solo, number and order of the lanes, pitch, etc.).

To save a pattern bank, proceed as follows:

1. In the Beat Designer, click on the Preset Management button to the right of the preset name field.

2. On the pop-up menu select “Save Preset”.
A dialog appears.

3. Enter a name for the preset and click OK.

The preset will now be available on the Preset browser, in the MediaBay and on the Apply Track preset pop-up menu in the Track list.

Pattern banks are handled much like Track presets in the MediaBay. For further information, refer to the chapters “The MediaBay” and “Track Presets” in the Operation Manual.

Using the drum patterns in your project

You can use the drum patterns created with the Beat Designer in two ways: either by converting them to MIDI parts on a MIDI or Instrument track or by triggering the different patterns using MIDI notes.

Converting patterns into MIDI parts

You can convert the drum patterns created in the Beat Designer into a MIDI part by dragging them into the Project window.

Proceed as follows:

1. Set up one or more patterns of the same subbank.

2. In the lower part of the window, click on a pattern or subbank and drag it at the desired position onto a MIDI or instrument track in the Project window.
If you drag the pattern or subbank to an empty area in the Project window, a new MIDI track is created. This will be an exact copy of the original track for which you opened the Beat Designer.

Click here and drag to convert this subbank into separate MIDI parts.

Click here and drag to convert this pattern into a MIDI part.

- If you drag a single pattern into the Project window, one MIDI part is created containing the drum sounds of the pattern.

- If you drag a subbank into the Project window, several MIDI parts (one for each used pattern in the subbank) are created and inserted one after the other in the project.

⚠ Only the used patterns in a subbank are inserted, i.e. if you did not enter drum steps in a pattern, this will not be converted into a MIDI part.

You can also use the Edit menu to insert patterns or subbanks into the project, see "The Edit menu" on page 95.

⚠ When you have created MIDI parts for your drum patterns this way, make sure to deactivate the Beat Designer, to avoid doubling of the drums. The Beat Designer will continue to play as long as it is activated.

- If you import patterns that sound before the first step (due to flams or lane offsets), the MIDI part will be lengthened accordingly.

The inserted MIDI parts can now be edited as usual in the project. You can e.g. fine-tune your settings in the Drum Editor.

⇨ Once a pattern is converted into a MIDI part, it cannot be opened in the Beat Designer again.

Triggering patterns

When you want to be able to modify your drum patterns in the Beat Designer while working on the project, you cannot convert them into parts, as these cannot be opened again in the Beat Designer. Instead, you can trigger the patterns from within the project.

You can trigger the patterns in the Beat Designer using Note On events. These can either be events on a MIDI track or be played live via a MIDI keyboard. Which pattern will be triggered depends on the pitch of the MIDI notes. The trigger range is four octaves starting with C1 (i.e. C1 to B4).

Proceed as follows:

1. Open the Beat Designer for a track.
Again, this can be a MIDI or an instrument track.

2. Click to the left of the Jump field to activate Jump mode.
In this mode, a MIDI note-on event will trigger a new pattern.

Click here to activate Jump mode.

- When you want to trigger the patterns using a MIDI part containing trigger events, you can specify whether the pattern will be switched directly (at the moment the event is received) or at the next bar: Click in the field to the right (where it says "Now") to activate the immediate switching of patterns. When this is activated, the word Now is displayed in white. When the word Now is black, patterns will switch at the beginning of the next bar in the project.

- When you want to trigger the patterns "live" via a MIDI keyboard, the new patterns are always played when the next bar in the project is reached.
Switching immediatelly would always produce an undesirable interruption in playback.

Now, you can trigger the patterns in the following way:

1. Play back the project and press a key on your MIDI keyboard to trigger the next pattern.
The pattern will start at the next bar line.

2. Create a MIDI part and enter notes at the positions in the project where you want to switch patterns.
Depending on the Jump mode setting, the new pattern will be played directly or start at the following bar.

- You can also drag a pattern or subbank into the Project when Jump mode is active to automatically create MIDI parts containing the trigger events.

⇨ When triggering a pattern that contains sound before the first step (due to flams or lane offsets), these are taken into account as well.

Chorder

The Chorder is a MIDI chord processor, allowing you to assign complete chords to single keys in a multitude of variations. These can then be played back live or using recorded notes on a MIDI track.

There are three main operating modes: “All Keys”, “One Octave”, and “Global Key”. You can switch between these modes using the Chords pop-up, see below.

For every key you can record up to eight different chords or variations on so-called “layers”. This is described in detail in the section “Using Layers” on page 99.

Operating modes

In the lower left section of the Chorder window, you can choose an option from the Chords pop-up menu to decide which keys on the piano roll will be used to record your chords.

The Chorder window

All Keys

In this mode, you can assign chords to each key on the piano roll. When you play any of these keys, you will hear the assigned chords instead.

One Octave

The One Octave mode is similar to the All Keys mode, but you can only set up chords for each key of a single octave (that is, up to eight different chords on twelve keys). When you play a note (e.g. C) on a different octave, you will hear a transposed version of the chords set up for this key.

Global Key

In Global Key mode, you can set up chords for a single key only. These chords (that you recorded on C-3) are then played by all keys on the keyboard, but transposed according to the note you play.

The chord indicator lane

At the top of the keyboard display you will find a thin lane with a small rectangle for each key that you can use to record a chord. These rectangles are shown in blue for all keys that already have chords assigned to them.

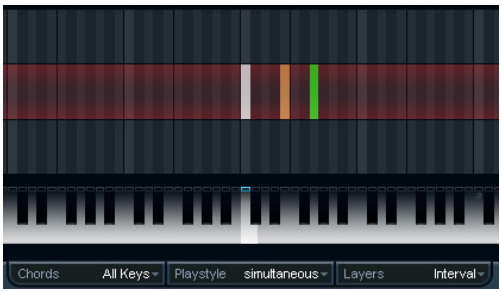
The chord indicator lane in One Octave mode with chords set up for 5 of the 12 available trigger keys.

⇨ In Global Key mode the C3 key has a special marking instead since this is the only key used in this mode.

Entering chords

To enter chords you need to switch to Learn mode. In this mode a transparent red bar indicates which element is ready for “learning” a note or chord. When you choose the trigger note for a chord, for examplem the piano roll is shown in red.

The piano roll in Learn mode



The second layer in Learn mode

Proceed as follows:

1. Click the Learn button at the bottom of the Chorder window to activate Learn mode.
The chord indicator lane is now tinted red, indicating that it is active.

2. Select the key to which you want to assign a chord by clicking on it on the piano roll display, or by pressing the key on a connected MIDI keyboard.
The red bar will now move to the first layer, indicating that you are ready to record the first chord.

⇨ In Global Key mode you do not have to choose a trigger key. The first layer is activated directly.

3. Play a chord on the MIDI keyboard and/or use the mouse to enter or change the chord in the layer display.
Any notes you enter are immediately shown in the Chorder display. The notes are shown in different colors, depending on the pitch.

- If you are entering chords via a MIDI keyboard, the Chorder will learn the chord as soon as you release all keys of your MIDI keyboard simultaneously.
  As long as a key is pressed, you can continue looking for the right chord.
- If more than one layer is shown, the Chorder will jump automatically to the next layer where you can record another chord.
  When all the layers for a key are filled, the red bar will jump back to the piano roll so that you can choose a different trigger key (in Global Key mode the Learn mode is deactivated instead).
- If you are entering chords with the mouse, the Chorder will not jump to the next layer automatically.
  You can select/deselect as many notes as you wish and then click on another layer or deactivate the Learn mode to continue.

4. Repeat the above with any other keys you wish to use.

Using Layers

The Layers pop-up menu at the bottom right of the window allows you to set up chord variations in the layer display above the piano roll. This works with all three modes and provides up to eight variations for each assignable key (that is, a maximum of 8 different chords in Global Key mode, 12 x 8 chords in One Octave mode and 128 x 8 chords in All Keys mode).

The different layers can be triggered by velocity or interval. Proceed as follows to set up your layers:

1. Open the Layers pop-up menu and select Velocity or Interval. Set this to Single Mode if you want to set up only one chord per key.

2. Use the slider below the Layers pop-up menu to specify how many variations (layers) you want to use.

3. Enter the chords as described above.

4. Now you can play the keyboard and trigger the variations according to the selected layer mode.

The layer modes work as follows:

Trigger mode	Description
Velocity	The full velocity range (1–127) is divided into “zones”, according to the number of layers you specified. For example, if you’re using two variations (Number of Layers is set to 2) there will be two velocity “zones”: 1–63 and 64–127. Playing a note with velocity 64 or higher will trigger the second layer, while playing a softer note will trigger the first layer. Using the “Velocity spread” slider at the bottom left of the window, you can change the velocity ranges of the layers so that a different layer will be activated using the same velocity value.
Interval	In this mode, the Chorder will play one chord at a time – you cannot play several different chords simultaneously. When the Interval mode is selected, you press two keys on your keyboard to trigger the desired layer, with the lower key determining the base note for the chord. The layer number will be the difference, i.e. the interval, between the two keys. To select layer 1, press a key one semitone higher than the base note, for layer 2, press a key two semitones higher, and so on.
Single Mode	Select this if you do not wish to use different layers.

Empty layers

If you enter less chords than layers present for a key, these layers will be filled automatically when you end the Learn mode.

This works according to the following rules:

- Empty layers are filled from bottom to top.
- If there are empty layers below the first layer with a chord, these are filled from top to bottom.

An example:

If you have a setup with 8 layers, and you enter the chord C in layer 3 and G7 in layer 7, you get the following result: chord C in layers 1 to 6 and G7 in layers 7 and 8.

Resetting layers

In Learn mode, you can use the “Reset layers” button at the top left of the Chorder window to delete all notes in the different layers for the selected trigger key.

Playstyle

From the Playstyle pop-up menu at the bottom of the pane you can choose one of seven different styles that determine in which order the individual notes of the chords are played back.

The options are as follows:

Playstyle	Description
simultaneous	In this mode all notes are played back simultaneously.
fast up	In this mode a small arpeggio is added, starting with the lowest note.
slow up	Similar to “fast up”, but using a slower arpeggio.
fast down	Similar to “fast up”, but starting with the highest note.
slow down	Similar to “slow up”, but starting with the highest note.
fast random	In this mode the notes are played back in a rapidly changing random order.
slow random	Similar to “fast random”, but the note changes occur more slowly.

Compressor

This MIDI compressor is used for evening out or expanding differences in velocity. Though the result is similar to what you get with the Velocity Compression track parameter, the Compress plug-in presents the controls in a manner more like regular audio compressors. The parameters are:

Parameter	Description
Threshold	Only notes with velocities above this value will be affected by the compression/expansion.
Ratio	This determines the rate of compression applied to the velocity values above the threshold level. Ratios greater than 1:1 result in compression (i.e. less difference in velocity) while ratios lower than 1:1 result in expansion (i.e. greater difference in velocity). What actually happens is that the part of the velocity value that is above the threshold value is divided by the ratio value.
Gain	This adds or subtracts a fixed value from the velocities. Since the maximum range for velocity values is 0–127, you may need to use the Gain setting to compensate, keeping the resulting velocities within the range. Typically, you would use negative Gain settings when expanding and positive Gain settings when compressing.

Context Gate

The Context Gate allows for selective triggering/filtering of MIDI data. It features two modes: in Poly Mode the Context Gate recognizes certain chords that are played and in Mono Mode only certain MIDI notes are let through. These modes can be used for context selective control of MIDI devices and are, for example, very useful in certain live scenarios.

The following parameters are available:

Poly Mode – Polyphony Gate

This allows you to filter MIDI according to the number of pressed keys within a given key range. This can be used independently or in conjunction with the Chord Gate function.

- The Key Range Limit sliders are used to set the key range.
  Only notes within this range will be let through.
- The “Minimum Polyphony” value field allows you to specify the minimum number of notes required to open the gate.

Poly Mode – Chord Gate

When Chord Gate is activated, only notes in recognized chords are let through.

- Two Recognition modes are available: Simple and Normal. In Simple mode, all standard chords (major/minor/b5/dim/sus/maj7 etc.) are recognized, whereas Normal mode takes more tensions into account.

Mono Mode – Channel Gate

When this is activated, only single note events in a specified MIDI channel are let through, which can be used with MIDI controllers that can send MIDI over several channels simultaneously, for example guitar controllers which send data for each string over a separate channel.

- You can set Mono Channel to a specific channel (1–16), or to “Any”, i.e. no channel gating.

Mono Mode – Velocity Gate

This can be used independently or in conjunction with the Channel Gate function. Played notes will sound (no note-off message) until a note is played inside the set range (and additionally the set Channel Gate channel, if checked).

- The Key Range Limit sliders are used to set the key range.

Only notes within this range will be let through.

- Notes below the Minimum Velocity threshold value will be gated.

Auto Gate Time

If there is no input activity, all resounding notes are sent a note-off message after the set time, in seconds or milliseconds.

Panic Reset button

Sends an “All Notes Off” message over all channels, in case of hanging notes.

Learn Reset button

When this is activated, you can specify a Reset trigger event via MIDI. Whenever this specific MIDI event is sent, it triggers an “All Notes Off” message. When you have set the Reset event, the Learn button should be deactivated.

Application examples

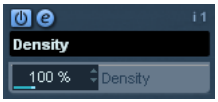
Poly Mode

In Poly mode, you could use the Context Gate to accompany yourself during a live guitar performance using a VST instrument. To do this, you might use a guitar to MIDI converter: You could then program the Context Gate, for example, to allow only those notes to pass the gate that are part of a four-note chord. During your performance you would then play a four-note chord every time that you want to trigger the VST instrument. The instrument will play until the Auto Gate Time is reached and fade out. For more complex performances this can be combined with an arpeggiator, without having to use external pedals to trigger the effect.

Mono Mode

In Mono Mode you could use the Context Gate to trigger variations played with a drum machine/VST instrument. To do this, you will need a guitar to MIDI converter: You could then filter the MIDI channel using the Input Transformer (optional) and program the Context Gate to allow only certain notes on your guitar to pass the gate (e.g. beginning at the 12th band). When you now play one of these notes, the note-off command will not be send out and the corresponding note will sound until the note is played again, a new note is let through, or the Auto Gate Time is reached. This way you can trigger lots of different effects or notes using the high notes on you guitar without having to use an additional MIDI instrument.

Density



This generic control panel affects the “density” of the notes being played from (or thru) the track. When this is set to 100%, the notes are not affected. Lowering the Density setting below 100% will randomly filter out or “mute” notes. Raising the setting above 100% will instead randomly add notes that have been played before.

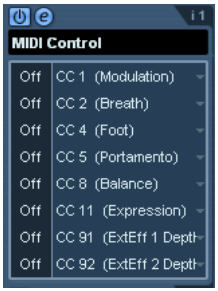
Micro Tuner

The Micro Tuner lets you set up a different microtuning scheme for the instrument, by detuning each key.

- Each Detune slider corresponds to a key in an octave (as indicated by the keyboard display). Adjust a Detune field to raise or lower the tuning of that key, in cents (hundreds of a semitone).
- By keeping the [Alt]/[Option] key pressed, you can adjust all keys by the same amount.

The Micro Tuner comes with a number of presets, including both classical and experimental microtuning scales.

MIDI Control



This generic control panel allows you to select up to eight different MIDI controller types, and use the value fields or sliders (which are displayed when you click on a value field while holding down the [Alt]/[Option] key) to set values for these. A typical use for this would be if you’re using a MIDI instrument with parameters that can be controlled by MIDI controller data (e.g. filter cutoff, resonance, levels, etc.). By selecting the correct MIDI controller types, you can use the plug-in as a control panel for adjusting the sound of the instrument from within Cubase, at any time.

- To select a controller type, use the pop-up menus to the right.
- To deactivate a controller slider, set it to “Off” (drag the slider all the way down).

MIDI Echo

This is an advanced MIDI Echo, which will generate additional echoing notes based on the MIDI notes it receives. It creates effects similar to a digital delay, but also features MIDI pitch shifting and much more. As always it is important to remember that the effect doesn't "echo" the actual audio, but the MIDI notes which will eventually produce the sound in the synthesizer.

The following parameters are available:

Velocity Offset

This parameter allows you to raise or lower the velocity values for each repeat so that the echo fades away or increases in volume (provided that the sound you use is velocity sensitive). For no change of velocity, set this to 0 (middle position).

Pitch Offset

If you set this to a value other than 0, the repeating (echoing) notes will be raised or lowered in pitch, so that each successive note has a higher or lower pitch than the previous. The value is set in semitones.

For example, setting this to -2 will cause the first echo note to have a pitch two semitones lower than the original note, the second echo note two semitones lower than the first echo note, and so on.

Repeats

This is the number of echoes (1 to 12) from each incoming note.

Beat Align

During playback, the Beat Align parameter quantizes the position of the first echo note. You can either set this to "rhythmically exact" values (displayed as note values – see the table below) or activate the PPQ button and choose a PPQ value.

Setting this to "1/8", for example, will cause the first echo note to sound on the first eighth position after the original note.

⇨ The echo time can also be affected by the Delay Decay parameter.

⇨ During live mode, this parameter has no effect since the first echo will always be played together with the note event itself.

Delay

The echoed notes will be repeated as set up with this parameter. You can either set this to "rhythmically exact" values (displayed as note values – see the table below) or activate the PPQ button and choose a PPQ value. This makes it easy to find rhythmically relevant delay values, but still allows experimental settings in between.

Delay Decay

This parameter lets you adjust how the echo time should be changed with each successive repeat. The value is set as a percentage.

- When set to 100% (middle position) the echo time will be the same for all repeats (as set with the Delay parameter).
- If you raise the value above 100%, the echoing notes will play with gradually longer intervals (i.e. the echo will become slower).
- If you lower the value below 100%, the echoing notes will become gradually faster, like the sound of a bouncing ball.

Length

This sets the length of the echoed notes. This can either be identical with the length of the original notes (parameter set to its lowest value) or the length you specify manually. You can either set this to "rhythmically exact" values (displayed as note values – see the table below) or activate the PPQ button and choose a PPQ value.

⇨ The length can also be affected by the Length Decay parameter.

Length Decay

This parameter lets you adjust how the length of the echoed notes should change with each successive repeat. The higher the setting (25–100), the longer the echoed notes will be, compared to their original notes.

About ticks and note values

The timing and position-related parameters (Delay, Length and Pre Delay) can all be set in ticks (or PPQ which denotes the same thing here). There are 480 ticks to each quarter note. While the parameters allow you to step between the rhythmically relevant values (displayed as note values), the following table can also be of help, showing you the most common note values and their corresponding number of ticks:

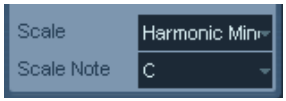
Note Value	Ticks
1/32 note	60
1/16 note triplet	90
1/16 note	120
1/8 note triplet	160
1/8 note	240
Quarter note triplet	320
Quarter note	480
Half note	960

MIDI Modifiers

This plug-in is essentially a duplicate of the MIDI Modifiers section in the Inspector. This can be useful, for example, if you need extra Random or Range settings.

The MIDI Modifiers effect also includes an additional function that isn't available among the track parameters:

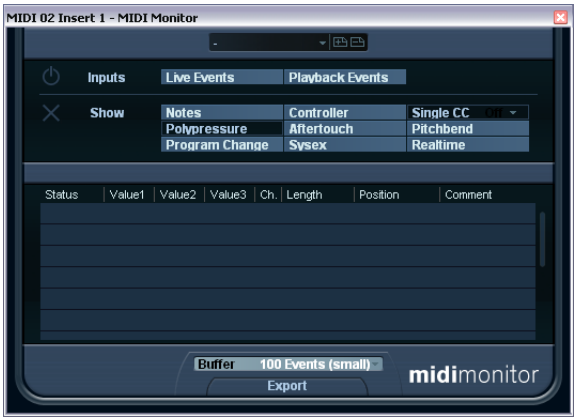
Scale Transpose



This allows you to transpose each incoming MIDI note, so that it fits within a selected musical scale. The scale is specified by selecting a key (C, C#, D, etc.) and a scale type (major, melodic or harmonic minor, blues, etc.).

⇨ To turn Scale Transpose off, select "No Scale" from the Scale pop-up menu.

MIDI Monitor



The MIDI Monitor is used used to monitor incoming MIDI events. You can choose whether to analyze live or playback events and which types of MIDI data are to be monitored. Use this, for example, to analyze which MIDI events are being generated by a MIDI track, or to find “suspicious” events, such as notes with velocity 0 that certain MIDI devices might fail to interpreted as note-off events.

Inputs section

In this section you can choose whether to monitor Live Events or Playback Events.

Show section

Here, you can activate/deactivate the different types of MIDI events, e.g. notes or program change events. If you choose the Controller option you can also define which type of controller to monitor.

Data table

In the table in the lower section of the window, you will see detailed information about the monitored MIDI events.

Buffer pop-up menu

In the Buffer pop-up menu you can set the buffer size to 100, 1000 or 10000 events. This is the maximum number of events that is kept in the list of monitored events. Once this list is full, the oldest entries will be deleted when new events are received.

⇨ The larger the buffer, the more processing resources are required. To avoid a negative impact on your system’s performance, make sure to use the smallest possible buffer size.

Export function

Click the Export button to export the monitoring data as a simple text file.

Record events button

Use this button to the left of the Inputs section to start or stop the monitoring of MIDI events.

Clear list button

The Clear List button to the left of the Show section allows you to clear the table of recorded MIDI events.

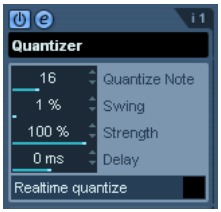
Note to CC

This effect will generate a MIDI continuous controller event for each incoming MIDI note. The value of the controller event corresponds to the velocity of the MIDI note, which is then used to control the selected MIDI controller (by default CC 7, Main Volume). For each note end, another controller event with the value 0 is sent. The incoming MIDI notes pass through the effect unaffected.

The purpose of this plug-in is to generate a gate effect. This means that the notes played are used to control something else. For example, if Main Volume (CC 7) is selected, notes with low velocity will lower the volume in the MIDI instrument, while notes with a high velocity will raise the volume.

⚠ Note that a controller event is sent out each time a new note is played. If high and low notes are played simultaneously, this may lead to confusing results. Therefore, the Note to CC effect is best applied to monophonic tracks (playing one note at a time).

Quantizer



Quantizing is a function that changes the timing of notes by moving them towards a “quantize grid”. This grid may consist of e.g. straight sixteenth notes (in which case the notes would all get perfect sixteenth note timing), but could also be more loosely related to straight note value positions (applying a “swing feel” to the timing, etc.).

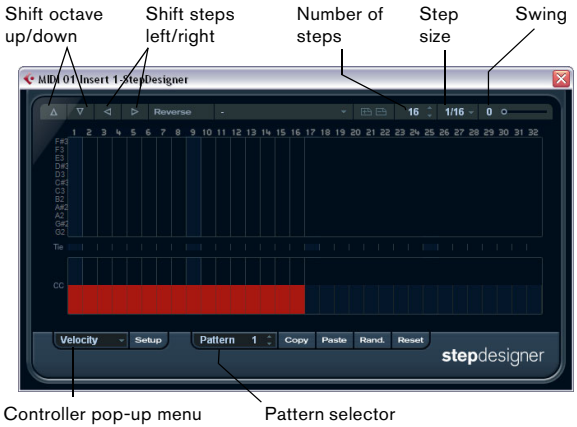
⇨ The main Quantize function in Cubase is described in the Operation Manual.

While the Quantize function on the MIDI menu applies the timing change to the actual notes on a track, the Quantizer effect allows you to apply quantizing “on the fly”, changing the timing of the notes in real time. This makes it easier to try out different settings when creating grooves and rhythms. Note however, that the main Quantize function contains settings and features that are not available in the Quantizer.

The Quantizer has the following parameters:

Parameter	Description
Quantize Note	This sets the note value on which the quantize grid is based. Straight notes, triplets and dotted notes are available. For example, “16” means straight sixteenth notes and “8T” means eighth note triplets.
Swing	This allows you to offset every second position in the grid, creating a swing or shuffle feel. The value is a percentage – the higher you set this, the farther to the right every even grid position is moved.
Strength	This determines how close the notes should be moved to the quantize grid. When set to 100%, all notes will be forced to the closest grid position; lowering the setting will gradually loosen the timing.
Delay	This delays (positive values) or advances (negative values) the notes in milliseconds. Unlike the Delay setting in the Track Parameters, this delay can be automated.
Realtime quantize	During live mode this option can be used to change the timing of the notes played so that they fit the quantize grid.

Step Designer



The Step Designer is a MIDI pattern sequencer that sends out MIDI notes and additional controller data according to the pattern you set up. It does not make use of the incoming MIDI, other than automation data (such as recorded pattern changes).

Creating a basic pattern

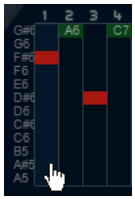
1. Use the Pattern selector to choose which pattern to create.
Each Step Designer can hold up to 200 different patterns.

2. Use the “Step size” setting to specify the “resolution” of the pattern.
In other words, this setting determines how long each step is. For example, if this is set to “1/16” each step will be a sixteenth note.

3. Specify the number of steps in the pattern with the “Number of steps” setting.
As you can see in the note display, the maximum number of steps is 32. For example, setting “Step size” to 16 and “Number of steps” to 32 would create a two bar pattern with sixteenth note steps.

4. Click in the note display to insert notes.
You can insert notes on any of the 32 steps, but the Step Designer will only play back the number of steps set with the Step size parameter.

- The display spans one octave (as indicated by the pitch list to the left). You can scroll the displayed octave up or down by clicking in the pitch list and dragging up or down.
This way you can insert notes at any pitch. Note that each step can contain one note only – the Step Designer is monophonic.

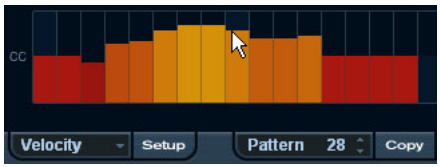


Click and drag to view other octaves.

- To remove a note from the pattern, click on it again.

5. On the Controller pop-up menu, select Velocity.
This pop-up menu determines what is shown in the lower controller display.

6. Adjust the velocity of the notes by dragging the velocity bars in the controller display.



7. To make notes shorter, select “Gate” on the Controller pop-up menu and lower the bars in the controller display.
When a bar is set to its maximum value (fully up), the corresponding note will be the full length of the step (as set with the Step size parameter).

8. To make notes longer, you can tie two notes together. This is done by inserting two notes and clicking in the Tie column for the second note.
When two notes are tied, the second note will not be triggered – the previous note is lengthened instead. Also, the tied (second) note will automatically get the same pitch as the first note. You can add more notes and tie them in the same way, creating longer notes.

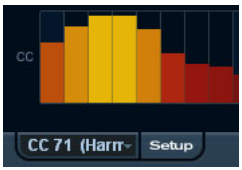
9. If you now start playback in Cubase, the pattern will play as well, sending out MIDI notes on the track’s MIDI output and channel (or, if you have activated the Step Designer as a send effect, on the MIDI output and channel selected for the send in the Inspector).

Adding controller curves

The Controller pop-up menu has two more items: two controller types.

- You can select which two controller types (filter cutoff, resonance, volume, etc.) should be available on the pop-up menu by clicking the Setup button and selecting controllers from the lists that appears.
This selection is global, i.e. it applies to all patterns.

- To insert controller information in a pattern, select the desired controller from the pop-up menu and click in the controller display to draw events.
The MIDI controller events will be sent out during playback along with the notes.



⇨ If you drag a controller event bar all the way down, no controller value is sent out on that step.

Other pattern functions

The following functions make it easier to edit, manipulate and manage patterns:

Function	Description
Shift Octave up/down	These buttons allow you to shift the entire pattern up or down in octave steps.
Shift Steps left/right	Moves the pattern one step to the left or right.
Reverse	Reverses the pattern, so that it plays backwards.
Copy/Paste	Allows you to copy the current pattern and paste it in another pattern location (in the same Step Designer instance or another).
Reset	Clears the pattern, removing all notes and setting controller values to default.
Randomize	Generates a completely random pattern – useful for experimenting.
Swing	The Swing parameter allows you to offset every second step, creating a swing or shuffle feel. The value is a percentage – the higher you set this, the farther to the right every even step is moved.
Presets	Handling of Presets is described in the chapter “MIDI realtime settings” in the Operation manual. Note that a stored Preset contains all 200 patterns in the Step Designer.

Automating pattern changes

You can create up to 200 different patterns in each Step Designer – just select a new pattern and add notes and controllers as described above.

Typically, you want the pattern selection to change during the project. You can accomplish this by automating the Pattern selector, either in real time by activating the Write automation and switching patterns during playback or by drawing in the automation subtrack for the Step Designer’s MIDI track. Note that you can also press a key on your MIDI keyboard to change patterns. For this, you have to set up the Step Designer as an insert effect for a record enabled MIDI track. Press C1 to select pattern 1, C#1 to select pattern 2, D1 to select pattern 3, D#1 to select pattern 4 and so on. If you want, you can record these pattern changes as note events on a MIDI track. Proceed as follows:

1. Select the desired MIDI track or create a new one and activate the Step Designer as an insert effect.

2. Set up several patterns as described above.

3. Press the Record button and press the desired keys on your keyboard to select the corresponding patterns.
The pattern changes will be recorded on the MIDI track.

4. Stop recording and play back the MIDI track.
You will now hear the recorded pattern changes.

⇨ This will only work for the first 92 patterns.

Track Control

The Track Control effect contains three ready-made control panels for adjusting parameters on a GS or XG compatible MIDI device. The Roland GS and Yamaha XG protocols are extensions of the General MIDI standard, allowing for more sounds and better control of various instrument settings. If your instrument is compatible with GS or XG, the Track Controls effect allows you to adjust sounds and effects in your instrument from within Cubase.

Selecting a control panel

At the top of the Track Controls effect window you will find a pop-up menu. This is where you select which of the available control panels to use:

Control panel	Description
GS 1	Effect sends and various sound control parameters for use with instruments compatible with the Roland GS standard.
XG 1	Effect Sends and various sound control parameters for use with instruments compatible with the Yamaha XG standard.
XG 2	Global settings (affecting all channels) for instruments compatible with the Yamaha XG standard.

About the Reset and Off buttons

Regardless of the selected mode, you will find two buttons labelled “Off” and “Reset” at the top of the control panel:

- Clicking the Off button will set all controls to their lowest value, without sending out any MIDI messages.
- Clicking the Reset button will set all parameters to their default values, and send out the corresponding MIDI messages.

For most parameters, the default values will be zero or “no adjustment”, but there are exceptions to this. For example, the default “Send 1” setting is 64.

GS 1

The following controls are available when the GS 1 Controls mode is selected:

Control	Description
Send 1	Send level for the reverb effect.
Send 2	Send level for the chorus effect.
Send 3	Send level for the “variation” effect.
Attack	Adjusts the attack time of the sound. Lowering the value shortens the attack, while raising it gives a slower attack. Middle position (64) means no adjustment is made.
Decay	Adjusts the decay time of the sound. Lowering the value shortens the decay, while raising it makes the decay longer.
Release	Adjusts the release time of the sound. Lowering the value shortens the release, while raising it makes the release time longer.
Cutoff	Adjusts the filter cutoff frequency.
Resonance	Adjusts the filter resonance.
Express	Allows you to send out expression pedal messages on the track’s MIDI channel.
Ch. Press.	Allows you to send out aftertouch (channel pressure) messages on the track’s MIDI channel. This is useful if your keyboard cannot send aftertouch, but you have sound modules that respond to aftertouch. The default value for this parameter is zero.
Breath	Allows you to send breath control messages on the track’s MIDI channel.
Modul.	Allows you to send modulation messages on the track’s MIDI channel (just as you normally do with a modulation wheel on a MIDI keyboard).

XG 1

The following controls are available when the XG 1 mode is selected:

Control	Description
Send 1	Send level for the reverb effect.
Send 2	Send level for the chorus effect.
Send 3	Send level for the “variation” effect.
Attack	Adjusts the attack time of the sound. Lowering this value shortens the attack, while raising it gives a slower attack. Middle position means no adjustment is made.
Release	Adjusts the release time of the sound. Lowering this value shortens the release, while raising it makes the release time longer. Middle position means no adjustment is made.
Harm.Cont	Adjusts the harmonic content of the sound.
Bright	Adjusts the brightness of the sound.
CutOff	Adjusts the filter cutoff frequency.
Resonance	Adjusts the filter resonance.

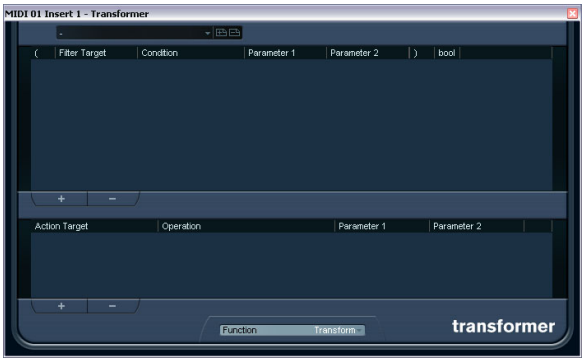
XG 2

In this mode, the parameters affect global settings in the instrument(s). Changing one of these settings for a track will in fact affect all MIDI instruments connected to the same MIDI output, regardless of the MIDI channel setting of the track. Therefore, to avoid confusion it might be a good idea to create an empty track and use this only for these global settings.

The following controls are available:

Control	Description
Eff. 1	This allows you to select which type of reverb effect should be used: No effect (the reverb turned off), Hall 1–2, Room 1–3, Stage 1–2 or Plate.
Eff. 2	This allows you to select which type of chorus effect should be used: No effect (the chorus turned off), Chorus 1–3, Celeste 1–3 or Flanger 1–2.
Eff. 3	This allows you to select one of a large number of “variation” effect types. Selecting “No Effect” is the same as turning off the variation effect.
Reset	Sends an XG reset message.
MastVol	This is used to control the Master Volume of an instrument. Normally you should leave this in its highest position and set the volumes individually for each channel (with the volume faders in the Cubase mixer or in the Inspector).

Transformer



The Transformer is a real-time version of the Logical Editor. With this you can perform very powerful MIDI processing on the fly, without affecting the actual MIDI events on the track.

The Logical Editor is described in the corresponding chapter in the Operation Manual. As the parameters and functions are almost identical, the descriptions for the Logical Editor also apply to the Transformer. Where there are differences between the two, this is clearly stated.

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