Plug-in Reference

Cristina Bachmann, Heiko Bischoff, Marion Bröer, Christina Kaboth, Insa Mingers, Sabine Pfeifer, Benjamin Schütte

This document covers the plug-in effects and instruments included in Cubase 7, Cubase Artist 7, Cubase Elements 7, Cubase AI 7, Cubase LE 7, and Nuendo 6.

This PDF provides improved access for vision-impaired users. Please note that due to the complexity and number of images in this document, it is not possible to include text descriptions of images.

Release Date: December 3, 2012

Table of Contents

The Included Effect Plug-ins

Introduction

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

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.

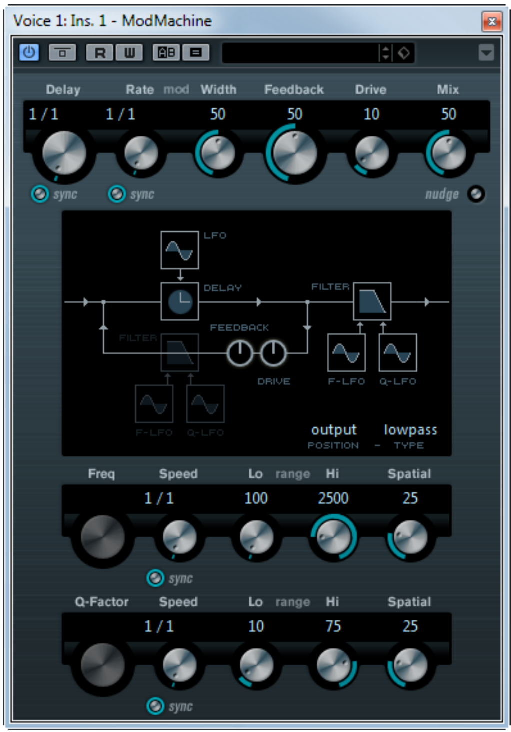
⇨ Most of the included effects are compatible with VST 3. For more information, see the Operation Manual.

Delay Plug-ins

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

ModMachine

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–



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 following parameters are available:

Parameter	Description
Delay	If tempo sync is on, this is where you specify the base note value for the delay (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the delay time can be set freely in milliseconds.
Delay – Sync button	The button below the Delay knob switches tempo sync for the Delay parameter on or off.
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.
Rate – Sync button	The button below the Rate knob switches tempo sync for the Rate parameter on or off.
Width	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	Sets the number of repeats for the delay.
Drive	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 and the wet signal. If ModMachine is used as a send effect, set this to the maximum value (100%) as you can control the dry/effect balance with the send.
Nudge button	Clicking the Nudge button once momentarily speeds up the audio coming into the plug-in, simulating an analog tape nudge type sound effect.
Signal path graphic and Filter position	The filter can either be placed in the feedback loop of the delay or in the output path of the effect (after the Drive and Feedback parameters). To switch between the “loop” and “output” positions, click on the Filter section displayed in the graphic or click on the Position field at the bottom right of the graphic.
Filter type (in graphic display)	Allows you to select a filter type. A low-pass, band-pass, and high-pass filter are available.
Freq	Sets the cutoff frequency for the filter. It is only available if tempo sync for the Speed parameter is deactivated and the parameter is set to 0.
Speed	Sets the speed of the filter frequency LFO modulation. When using tempo sync, 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 speed can be set freely.
Speed – Sync button	The button below the Speed knob switches tempo sync for the Speed parameter on or off.
Range Lo/Hi	These knobs specify the range of the filter frequency modulation. Both positive (for example, Lo set to 50 and Hi set to 10000) and negative (for example, 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 controlled by the Freq parameter instead.

Parameter	Description
Spatial	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	Sets the resonance of the filter. It is only available if filter resonance LFO tempo sync is deactivated and the Speed parameter is set to 0. When using tempo sync, the resonance is controlled by the Speed and Range parameters.
Speed	Sets the speed of the filter resonance LFO modulation. When using tempo sync, 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 speed can be set freely.
Speed – Sync button	The button below the Speed knob switches tempo sync for the Speed parameter on or off.
Range Lo/Hi	These knobs specify the range of the filter resonance modulation. Both positive (for example, Lo set to 50 and Hi set to 100) and negative (for example, 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	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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

This is a mono delay effect that can either be tempo-based or use freely specified delay time settings.

The following parameters are available:

Parameter	Description
Delay	If tempo sync is on, this is where you specify the base note value for the delay (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the delay time can be set freely in milliseconds.
Sync button	The button below the Delay knob switches tempo sync on or off.
Feedback	Sets the number of repeats for the delay.

Parameter	Description
Filter Lo	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	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.
Mix	Sets the level balance between the dry and the wet signal. If MonoDelay is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

⇨ If side-chaining is supported, the delay can also be controlled from another signal source via 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 Operation Manual.

PingPongDelay

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

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 following parameters are available:

Parameter	Description
Delay	If tempo sync is on, this is where you specify the base note value for the delay (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the delay time can be set freely in milliseconds.
Sync button	The button below the Delay knob switches tempo sync on or off.
Feedback	Sets the number of repeats for the delay.
Filter Lo	Affects the feedback loop of the effect signal and allows you to roll off low frequencies up to 800 Hz. The button below the knob activates/deactivates the filter.
Filter Hi	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
Spatial	Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo ping-pong effect.
Mix	Sets the level balance between the dry and the wet signal. If PingPongDelay is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

⇨ If side-chaining is supported, the delay can also be controlled from another signal source via 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 Operation Manual.

StereoDelay

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

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

The following parameters are available:

Parameter	Description
Delay 1 & 2	If tempo sync is on, this is where you specify the base note value for the delay (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the delay time can be set freely in milliseconds.
Sync buttons	The buttons below the Delay knobs turn tempo sync on or off for the corresponding delay.
Feedback 1 & 2	Set the number of repeats for each delay.
Filter Lo 1 & 2	Affect the feedback loop of the effect signal and allow you to roll off low frequencies up to 800 Hz. The buttons below the knobs activate/deactivate the filter.
Filter Hi 1 & 2	Affect the feedback loop of the effect signal and allow you to roll off high frequencies from 20 kHz down to 1.2 kHz. The buttons below the knobs activate/deactivate the filter.
Pan 1 & 2	Set the stereo position for each delay.
Mix 1 & 2	Set the level balance between the dry and the wet signal. If StereoDelay is used as a send effect, set these controls to the maximum value (100%) as you can control the dry/effect balance with the send.

⇨ If side-chaining is supported, the delay can also be controlled from another signal source via 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 Operation Manual.

Distortion Plug-ins

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

AmpSimulator

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–

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 following parameters are available:

Parameter	Description
Amplifier pop-up menu	This pop-up menu is opened by clicking on the amplifier name shown at the top of the amp section. It allows you to select an amplifier model. The amp section can be bypassed by selecting “No Amp”.
Drive	Controls 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	Boosts or dampens the higher frequencies.
Volume	Controls the overall output level.

Parameter	Description
Cabinet pop-up menu	This pop-up menu is opened by clicking on the cabinet name shown at the top of the cabinet section. It allows you to select a speaker cabinet model. 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.

BitCrusher

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–

If you are 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. For example, you can make a 24-bit audio signal sound like an 8 or 4-bit signal, or even render it completely garbled and unrecognizable.

The following parameters are available:

Parameter	Description
Mode	Allows you to select one of the four operating modes of BitCrusher. In each mode the plug-in sounds differently. Modes I and III are nastier and noisier, while modes II and IV are more subtle.
Sample Divider	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 is eliminated, turning the signal into unrecognizable noise.
Depth	Defines the bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 creates mostly noise.
Output slider	Governs the output level from BitCrusher. Drag the slider upwards to increase the level.
Mix slider	Regulates the balance between the output from BitCrusher and the original audio signal. Drag the slider upwards for a more dominant effect, and downwards if you want the original signal to be more prominent.

DaTube

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–

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

The following parameters are available:

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	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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–

Distortion adds crunch to your tracks.

The following parameters are available:

Parameter	Description
Boost	Increases the distortion amount.
Feedback	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.

Parameter	Description
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.

Grungelizer

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–

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 following parameters are available:

Parameter	Description
Crackle	Adds crackle to create that old vinyl record sound. The farther to the right you turn the knob, the more crackle is added.
RPM switch	When emulating the sound of a vinyl record, this switch lets you set the speed of the record in RPM (revolutions per minute).
Noise	Regulates the amount of static noise added.
Distort	Adds distortion.
EQ	Turn this knob to the right to cut off the low frequencies, and create a more hollow, lo-fi sound.
AC	Emulates a constant, low hum of AC current.
Frequency switch	Sets the frequency of the AC current (50 or 60 Hz), and thus the pitch of the AC hum.
Timeline	Regulates the amount of overall effect. The farther to the right (1900) you turn the knob, the more noticeable the effect.

SoftClipper

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

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

The following parameters are available:

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	Adjusts the amount of the second harmonic in the processed signal.
Third	Adjusts the amount of the third harmonic in the processed signal.

VST Amp Rack

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–

The VST Amp Rack is a powerful guitar amp simulator. It offers a choice of amplifiers and speaker cabinets that can be combined with stomp box effects.

At the top of the plug-in panel there are six buttons, arranged according to the position of the corresponding elements in the signal chain. These buttons open different pages in the Display section of the plug-in panel: Pre-Effects, Amplifiers, Cabinets, Post-Effects, Microphone Position, and Master.

Below the Display section, the selected amplifier is shown. The color and texture of the area below the amplifier indicate the selected cabinet.

Pre/Post-Effects

On the Pre-Effects and the Post-Effects pages, you can select up to six common guitar effects. On both pages the same effects are available, the only difference being the position in the signal chain (before and after the amplifier). On each page, every effect can be used once.

Each effect features an On/Off button known from stompbox effects, as well as individual parameters. The following effects and parameters are available:

Effect	Option	Description
Wah Wah	Pedal	Controls the filter frequency sweep.
Volume	Pedal	Controls the level of the signal passing through the effect.
Compressor	Intensity	Changes the intensity of the compressor effect.

Effect	Option	Description
Limiter	Threshold	Determines the maximum output level. Signal levels above the set threshold are cut off.
	Release	Sets the time after which the gain returns to the original level.
Maximizer	Amount	Determines the loudness of the signal.
Chorus	Rate	Allows you to set the sweep rate. This parameter can be synchronized to the project tempo, see “Sync Mode” on page 18.
	Width	Determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Phaser	Rate	Allows you to set the sweep rate. This parameter can be synchronized to the project tempo, see “Sync Mode” on page 18.
	Width	Determines the width of the modulation effect between higher and lower frequencies.
Flanger	Rate	Allows you to set the sweep rate. This parameter can be synchronized to the project tempo, see “Sync Mode” on page 18.
	Feedback	Determines the character of the flanger effect. Higher settings produce a more metallic sounding sweep.
	Mix	Sets the level balance between the dry and the wet signal.
Tremolo	Rate	Allows you to set the modulation speed. This parameter can be synchronized to the project tempo, see “Sync Mode” on page 18.
	Depth	Governs the depth of the amplitude modulation.
Octaver	Direct	Adjusts the mix of the original signal and the generated voices. 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	Adjusts the level of the signal that is generated one octave below the original pitch. A setting of 0 means that the voice is muted.
	Octave 2	Adjusts the level of the signal that is generated two octaves below the original pitch. A setting of 0 means that the voice is muted.
Delay	Delay	Sets the delay time in milliseconds. This parameter can be synchronized to the project tempo, see “Sync Mode” on page 18.
	Feedback	Sets the number of repeats for the delay.
	Mix	Sets the level balance between the dry and the wet signal.

Effect	Option	Description
Tape Delay	Delay	Tape Delay creates a delay effect known from tape machines. The Delay parameter sets the delay time in milliseconds. This parameter can be synchronized to the project tempo, see "Sync Mode" on page 18.
	Feedback	Sets the number of repeats for the delay.
	Mix	Sets the level balance between the dry and the wet signal.
Tape Ducking Delay	Delay	Tape Ducking Delay creates a delay effect known from tape machines with a ducking parameter. The Delay parameter sets the delay time in milliseconds. This parameter can be synchronized to the project tempo, see "Sync Mode" on page 18.
	Feedback	Sets the number of repeats for the delay.
	Duck	Works like an automatic mix parameter. If the level of the input signal is high, the portion of the effect signal is lowered, or ducked (low internal mix value). If the level of the input signal is low, the portion of the effect signal is raised (high internal mix value). This way the delayed guitar signal stays rather dry during loud or intensely played passages.
Overdrive	Drive	Overdrive creates a tube-like overdrive effect. The higher the Drive value, the more harmonics are being added to the output signal of this effect.
	Tone	Works as a filter effect on the added harmonics.
	Level	Adjusts the output level.
Fuzz	Boost	Fuzz creates a rather harsh distortion effect. The higher the Boost value, the more distortion is being created.
	Tone	Works as a filter effect on the added harmonics.
	Level	Adjusts the output level.
Gate	Threshold	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 close the gate.
	Release	Sets the time after which the gate closes.
Equalizer	Low	Changes the level of the low-frequency portion of the incoming signal.
	Middle	Changes the level of the mid-frequency portion of the incoming signal.
	High	Changes the level of the high-frequency portion of the incoming signal.
Reverb	Type	A convolution-based reverb effect. The Type parameter allows you to switch between different reverb types (Studio, Hall, Plate, and Room).
	Mix	Sets the level balance between the dry and the wet signal.

Sync Mode

For some controls, the sync mode can be activated to synchronize the corresponding parameter with the tempo of the host application. These plug-in parameters are then used to specify the base note value for tempo syncing (1/1 to 1/32, straight, triplet, or dotted).

The names of these parameters are underlined. Click a knob to activate or deactivate tempo sync. An LED at the top right of the knob indicates that Sync mode is active. You can then select a base note value for tempo syncing from the pop-up menu above the control.

Using Effects

- To insert a new effect, click the plus button that appears when you point the mouse at an empty plug-in slot or at one of the arrows before or after a used effect slot.
- To remove an effect from an effect slot, click the effect name and select “None” from the pop-up menu.
- To change the order of the effects in the chain, click on an effect and drag it to another position.
- To activate or deactivate an effect, click the pedal-like button below the effect name.
  When an effect is active, the LED next to the button is lit.

⚠ Pre-effects and post-effects can be mono or stereo, depending on the track configuration.

⇨ Using quick controls you can conveniently set up an external MIDI device such as a foot controller to control the VST Amp Rack effects. For more information about quick controls, see the Operation Manual.

Amplifiers

The amps available on the Amplifiers page were modeled on real-life amplifiers. Each amp features settings typical for guitar recording, such as gain, equalizers, and master volume. The sound-related parameters Bass, Middle, Treble, and Presence have a significant impact on the overall character and sound of the corresponding amp.

The following amp models are available:

- Plexi – Classic British rock tone; extremely transparent sound, very responsive.
- Plexi Lead – British rock tone of the 70’s and 80’s.
- Diamond – The cutting edge hard rock and metal sounds of the 90’s.
- Blackface – Classic American clean tone.
- Tweed – Clean and crunchy tones; originally developed as a bass amp.
- Deluxe – American crunch sound coming from a rather small amp with a big tone.
- British Custom – Produces the sparkling clean or harmonically distorted rhythm sounds of the 60’s.

The different amps keep their settings when you switch models. However, if you want to use the same settings after reloading the plug-in, you need to set up a preset.

Using Amplifiers

- To switch amps on the Amplifiers page, click the model that you want to use.
  Select “No Amplifier” if you only want to use the cabinets and effects.

Cabinets

The cabinets available on the Cabinets page simulate real-life combo boxes or speakers. For each amp, a corresponding cabinet type is available. However, you can combine amps and cabinets at will.

Using Cabinets

- To switch cabinets on the Cabinets page, click the model that you want to use. Select “No Cabinet” if you only want to use the amps and effects.
- If you select “Link Amplifier & Cabinet Choice”, the plug-in automatically selects the cabinet corresponding to the selected amp model.

Microphone Position

On the Microphone Position page, you can choose between 7 positions to place the microphone. These positions result from two different angles (center and edge) and three different distances from the speaker, as well as an additional center position at an even greater distance from the speaker.

You can choose between two microphone types: a large-diaphragm condenser microphone and a dynamic microphone. Crossfading between the characteristics of the two microphones is also possible.

Placing the Microphone

- To select a microphone position, click the corresponding ball in the graphic.
  The selected position is marked in red.
- To select one of the microphone types or blend between the two types, turn the Mix control between the two microphones.

Master

Use the Master page to fine-tune the sound.

Input/Output Level Meters

The input and output level meters on the left and the right of the Master section show the signal level of your audio. The rectangle on the input meter indicates the optimum incoming level range. In compact view, the input and output levels are indicated by two LEDs at the top left and right.

Using the Master Controls

- To activate/deactivate the Equalizer, click the pedal-like On/Off button.
  When the Equalizer is active, the LED next to the button is lit.
- To activate/deactivate an equalizer band, click the corresponding Gain knob.
  When a band is active, the LED to the left of the Gain knob is lit.
- To tune your guitar strings, click the pedal-like On/Off button to activate the Tuner and play a string.
  When the correct pitch is displayed and the row of LEDs below the digital display is green, the string is tuned correctly. The more red LEDs on the left/right are lit, the lower/higher the pitch.
- To mute the output signal of the plug-in, click the pedal-like Master button.
  When the LED is off, the output is muted. Use this to tune your guitar in silence, for example.
- To change the volume of the output signal, use the Level control in the Master section.

- To process the pre-effects, the amplifier, and the cabinets in full stereo mode, make sure that the plug-in is inserted on a stereo track, and activate the Stereo button.

View Settings

Two different views for the VST Amp Rack plug-in panel are available: the default view and a compact view, which takes up less screen space.

In the default view, you can use the top buttons to open the corresponding page in the Display section above the amp controls. You can horizontally resize the plug-in panel by clicking and dragging the edges or corners.

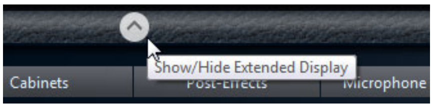
In the compact view the page display is hidden from view. You can still change the amp settings and switch amps or cabinets using the mouse wheel.

Using the Smart Controls

Smart controls become visible on the plug-in frame when the mouse pointer is positioned on the plug-in panel.

Switching between Default and Compact View

- To toggle between the different views, click the down/up arrow button (Show/Hide Extended Display) at the top center of the plug-in frame.

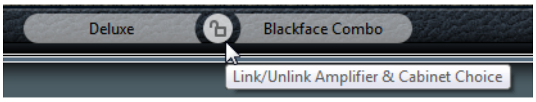


Changing the Amplifier and Cabinet Selection in the Compact View

In the compact view, a smart control on the lower border of the plug-in frame allows you to select different amplifier and cabinet models.

- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the “Link/Unlink Amplifier & Cabinet Choice” button.

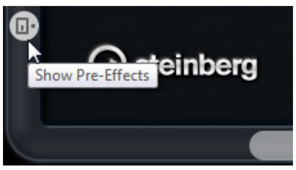
  If you now select another amp model, the cabinet selection follows. However, if you select a different cabinet model, the lock is deactivated.



Previewing Effect Settings

In both views, you can show a preview of the pre- and post-effects that you selected on the corresponding pages:

- Click and hold the “Show Pre-Effects” or “Show Post-Effects” button at the bottom left or right of the plug-in frame.



Dynamics Plug-ins

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

Brickwall Limiter

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

Brickwall Limiter ensures that the output level never exceeds a set limit. Due to its fast attack time, Brickwall Limiter can reduce even short audio level peaks without creating audible artifacts. However, this plug-in creates a latency of 1 ms. Brickwall Limiter features separate meters for input, output, and the amount of limiting. Position this plug-in at the end of the signal chain, before dithering.

The following parameters are available:

Parameter	Description
Threshold (-20 to 0 dB)	Only signal levels above the set threshold are processed.
Release (10 to 1000 ms or Auto mode)	Sets the time after which the gain returns to the original level when the signal drops below the threshold. If the Auto button is activated, Brickwall Limiter automatically finds the optimal release setting, depending on the audio material.
Link button	If this button is activated, Brickwall Limiter uses the channel with the highest level to analyze the input signal. If the Link button is deactivated, each channel is analyzed separately.
Detect Intersample Clipping	In this mode, Brickwall Limiter detects and limits signal levels between two samples to prevent distortion when converting digital signals to analog.

⇨ Brickwall Limiter is designed for the reduction of occasional peaks in the signal. If the Gain Reduction meter indicates constant limiting, try raising the threshold or lowering the overall level of the input signal.

Compressor

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

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 following parameters are available:

Parameter	Description
Threshold (-60 to 0dB)	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)	Determines the amount of gain reduction applied to signals above the set threshold. A ratio of 3:1 means that for every 3dB the input level increases, the output level increases by only 1dB.
Soft Knee button	If this button is off, signals above the threshold are compressed instantly according to the set ratio (hard knee). When Soft Knee is activated, the onset of compression is more gradual, producing a less drastic result.
Make-up (0 to 24dB or Auto mode)	Compensates for output gain loss, caused by compression. If the Auto button is activated, the knob becomes dark and the output is automatically adjusted for gain loss.
Attack (0.1 to 100ms)	Determines how fast Compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.
Hold (0 to 5000ms)	Sets the time the applied compression affects the signal after exceeding the threshold. Short hold times are useful for DJ-style ducking, while longer hold times are required for music ducking, for example, when working on a documentary film.

Parameter	Description
Release (10 to 1000 ms or Auto mode)	Sets the time after which the gain returns to the original level when the signal drops below the threshold. If the Auto button is activated, Compressor automatically finds an optimal release setting that varies depending on the audio material.
Analysis (0 to 100) (Pure Peak to Pure RMS)	Determines whether the input signal is analyzed 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 button	When this button is activated, the look-ahead feature of Compressor is disengaged. Look ahead produces more accurate processing, but adds 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.

⇨ If side-chaining is supported, the compression can also be controlled from another signal source via 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 Operation Manual.

DeEsser

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

A de-esser reduces 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.

The following parameters are available:

Parameter	Description
Reduction	Controls the intensity of the de-essing effect.
Threshold	When the Auto option is deactivated, you can use this control to set a threshold for the incoming signal level, above which the plug-in starts to reduce the sibilants.

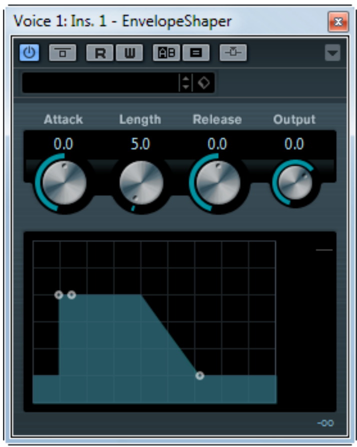
Parameter	Description
Auto	Automatically and continually chooses an optimum threshold setting independent of the input signal. The Auto option does not work for low-level signals (< -30dB peak level). To reduce the sibilants in such a file, set the threshold manually.
Release	Sets the time after which the de-essing effect returns to zero when the signal drops below the threshold.
Level meters	Indicate the dB values of the input (IN) and output (OUT) signals as well as the value by which the level of the sibilant (or s-frequency) is reduced (GR). The gain reduction meter shows values between 0dB (no reduction) and -20dB (the s-frequency level is lowered by 20dB).

Positioning the DeEsser in the Signal Chain

When recording a voice, the de-esser's position in the signal chain is usually located after the microphone pre-amp and before a compressor/limiter. This keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics.

EnvelopeShaper

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–
Side-chain support	–	–	–	X	X	X	–



EnvelopeShaper can be used to attenuate or boost the gain of the attack and release phase of audio material. You can either use the knobs or drag the breakpoints in the graphical display to change parameter values. Be careful with levels when boosting the gain and if needed reduce the output level to avoid clipping.

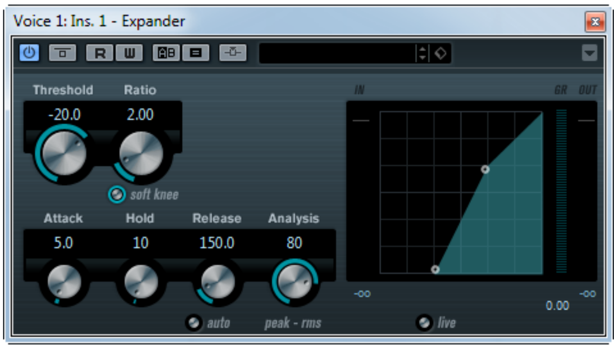
The following parameters are available:

Parameter	Description
Attack (-20 to 20dB)	Changes the gain of the attack phase of the signal.
Length (5 to 200ms)	Determines the length of the attack phase.
Release (-20 to 20dB)	Changes the gain of the release phase of the signal.
Output (-24 to 12dB)	Sets the output level.

⇨ If side-chaining is supported, the effect can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the effect is triggered. For a description of how to set up side-chain routing, see the Operation Manual.

Expander

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–
Side-chain support	–	–	–	–	X	X	–



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 graphical display to change the Threshold and the Ratio parameter values.

The following parameters are available:

Parameter	Description
Threshold (-60 to 0 dB)	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 to 8:1)	Determines the amount of gain boost applied to signals below the set threshold.
Soft Knee button	If this button is off, signals below the threshold are expanded instantly according to the set ratio (hard knee). When “Soft Knee” is activated, the onset of expansion is more gradual, producing a less drastic result.
Attack (0.1 to 100 ms)	Determines how fast Expander responds to signals below the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.
Hold (0 to 2000 ms)	Sets the time the applied expansion affects the signal below the threshold.
Release (10 to 1000 ms or Auto mode)	Sets the time after which the gain returns to the original level when the signal exceeds the threshold. If the Auto button is activated, Expander automatically finds an optimal release setting that varies depending on the audio material.

Parameter	Description
Analysis (0 to 100) (Pure Peak to Pure RMS)	Determines whether the input signal is analyzed 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 button	When this button is activated, the look-ahead feature of Expander is disengaged. Look ahead produces more accurate processing, but adds 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.

⇨ If side-chaining is supported, the expansion can also be controlled from another signal source via 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 Operation Manual.

Gate

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

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

The following parameters are available:

Parameter	Description
Threshold (-60 to 0dB)	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 close the gate.
State LED	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 section (LP, BP, and HP)	When the Side-Chain button is activated, you can use these buttons to set the filter type to low-pass, band-pass, or high-pass.

Parameter	Description
Side-Chain button	Activates the side-chain filter. The input signal can then be shaped according to set filter parameters. Internal side-chaining can be useful for tailoring how the gate operates.
Center (50 to 20000 Hz)	When the Side-Chain button is activated, this sets the center frequency of the filter.
Q-Factor (0.01 to 10000)	When the Side-Chain button is activated, this sets the resonance of the filter.
Monitor button	Allows you to monitor the filtered signal.
Attack (0.1 to 1000 ms)	Sets the time after which the gate opens after being triggered. Deactivate the Live button to make sure that the gate is already 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 to 2000 ms)	Determines how long the gate stays open after the signal drops below the threshold level.
Release (10 to 1000 ms or Auto mode)	Sets the time after which the gate closes (after the set hold time). If the Auto button is activated, Gate finds an optimal release setting, depending on the audio material.
Analysis (0 to 100) (Pure Peak to Pure RMS)	Determines whether the input signal is analyzed 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 button	When this button is activated, the look-ahead feature of Gate is disengaged. Look ahead produces more accurate processing, but adds 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.

⇨ If side-chaining is supported, the gate can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the gate opens. For a description of how to set up side-chain routing, see the Operation Manual.

Limiter

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–

Limiter is designed to ensure that the output level never exceeds a 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 following parameters are available:

Parameter	Description
Input (-24 to +24dB)	Adjusts the input gain.
Output (-24 to +6dB)	Determines the maximum output level.
Release (0.1 to 1000ms or Auto mode)	Sets the time after which the gain returns to the original level. If the Auto button is activated, Limiter automatically finds an optimal release setting that varies depending on the audio material.

Maximizer

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

Maximizer raises 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 tube-like distortion to the signal.

The following parameters are available:

Parameter	Description
Output (-24 to +6dB)	Determines the maximum output level. Should normally be set to 0 to avoid clipping.
Optimize (0 to 100)	Determines the loudness of the signal.
Soft Clip button	When this button is activated, Maximizer starts limiting (or clipping) the signal softly, at the same time generating harmonics which add a warm, tube-like characteristic to the audio material.

MIDI Gate

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

Gating, in its fundamental form, silences audio signals below a set threshold level. 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 not triggered by threshold levels, but MIDI notes. Hence it needs both audio and MIDI data to function.

Setting up

To set up MIDI Gate, proceed as follows:

1. Select the audio to be affected by 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 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 effect.
   This can be an empty MIDI track or a MIDI track containing data, it does not matter. However, if you wish to use MIDI Gate in realtime – as opposed to using a recorded part – 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 effect.

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

5. Make sure the MIDI track is selected, and start playback.
6. 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	Determines how long it takes 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).
Release	Determines how long it takes for the gate to close (in addition to the value set with the Hold parameter).
Note To Attack	Determines to which extent the velocity values of the MIDI notes affect the attack. The higher the value, the more the attack time increases with high note velocities. Negative values 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	Determines to which extent the velocity values of the MIDI notes affect the release. The higher the value, the more the release time increases. If you do not wish to use this parameter, set it to the 0 position.
Velocity To VCA	Controls to which extent the velocity values of the MIDI notes determine the output volume. At a value of 127 the volume is controlled entirely by the velocity values, and at a value of 0 the velocities have no effect on the volume.
Hold Mode	Use this switch to set the Hold Mode. In Note-On mode, the gate only remains 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, the gate remains open for as long as the MIDI note plays, and then the Hold and Release parameters are applied.

MultibandCompressor

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

The MultibandCompressor allows a signal to be split into a maximum of 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 attenuate or boost the input gain by ±15 dB 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 is the threshold point.

For each of the four bands the following compressor parameters are available:

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

The Output Control

The Output knob controls the total output level of the MultibandCompressor. The range is from -24 to +24dB.

Tube Compressor

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–
Side-chain support	–	–	X	X	X	X	–

This versatile compressor with integrated tube-simulation allows you to achieve smooth and warm compression effects. The VU meter shows the amount of gain reduction. Tube Compressor features an internal side-chain section that lets you filter the trigger signal.

The following parameters are available:

Parameter	Description
Drive (1.0 to 6.0)	Controls the amount of tube saturation.
Input (-24.0 to +48.0)	Determines the compression amount. The higher the input gain setting, the more compression is applied.
Limit button	Increases the ratio of the compressor for a limiting effect.
Output (-12.0 to +12.0)	Sets the output gain.
Attack (0.1 to 100.0)	Determines how fast the compressor responds. If the attack time is long, more of the initial part of the signal (attack) passes through unprocessed.
Release (10 to 1000 ms or Auto mode)	Sets the time after which the gain returns to the original level. If the Auto button is activated, Tube Compressor automatically finds an optimal release setting that varies depending on the audio material.
Mix (0 to 100)	Adjusts the mix between dry and processed signal preserving the transients of the input signal.
In/Out Meters	Show the highest peaks of all available input and output channels.
VU Meter	Shows the amount of gain reduction.
Side-chain button (if supported)	Activates/deactivates the internal side-chain filter. The input signal can then be shaped according to set filter parameters. Internal side-chaining is useful for tailoring how the compressor operates.
Filter section (LP, BP, and HP)	When the Side-Chain button is activated, you can use these buttons to set the filter type to low-pass, band-pass, or high-pass.
Side-Chain section: Center	Sets the center frequency of the filter.
Side-Chain section: Q-Factor	Sets the resonance or width of the filter.
Side-Chain section: Monitor	Allows you to monitor the filtered signal.

VintageCompressor

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–
Side-chain support	–	–	–	–	X	X	–

This is modelled after vintage type compressors. This compressor features separate controls for input and output gain, attack, and release. 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.

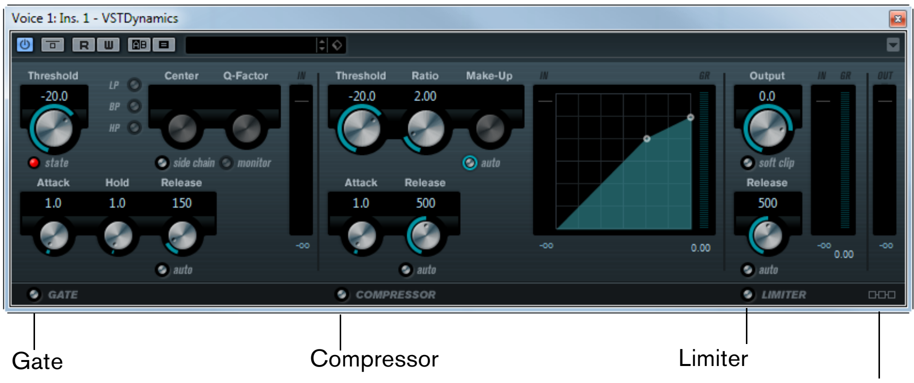
The available parameters work as follows:

Parameter	Description
Input (-24 to 48 dB)	In combination with the Output setting, this parameter determines the compression amount. The higher the input gain setting and the lower the output gain setting, the more compression is applied.
Output (-48 to 24 dB)	Sets the output gain.
Attack (0.1 to 100 ms)	Determines how fast the compressor responds. If the attack time is long, more of the early part of the signal (attack) passes 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 to 1000 ms or Auto mode)	Sets the time after which the gain returns to the original level. If the Auto button is activated, Vintage Compressor automatically finds an optimal release setting that varies depending on the audio material.
Ratio (2:1, 4:1, 8:1, and 20:1)	Determines the amount of gain reduction applied to signals above the threshold. A ratio of 4:1 means that for every 4 dB the input level increases, the output level increases by only 1 dB.
VU Meter	Shows the amount of gain reduction.
In/Out Meters	Show the highest peaks of all available input and output channels.

⇨ If side-chaining is supported, the compression can also be controlled from another signal source via 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 Operation Manual.

VSTDynamics

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–



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 set threshold level. As soon as the signal level exceeds the set threshold, the gate opens to let the signal through.

The following parameters are available:

Parameter	Description
Threshold (-60 to 0dB)	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 close the gate.
State LED	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 button (if supported)	Activates the internal side-chain filter. You can use this to filter out parts of the signal that might otherwise trigger the gate in places you not want it to, or to boost frequencies you wish to accentuate, allowing for more control over the gate function.
Filter section (LP, BP, and HP)	When the Side-Chain button is activated, you can use these buttons to set the filter type to low-pass, band-pass, or high-pass.
Center (50 to 22000Hz)	Sets the center frequency of the filter.
Q-Factor (0.001 to 10000)	Sets the resonance or width of the filter.
Monitor (On/Off)	Allows you to monitor the filtered signal.

Parameter	Description
Attack (0.1 to 100 ms)	Sets the time after which the gate opens after being triggered.
Hold (0 to 2000 ms)	Determines how long the gate stays open after the signal drops below the threshold level.
Release (10 to 1000 ms or Auto mode)	Sets the time after which the gate closes (after the set hold time). If the Auto button is activated, Gate finds an optimal release setting, depending on the audio material.
Input gain meter	Shows the input gain.

The Compressor section

The compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. It works like a standard compressor with separate controls for threshold, ratio, attack, release and make-up gain. The compressor features a separate display that graphically illustrates the compressor curve shaped according to the Threshold, Ratio and Make-Up Gain parameter settings. It also features meters for input gain and gain reduction and a program-dependent Auto feature for the Release parameter.

The available parameters work as follows:

Parameter	Description
Threshold (-60 to 0 dB)	Determines the level where the compressor kicks in. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1:1 to 8:1)	Determines the amount of gain reduction applied to signals above the set threshold. A ratio of 3:1 means that for every 3 dB the input level increases, the output level increases by only 1 dB.
Make-Up (0 to 24 dB)	Compensates for output gain loss, caused by compression. When the Auto button is activated, gain loss is being compensated automatically.
Attack (0.1 to 100 ms)	Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.
Release (10 to 1000 ms or Auto mode)	Sets the time after which the gain returns to the original level when the signal drops below the threshold. If the Auto button is activated, the compressor automatically finds an optimal release setting that varies depending on the audio material.
Graphical display	Use the graphical display to graphically set the Threshold and Ratio values. To the left and right of the graphical display you find two meters that show the amount of input gain and gain reduction in dB.

The Limiter Section

The limiter is designed to ensure that the output level never exceeds a set threshold, 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. The limiter adjusts and optimizes these parameters automatically according to the audio material. You can also adjust the Release parameter manually.

The following parameters are available:

Parameter	Description
Output (-24 to +6 dB)	Determines the maximum output level. Signal levels above the set threshold are affected, but signal levels below are left unaffected.
Soft Clip button	If this button is activated, the limiter acts differently. When the signal level exceeds -6 dB, Soft Clip starts limiting (or clipping) the signal softly, at the same time generating harmonics which add a warm, tube-like characteristic to the audio material.
Release (10 to 1000 ms or Auto mode)	Sets the time after which the gain returns to the original level when the signal drops below the threshold. If the Auto button is activated, the limiter automatically finds an optimal release setting that varies depending on the audio material.
Meters	The three meters show the input gain (IN), the gain reduction (GR) and the output gain (OUT).

The Module Configuration Button

Using the Module Configuration button in the bottom right corner of the plug-in panel, 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.

DJ-EQ

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

DJ-Eq is an easy-to-use 3-band parametric equalizer that resembles the EQs found on typical DJ mixers. This plug-in is designed for quick sound fixes.

To set the Low, Mid, and High frequency bands, you can:

- Move the mouse over the curve display, and click and drag the EQ points.
  Press the [Shift] key and drag to adjust the values in smaller steps. Press [Ctrl]/[Command] and click a parameter to set it to zero.
- Click the Gain values and move the mouse up or down to change them.

The following parameters are available:

Parameter	Description
Low Gain	Sets the amount of attenuation/boost for the low band.
Low Kill (Activates Low Cut)	Cuts the low band.
Mid Gain	Sets the amount of attenuation/boost for the mid band.
Mid Kill (Activates Mid Cut)	Cuts the mid band.
Hi Gain	Sets the amount of attenuation/boost for the high band.
Hi Kill (Activates High Cut)	Cuts the high band.
Output meter	Shows the overall output level.

GEQ-10/GEQ-30

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–/–	–/–	X/–	X/X	X/X	X/X	–/–

These graphic equalizers are identical in every respect except for the number of available frequency bands (10 and 30, respectively). Each band can be attenuated or boosted by up to 12dB, 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 individual frequency bands are shown in Hz.
- At the top of the display the amount of attenuation/boost is shown in dB.

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

Parameter	Description
Output	Controls the overall gain of the equalizer.
Flatten button	Resets all the frequency bands to 0dB.
Range	Allows you to relatively adjust how much a set curve attenuates or boosts the signal. If the Range parameter is turned fully clockwise, the range is ±12dB.
Invert button	Inverts the current response curve.
Mode pop-up menu	The filter mode set here determines how the various frequency band controls interact to create the response curve, see below.

About the Filter Modes

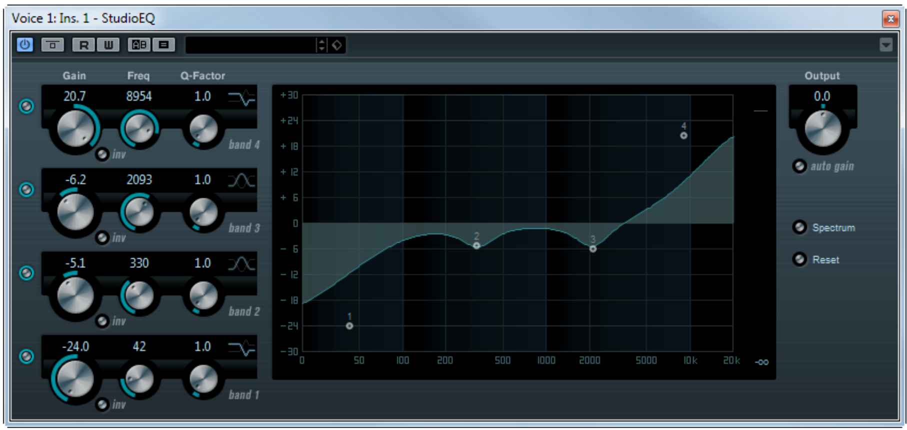
On the pop-up menu in the lower right corner there are several different EQ modes available. These modes can add color or character to the equalized output in various ways. The following filter modes are available:

Filter mode	Description
True Resp	Applies serial filters with an accurate frequency response.

Filter mode	Description
Digi Stand	In this mode the resonance of the last band depends on the sample rate.
Classic	Applies a classic parallel filter structure where the response does not follow the set gain values accurately.
Variable Q	Applies parallel filters where the resonance depends on the amount of gain.
ConstQ u	Applies parallel filters where the resonance of the first and last bands depends on the sample rate.
ConstQ s	Applies parallel filters where the resonance is raised when boosting the gain and vice versa.
Resonant	Applies serial filters where a gain increase of one band lowers the gain in adjacent bands.

StudioEQ

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



StudioEQ is a high-quality 4-band parametric stereo equalizer with two fully parametric mid-range bands. The low and high bands can act as either shelving filters (three types), or as a Peak (band-pass) or Cut (low-pass/high-pass) filter.

Making Settings

1. Click the corresponding On button on the left of the plug-in panel to activate any or all of the 4 equalizer bands (Low, Mid 1, Mid 2, and High).

   When a band is activated, the 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 on the numeric values and entering new values.
- By using the mouse to drag points in the EQ curve display.

When using the mouse to change the parameter settings, the following modifier keys can be used:

Modifier key	Description
–	When no modifier key is pressed and you drag an EQ point in the display, the Gain and Frequency parameters are adjusted simultaneously.
[Shift]	Keep the [Shift] key pressed and drag the mouse to change the Q-factor of the corresponding EQ band.
[Alt]/[Option]	Keep the [Alt]/[Option] key pressed and drag the mouse to change the frequency of the corresponding EQ band.
[Ctrl]/[Command]	Keep the [Ctrl]/[Command] key pressed and drag the mouse to change the gain value of the corresponding EQ band.

The following parameters are available:

Parameter	Description
Band 1 Gain (-20 to +24dB)	Sets the amount of attenuation/boost for the low band.
Band 1 Inv button	Inverts the gain value of the filter. Use this button to filter out unwanted noise. When looking for the frequency to omit, it sometimes helps to boost it first (set the filter to positive gain). After you have found it, you can use the Inv button to cancel it out.
Band 1 Freq (20 to 2000Hz)	Sets the frequency of the low band.
Band 1 Q-Factor (0.5 to 10)	Controls the width or resonance of the low band.
Band 1 Filter mode	For the low band, you can select between three types of shelving filters, a Peak (band-pass), and a Cut (low-pass/high-pass) filter. When Cut mode is selected, the Gain parameter is fixed. -Shelf I adds resonance in the opposite gain direction slightly above 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.
Band 2 Gain (-20 to +24dB)	Sets the amount of attenuation/boost for the mid 1 band.
Band 2 Inv button	Inverts the gain value of the filter (see the description of the Invert button for Band 1).
Band 2 Freq (20 to 20000Hz)	Sets the center frequency of the mid 1 band.
Band 2 Q-Factor (0.5 to 10)	Sets the width of the mid 1 band: the higher this value, the narrower the bandwidth.
Band 3 Gain (-20 to +24dB)	Sets the amount of attenuation/boost for the mid 2 band.
Band 3 Inv button	Inverts the gain value of the filter (see the description of the Invert button for Band 1).
Band 3 Freq (20 to 20000Hz)	Sets the center frequency of the mid 2 band.
Band 3 Q-Factor (0.5 to 10)	Sets the width of the mid 2 band: the higher this value, the narrower the bandwidth.

Parameter	Description
Band 4 Inv button	Inverts the gain value of the filter (see the description of the Invert button for Band 1).
Band 4 Gain (-20 to +24dB)	Sets the amount of attenuation/boost for the high band.
Band 4 Freq (200 to 20000Hz)	Sets the frequency of the high band.
Band 4 Q-Factor (0.5 to 10)	Controls the width or resonance of the high band.
Band 4 Filter mode	For the high band, you can select between three types of shelving filters, a Peak, and a Cut filter. When Cut mode is selected, the Gain parameter is fixed. -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 +24dB)	This knob on the top right of the plug-in panel adjusts the overall output level.
Auto Gain button	When this button is activated, the gain is automatically adjusted, keeping the output level constant regardless of the EQ settings.
Spectrum	Shows the spectrum before and after filtering.
Reset	Resets the EQ settings.

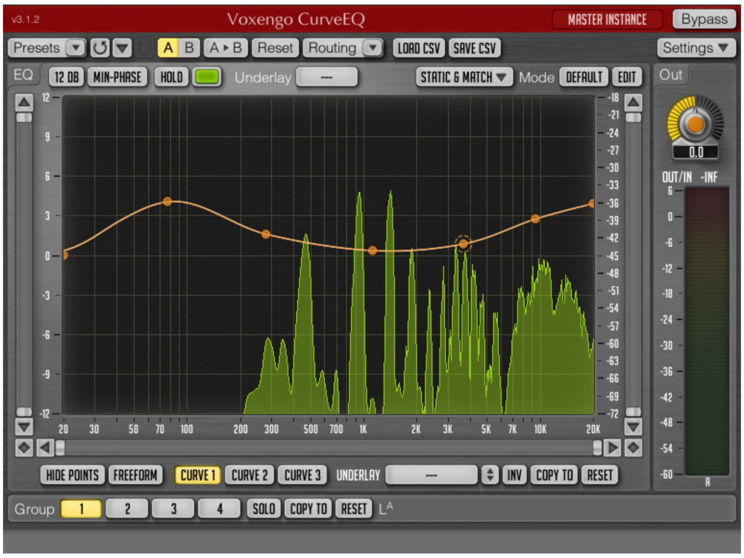
CurveEQ

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	-	X	X	–

Voxengo CurveEQ is a spline equalizer for professional music and audio production applications. CurveEQ shows the filter response you are designing by means of a spline, that is, a smooth curvy line. This way you can see how the EQ alters the sound.

CurveEQ implements spectrum matching technology that allows you to transfer the spectral shape of one recording to another. In other words, you can copy the frequency balance of existing time-proven mixes so that other mixes can be improved. CurveEQ's filters can be switched between linear-phase and minimum-phase modes. CurveEQ also features a customizable spectrum analyzer. Furthermore, you can display, save, and load static spectrum plots for comparison and matching purposes.

Main Layout



Title Bar

Parameter	Description
Plug-in instance name	This text box allows you to name the current plug-in instance.
Bypass	Use this button to compare the sound of the unprocessed signal to that of the processed signal. The Bypass button does not reduce the plug-in's CPU load when switched on. The bypass state is not saved between project sessions and is not restored when the project is reloaded.

General Control Bar

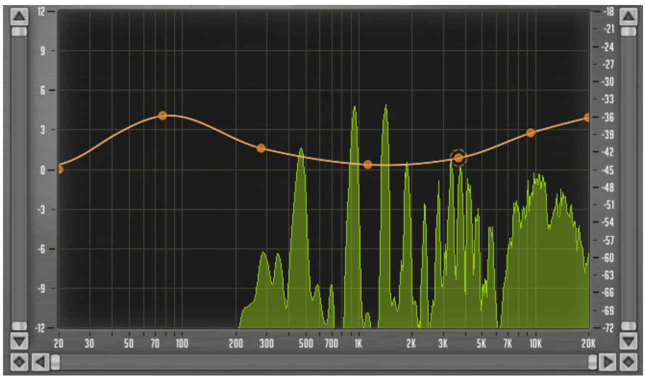
Parameter	Description
Presets selector	Allows you to store and restore custom settings, see “Main Preset Manager” on page 54.
Undo	Allows you to undo changes.
History	Opens a change log that lists up to 32 changes in the order you have made them. Parameter changes are logged with the group name in parentheses, for example, “Gain (Ls) change”.
Redo	Allows you to redo changes that were undone.
A/B button	By pressing the A/B button, you can switch between two plug-in states (A and B).
A>B (B>A) button	Copies the current plug-in state to the other state (A or B). This is useful to copy programs between Session Bank slots.
Reset	This is the master reset button. It resets the plug-in to its default state. The default state can be chosen in the Preset Manager window, see “Preset Manager” on page 54.
Routing selector	The Routing button opens the Channel Routing Window, where you can change several routing options. The pop-up menu provides access to common routing options, see “Channel Routing Window” on page 55.
Save CSV	Allows you to save the selected EQ curve in a comma-separated text file. The EQ curve is stored as series of frequency/gain pairs, one per line, in the following form: 20.00,3.00 400.00,2.51 1000.00,1.45 # comment 5000.00,3.40 20000.00,1.05 Each pair defines the position of a single control point on the CurveEQ’s control surface. Write decimal points as a period, not as a comma. Comments can be added at any position, starting with a hash character.
Load CSV	Allows you to load a previously saved CSV file or any externally-generated EQ curve specification, such as room correction or RIAA phono correction. Frequencies defined in the file should lie between 20 and 20000 Hz.
Settings	Allows you to change general settings, see “CurveEQ Settings” on page 56.

EQ Top Control Bar

Parameter	Description
Equalizer dB gain range	Lets you change the maximum gain when boosting/decreasing frequencies per band.

Parameter	Description
MIN-Phase	Enables minimum-phase filtering instead of linear-phase filtering. Minimum-phase filtering sounds better at steeper EQ slopes because it lacks pre-ringing artifacts present in linear-phase filters. Furthermore, it does not add a considerable processing latency.
Static & Match	Opens the Static Spectrums Editor, where you can display static spectrums and perform spectrum matching. Spectrum matching allows you to match the spectrum shape of a sound recording to that of another sound recording.
Mode selector	Allows you to select a mode for spectrum matching, see “Spectrum Matching” on page 50.
Edit	Opens the Spectrum Mode Editor, see “Spectrum Mode Editor” on page 51.

Main EQ Control Surface



The heart of CurveEQ is the equalizer control surface with a built-in real-time spectrum analyzer.

- To add a control point, double-click the curve.
- To delete a control point, double-click it.

The picture above shows the equalizer control surface with control points that can be dragged with the left mouse button to adjust the filter’s gain and frequency. For more precise adjustments, hold [Shift] while dragging.

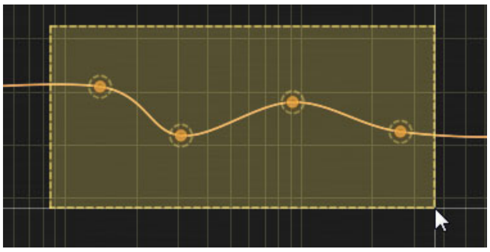
The readouts show the mouse cursor position within the display, the musical note and detune in cents that correspond to the frequency position, and the mouse cursor position within the spectrum power range.

If two or three curves are displayed, a white curve shows the summary frequency response of all currently enabled filters.

While dragging a control point with the left mouse button, you can adjust the filter’s bandwidth by additionally holding the right mouse button or pressing [Alt]/[Option]. Alternatively, you can use the mouse wheel to adjust the filter’s bandwidth.

- To enable the gain adjustment only, press [Ctrl]/[Command] while dragging a point.
- To enable frequency adjustment only, press [Ctrl]/[Command]-[Alt]/[Option].
- To set a control point to 0dB, press [Ctrl]/[Command], and double-click it.

Equalizer – Group Editing



You can perform editing operations on a group of control points.

- To select several control points, click inside the equalizer control surface and drag a rectangle over the control points that you want to select.
- To select all control points at once, right-click the control surface.
- To deselect any currently selected points, click in the control surface.
- To add control points to the current selection, press [Shift] and click the control points that you want to add.
- To remove control points from the selection, hold [Shift] and click the control point that you want to remove.

For group editing, the following buttons are available:

Option	Description
Up/down arrow button	Allows you to scale the gain of the selected control points.
Inv	Inverts the gain of the selected control points.
Reset	Resets the current filter to its default state.

Equalizer – Spectrum

The equalizer control surface can display the Fourier spectrum analysis plot. The spectrum analysis and the display of parameters can be selected via the Mode selector. The Spectrum Mode Editor can be used to customize these parameters further. You can also click the control surface anywhere to reset the spectrum analysis display.

A red vertical line is displayed if the visible frequency range is wide. This line shows the maximum frequency of the input signal and depends on the input sample rate. Note that until you start the audio playback, the red line cannot be placed correctly, because the plug-in does not know the correct input sample rate before the audio processing is started.

By default, Voxengo plug-ins use a slope value of 4.5 dB per octave for the spectrum display. This setting can be changed in the Spectrum Mode Editor window.

To zoom in on the spectrum’s peak values, [Alt]/[Option]-click and drag a selection rectangle.

If the spectrum does not fit the display, adjust the visible spectrum range in the Spectrum Mode Editor.

Equalizer – Narrow-Band Sweeping

To highlight the resonances in the sound, you can enable the narrow-band sweeping function by pressing [Ctrl]/[Command] and dragging in the control surface with the left mouse button. As a result of this action, the curve of the band-pass filter only passes the selected frequency range. You can adjust the bandwidth of the filter with the mouse wheel.

The band-pass filter’s curve is applied on top of the existing equalizer curve. This means that the curve you see when engaging the narrow-band sweeping is composed of the existing equalizer curve and band-pass filter’s own equalizer curve.

Zooming

- To zoom into the spectrum display, press [Alt]/[Option] and drag the control surface.
- To zoom out of the spectrum display, press [Alt]/[Option] and double-click the control surface.

Scrollbar

The horizontal and vertical scrolling controls feature zooming functionality. The scrollbars are found at the sides of the equalizer control surface.

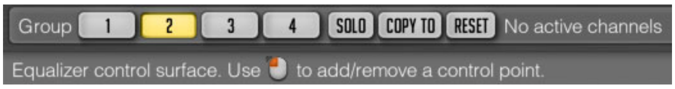
The diamond-shaped button between a horizontal and vertical scrollbar can be used to control the positions of both scrollbars at once in a single X-Y coordinate space.

You can double-click scrollbars and diamond-shaped buttons to quickly switch between the zoomed and non-zoomed views of the control surface.

EQ Bottom Control Bar

Parameter	Description
Hide Points	Hides the control points, which allows you to evaluate the EQ curve more precisely.
Freeform	Enables freeform mode, in which you can draw the EQ curve manually by drawing on the control surface with the left mouse button. Note that switching to freeform mode and back can be destructive and some EQ curve features can be lost.
Curve 1/2/3	You can define up to 3 equalizer curves for every channel group. This is useful when you are using spectrum matching. For example, you can apply a matching EQ curve generated automatically and at the same time apply any additional EQ curve that you draw manually. Note that CurveEQ has a lower resolution at the frequencies below 200 Hz. At these frequencies, the EQ curve does not always follow the control point positions.
Underlay	Allows you to select another EQ curve from any other channel group that is displayed as an underlay.
Up/down arrow button	Allows you to scale the gain of the EQ curve.
Inv	Inverts the current EQ curve.
Copy To	Copies the envelope to the same envelope in another group.
Reset	Resets the current EQ curve to its default state.

Group Bar and Hint Line



Parameter	Description
Group 1/2/3/4	These buttons represent the channel groups. You can select the channel group whose parameters are being edited or monitored. Only groups that are assigned to the internal channels in the Channel Routing window are shown.
Solo	Allows you to solo the output of the selected group. The state of the Solo button is not saved between project sessions and is not restored when the project is reloaded.
Copy To	Allows you to copy parameter settings defined for the selected channel group to another channel group.
Reset	Resets the parameters of the active group.

⇨ Note that the group bar is not visible if the "Min Infrastructure" option in the Settings window is activated. In that case, you can use the Routing selector to select a channel group.

Channel Group List

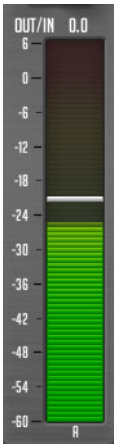
CurveEQ shows a list of input channels that are routed to the selected channel group. This list is connected to the Channel Routing window and displays routing settings defined by it. Internal channel names (A, B, C, etc.) that accept the corresponding input channel are displayed in a superscript style. These internal channel names are also displayed on the level meters. If more than one input channel is routed to the same internal channel, the sum is displayed in the form "(IN1+IN2)".

When the internal channel is assigned to a mid/side group, its input channels are written in parentheses with the "m" (mid) or "s" (side) prefix. For example, "s(IN1 & IN2)" means "side part of the mid/side pair consisting of IN1 and IN2 input channels".

Hint Line

This interface element displays hint messages and can also display other informational messages. The hint line can be disabled in the Settings window.

Level Meter



The level meter shows several bars that correspond to the channels (A, B, etc.) of the selected channel group. The level meter displays all available channels if the "Show All Channel Meters" button is activated in the Channel Routing window.

Level meters can show a small horizontal white bar that represents the peak level. In output level meters, such as peak level, it can turn red. This means that the output level has entered the area above the 0 dBFS signal level and clipping can occur if the plug-in is inserted at the final position in the signal chain of the host application. If the plug-in is inserted in an intermediate position, that is, before other plug-ins, clipping does not necessarily occur.

Level meter ballistics and peak level hold time can be defined for all instances of the plug-in in the Settings window.

Output level meters usually feature a "Out/In" display, showing the difference in RMS level between the input and output signals of the plug-in.

Spectrum Matching

With CurveEQ you can match the sound of any audio track to another, whether it is your to-die-for guitar intro or your favorite kick drum sample.

All spectrum related functions are located in the "Static & Match" display.

⇨ Spectrum matching uses parameters specified in the Spectrum Mode Editor. Only spectrums present in static spectrum slots can be used for matching. The usual realtime primary and secondary spectrums are not used for matching, unless taken as snapshots by means of the Take or "Take 2nd" buttons, respectively.

When you perform spectrum matching it is suggested to set the Type selector in the Spectrum Mode Editor to "Avg", so that average spectrum is used for matching. You must run the averaging for several seconds until the visible spectrum becomes smooth enough. After achieving the required spectrum shape on the screen you can click the Take (or "Take 2nd") button in the static spectrum slot to store this spectrum for matching purposes.

You need at least two spectrum snapshots in two slots for matching. The spectrum that you want to equalize and the reference spectrum should be marked with the "Apply To" and "Reference" switches, respectively. You can define more than one "Apply To" or "Reference" spectrum. In that case the mean value of the spectrums is used.

The Points parameter specifies how many equidistant points to use for matching. The more points you use the more precise the match will be. However, in many cases more precise match does not mean a better sounding match. It is suggested to try several values to determine which one sounds best.

⚠ The EQ curve present on the screen affects the spectrum averaging process, so the EQ curve should be flat when spectrum data is being collected.

⇨ The static spectrum's gain shift has no effect on the matching process.

Spectrum Mode Editor

Spectrum matching options are placed in the Spectrum Mode Editor, which can be opened by clicking the Edit button on the EQ top control bar.

Parameter	Description
Spectrum Disable	Disables the spectrum analysis function of the plug-in.
Filled Display	Enables additional semi-transparent filling of the spectrum display.
2nd Spectrum	Enables the secondary spectrum curve, which is displayed in a darker color.
Type selector	Allows you to select a spectrum analysis type. The "RT Avg" mode applies realtime spectrum averaging analysis. This type of analysis produces an RMS-averaged spectrum over the period specified by the "AVG Time" parameter. The analysis type "Max" produces a cumulative maximum power spectrum. The "Avg" type produces a cumulative average power spectrum. The "RT Max" mode produces a realtime maximum spectrum with spectrum fall-down. For better spectrum maximum estimate, use a higher Overlap setting. If you need an infinite peak hold, use the "Max" analysis type.

Parameter	Description
Block Size	Specifies the block size of the FFT (fast Fourier transform) spectrum analyzer. Higher block sizes provide more resolution in the lower frequency range, but decrease time coherence (time precision) in the higher frequency range; the higher frequency information becomes over-averaged. Also, at higher block size settings the spectrum is refreshed less frequently. This can be compensated by increasing the Overlap parameter. When working at increasingly higher sample rates, you need to increase the block size value, because the setting is used over the full spectral bandwidth. Therefore, at higher sample rates the analyzer's resolution in the visible frequency range will be lower for the given block size. If you want to measure the frequency of a low-frequency sound such as a drum or bass guitar precisely, use a higher "Block Size" value along with a higher Overlap value. In order to avoid clicks and glitches in playback when using high "Block Size" values, you need to increase the audio buffer size in your host application.
2nd Type	If "2nd Spectrum" is activated, you can use this pop-up menu to select an analysis type for the secondary spectrum. For example, by setting the "2nd Type" to "RT Max" and "Type" to "RT Avg", you can see the average and maximum spectrums simultaneously. Note that the secondary spectrum uses the same "Block Size" and "Avg Time" values as the primary spectrum.
Overlap	Controls the overlap between the adjacent FFT spectrum analysis windows. Higher overlap values allow spectrum to be updated more frequently at the expense of a higher CPU load.
AVG Time	Specifies the average (fall-down) time used when the "RT Avg" or "RT Max" analysis is active. This value specifies after how many milliseconds the spectrum level falls down by 20 dB.
Smoothing	Lets you select the smoothing function's resolution in octaves. Smoothing produces a drop of 6 dB per octave when stationary sine wave signals are used. For example, even if the signal consists of 2 sine waves (1 kHz and 2 kHz) of equal peak amplitude, the 2 kHz sine wave looks like it is 6 dB quieter. This happens because the fast Fourier transform produces a narrower spectrum for high-frequency stationary signals in comparison to low-frequency stationary signals. This drop does not appear when non-stationary (musical) signals are analyzed.
Freq Low/Freq High	Specify the visible frequency range of the spectrum view.
Range Low/Range High	Specify the accessible spectrum power range.
Slope	Allows you to adjust slope in the spectrum analyzer display around 1 kHz. Skewing the spectrum can be useful because higher frequencies usually have weaker power in comparison to the lower frequencies. By choosing an appropriate spectrum slope, you can compensate for this fact.

Static Spectrums Editor

CurveEQ features a static spectrum display that can be controlled via the Static Spectrums Editor.

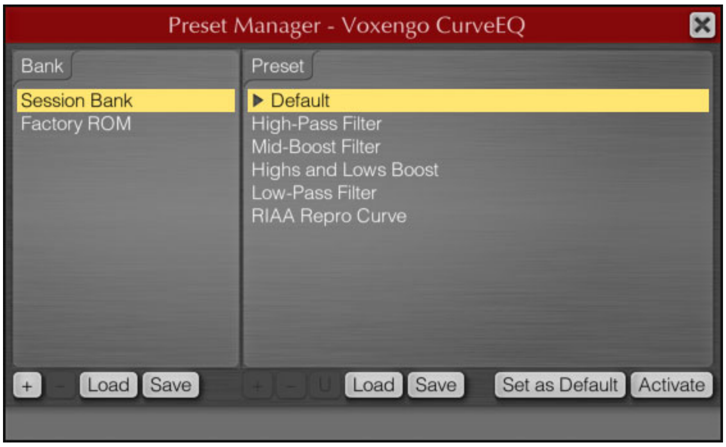
You can select the display name of the spectrum slot, its color, and the shift in dB of the static spectrum. The static spectrum can be shown or hidden using the visibility checkbox. The shift in dB can be used for a more convenient placing of the static spectrum on the screen and it does not affect the shape of the spectrum.

Parameter	Description
Take/Take 2nd	These buttons take a snapshot of the primary or secondary spectrum, respectively. The static spectrum snapshots are taken using the spectrum parameters specified in the Spectrum Mode Editor. Before taking a spectrum, choose a spectrum analysis type via the Spectrum Mode Editor, usually “Avg” or “Max”, and analyze long enough so that the spectrum becomes general enough. When analyzing a song, it is recommended to store separate spectrums for verse, chorus, and bridge parts, as they can have distinctively differing spectral balance. If no snapshot is taken after pressing a Take button, no spectrum is available. You either have to configure the spectrum mode or start the audio playback first.
Load/Save	You can save the spectrum in a static spectrum slot as a spectrum file with the extension .csf (compressed spectrum file).
X	Resets the spectrum in the selected slot.

Preset Manager

Main Preset Manager

You can use the main preset manager to save and load plug-in state presets.



Presets in the main preset manager are shared among all instances of the same Voxengo plug-in. All presets within the main preset manager are stored in user preset banks. Beside user preset banks two special banks exist: the Session Bank and Factory ROM bank.

The Session Bank contains programs rather than presets. Each program in the Session Bank contains its own undo/redo change log. The Session Bank lists programs that mirror programs of the host application. When you activate a program in the Session Bank, the program in the host application switches.

The Factory ROM bank contains presets that cannot be changed. The Factory ROM bank is loaded into the Session Bank every time a new instance of the plug-in is created in the host application.

The main preset manager contains the following control buttons:

Parameter	Description
+/-	Allow you to add and remove a bank or preset. Right-clicking the plus button (+) inserts the preset at the current list position rather than at the end of the list.
Load/Save	Allow you to save and load the bank or preset to and from a file.
U	Updates the selected preset with the current plug-in state.
Set as Default	Makes the selected preset the default preset. The default preset is loaded every time a new plug-in instance is created in the host application or when the master Reset button is pressed. If you want to restore the original default preset, select the “Default” preset in the Factory ROM bank and click the “Set as Default” button.
Activate	Loads the selected preset. You can also double-click a preset name.

⇨ Voxengo plug-ins use a proprietary format to store presets and preset banks. Add a meaningful prefix to bank and preset file names so that you do not mix up presets created in different Voxengo plug-ins. Voxengo plug-in preset files have the extension .cpf, preset bank files the extension .cbf.

To rename a preset or bank, select it and after a small delay click the item again.

Channel Routing Window

In the Channel Routing window, the following options are available:

Parameter	Description
Routing Presets	Opens a window that contains presets for the Channel Routing window, including channel labels.
Show all Channel Meters	Enables displaying of all channel meters and statistics counters regardless of the currently selected channel group. When this option is deactivated, only meters belonging to the currently selected channel group are shown. Activating this option is useful when you are using dual-mono or mid-side processing. This option allows you to see channel meters for left and right, or mid and side channels together.
Input and Output Routing	Allow you to route external plug-in inputs to internal plug-in channels and vice versa, and to route internal plug-in channels to external plug-in outputs. The plug-in has a pre-defined number of internal channels, but the number of input and output channels can vary depending on the host application's track or bus on which the plug-in is inserted. Note that if the input routing selector is red, the selector refers to a non-existent input channel. You can correct this by selecting an existing channel. External side-chain inputs are denoted by parenthesized labels, for example, "(IN3)", "(IN4)".
Mid/Side Pairs	Allow you to assign internal channels to mid/side pairs for encoding and decoding. The mid/side encoding is a wide-spread technique that allows you to process the middle (center) and side (spatial) information in stereo signals independently of each other, thus offering a great deal of control over that signal's stereophony. Mid/side encoding works with paired channels only and thus requires two channels to be assigned to the same mid/side pair. An input signal is mid/side encoded before it is processed by the plug-in, and decoded afterwards before it is routed to an output of the plug-in.

Parameter	Description
Group Assignments	The plug-in allows you to assign its internal audio channels to logical channel groups. Each group is affected by its own set of parameter values (EQ shape, gain factor, overdrive setting, etc.). The current channel group is selected via the channel group selector. Individual audio channels can be assigned to different channel groups. For example, you can make separate EQ settings for channel 1 and for channel 2 by assigning channel 1 to group 1 and channel 2 to group 2. In a surround setup you can assign left and right channels to group 1 and surround channels to group 2, and apply different EQ shapes to the groups. Each plug-in audio channel can be assigned to a single channel group only. Channel grouping also affects channel-linking in case of dynamics processing and other processes that estimate signal loudness envelope: channels assigned to the same group are linked during processing and signal loudness estimation.
IN Channel Labels	Opens the label assignment window where you change the display names for the input channels. You can also import channel labels from the host application by pressing the “Import labels from host” button. However, not all host applications provide distinctive input channel names.
Group Names	Opens the group names where you can change the display names for the groups.

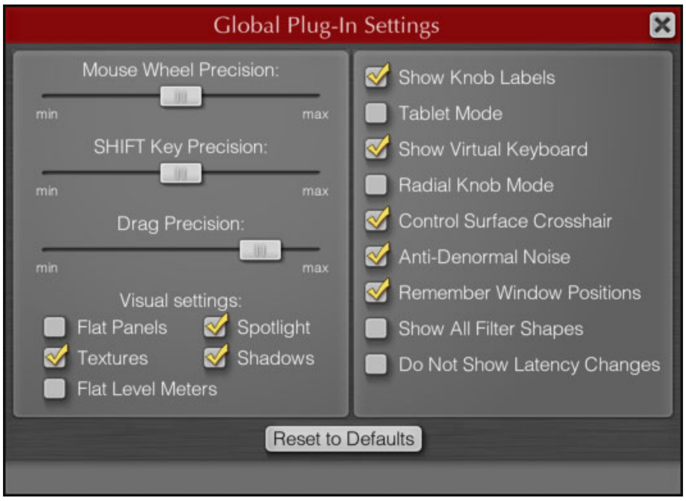
CurveEQ Settings

In the CurveEQ Settings window the following parameters are available:

Parameter	Description
Color scheme	The icons show possible color schemes. To change the color scheme, click an icon.
UI scale	Adjusts the size of the plug-in panel. Note that changing this setting requires a restart of the host application.
Show hints	If activated, hint messages appear at the bottom of the plug-in panel.
Min Infrastructure	Activate this to hide part of the plug-in interface in favor of showing a larger EQ control surface.

Parameter	Description
Level meter settings – Density mode	Activates the density metering mode. In this mode you can see levels at which a signal stays often. By examining the range of levels at which a signal stays, you can draw conclusions about the effective dynamic range of the material. Note that the signal level estimation is affected by the meter's integration and release times. In this mode, the display of the signal level is also affected by the Peak Level hold time setting.
Level meter settings – Integration time	Affects the level integration time of all level meters. The value reflects the time it takes for a signal level to fall down by 20 dB, or raise up from one steady level to another steady level. Note that this setting does not affect the peak level on the level meters, but directly affects the visible difference between the peak and RMS levels when a musical signal is measured.
Level meter settings – Release Time	Changes the level meter's release time. This is the time it takes for a signal to fall down by 20 dB.
Level meter settings – Peak level hold time	Adjusts the time that a registered peak level with a width of 1 sample stays unchanged on the level meter.

Global Plug-In Settings



The global plug-in settings can be accessed via the Settings window. The following parameters are available:

Parameter	Description
Mouse Wheel Precision	Affects the precision of the mouse wheel. The higher the precision, the finer the value changes using the mouse wheel.
SHIFT Key Precision	Affects the precision when using the [Shift] key and dragging a control with the mouse.
Drag Precision	Affects how quickly knobs and readouts react to mouse movements.

Parameter	Description
Visual settings	You can customize the look of the plug-in with the following settings: Flat Panels – When enabled, all buttons and panels of the plug-in look flat, without a gradient fill. Spotlight – Enables a wide light area that looks like a spotlight. Textures – Adds texture to the plug-in panel. Shadows – Enables shadows on graphical elements. Flat Level Meters – Enables the flat, non-blocky look of the level meters.
Show Knob Labels	Enables numeric labels that appear when you point the mouse at a knob.
Tablet Mode	When activated, you can control the plug-in with a pen tablet.
Show Virtual Keyboard	When this is activated, a virtual computer keyboard is shown when you enter values. The virtual computer keyboard is useful if the host application blocks certain keys from reaching the plug-in's user interface.
Radial Knob Mode	When this is activated, you can click on the corona to set the parameter value immediately.
Control Surface Crosshair	Displays a crosshair cursor in the control surface area.
Anti-Denormal Noise	Enables insertion of anti-denormalization noise on the plug-in inputs. This noise has an RMS value of -220dB – well below the audible dynamic range. If you are using the plug-in in a host application that applies such noise automatically, you can deactivate this option to save CPU power. Without anti-denormalization noise the filters of the plug-in can overload the CPU when silence is processed.
Remember Window Positions	When this is activated, the relative position of the plug-in windows is remembered after reopening the plug-in.
Show All Filter Shapes	When this is activated, all active filters are shown together with the shape of the selected filter.
Do Not Show Latency Changes	Disables the "Latency Changed" warning message completely.

Standard Control Elements in Detail

Knob

Knobs can be controlled as follows:

- If "Radial Knob Mode" is activated, you can drag the corona of a knob to adjust the value of the corresponding parameter. During dragging, you can move the mouse pointer away from the knob to increase value adjustment precision.
- Drag the center of a knob to adjust the value of the parameter with up and down mouse movements, linearly. If you press the left and right mouse buttons together while dragging the center, you enter high-precision adjustment mode. You can also enter this mode by holding down [Shift] when dragging. The dragging precision can be adjusted in the global settings window, see "Global Plug-In Settings" on page 57.
- Turn the mouse wheel to adjust the parameter.
- Double-click a knob to reset it to the default state.

When you point the mouse at a knob, an additional ring shows approximate parameter values at different knob positions. These values are also referred to as knob labels. Thousands are suffixed with an asterisk (2*). This ring can be disabled in the global settings window.

Keyboard value entry

Most readout values such as gain or frequency can be clicked to enter a new value.

Value list selector

This type of control allows you to choose a value or an option from the list. You can click the selector button to display the value list. You can also use the mouse's forward and backward buttons or the mouse wheel to scroll through the values of a list without opening it.

To reset a value list to its default value, right-click the selector.

Slider

Sliders can be dragged with the left mouse button. If you press the left and right mouse buttons together while dragging the slider, you enter high-precision adjustment mode. You can also enter this mode by holding down [Shift] when dragging.

Location of CurveEQ Files

CurveEQ creates settings files, including presets. All CurveEQ settings and presets are available to the specific user of the computer only.

On Windows systems, the files reside in the following folder: "\Users\<user name>\Application Data\Voxengo\Audio Plug-Ins\".

On Mac OS X systems, the files reside in the following folder: "/Users/<user name>/Library/Preferences/Voxengo/Audio Plug-Ins/".

You can safely remove, copy and replace these files, including the whole "Voxengo\Audio Plug-Ins\" subfolder.

Filter Plug-ins

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

DualFilter

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–

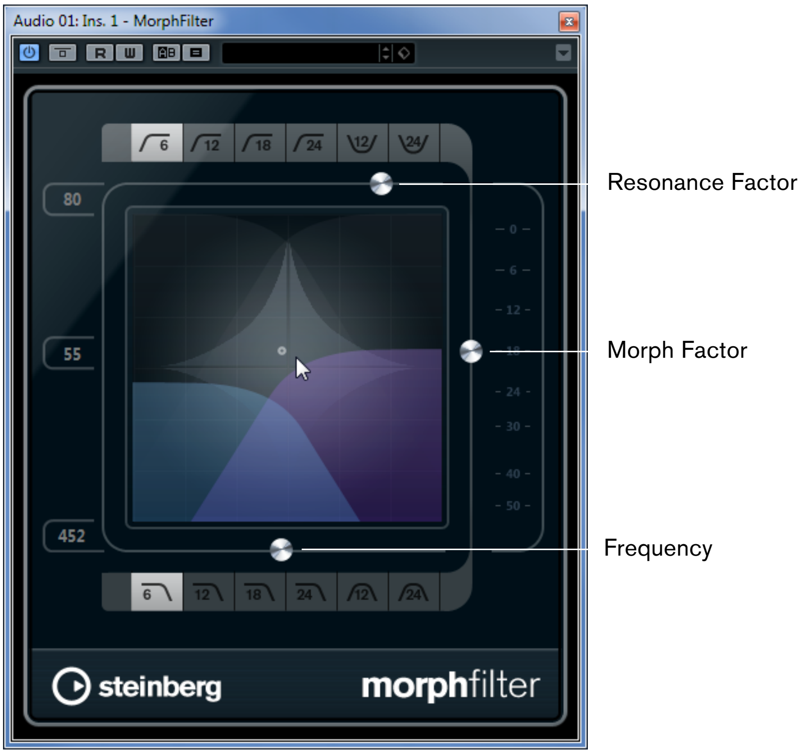
The DualFilter effect filters out certain frequencies while allowing others to pass through.

The following parameters are available:

Parameter	Description
Position	Sets the filter cutoff frequency. If you set this to a negative value, DualFilter acts 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.

MorphFilter

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



MorphFilter lets you mix low-pass, high-pass, band-pass and band-reduction filter effects, allowing for creative morphings between two filters.

The following parameters are available:

Parameter	Description
High Pass (6, 12, 18, 24 dB/per Decade)	Eliminates low-frequency signal components. Several filter slopes are available.
Band Rejection (12, 24 dB/per Decade)	Lets all frequencies pass, except those in the stop band. Several filter slopes are available.
Level meter	Shows the output level, giving you an indication of how the filtering affects the overall level of the edited event.
Resonance Factor	Changes the resonance value of the filters.
Morph Factor	Lets you mix the output between the two selected filters.
Frequency	Adjusts the cutoff frequency of the filters.
x/y control	Adjusts the Morph Factor and the Frequency parameters simultaneously.

Parameter	Description
Low Pass (6, 12, 18, 24 dB/per Decade)	Eliminates high-frequency signal components. Several filter slopes are available.
Band Pass (12, 24 dB/per Decade)	Allows signals falling within a certain frequency range to pass through. Several filter slopes are available.

PostFilter

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	–	X	–

PostFilter is the filter plug-in to use if you are working on a post-production mix, but of course you can use it in music production, too, as an alternative to complex EQ configurations. It allows quick and easy filtering of unwanted frequencies, creating room for the important sounds in your mix.

The PostFilter plug-in combines a low-cut filter, a notch filter and a high-cut filter. You can make settings by dragging the curve points in the graphical display, or by adjusting one of the controls below the display section.

Use the Preview buttons to listen to the result of your filtering.

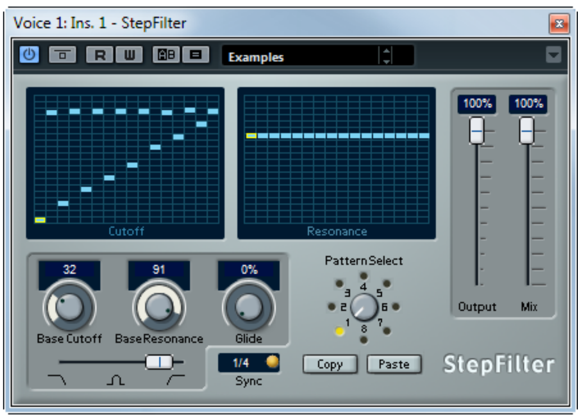
The following parameters are available:

Parameter	Description
Level meter	Shows the output level, giving you an indication of how the filtering affects the overall level of the edited event.
Low Cut Freq (20 to 1 kHz, or Off)	Use this low-cut filter to eliminate low-frequency noise. The filter is off when the curve point is moved all the way to the left.
Low Cut Slope pop-up menu	Allows you to choose a slope value for the low-cut filter.

Parameter	Description
Low Cut Preview button	This button is located between the “Low Cut Freq” button and the graphical display. Use it to switch the filter to a complementary high-cut filter. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.
Spectrum	Shows the spectrum before and after filtering.
Notch Freq	Sets the frequency of the notch filter.
Notch Gain	Adjusts the gain of the selected frequency. Use positive values to identify the frequencies that you want to filter out.
Notch Gain Invert button	Inverts the gain value of the notch filter. Use this button to filter out unwanted noise. When looking for the frequency to omit, it sometimes helps to boost it first (set the notch filter to positive gain). After you have found it, you can use the Invert button to cancel it out.
Notch Q-Factor	Sets the width of the notch filter.
Notch Preview button	This button is located between the notch filter buttons and the graphical display. Use it to create a band-pass filter with the peak filter’s frequency and Q. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.
Notches buttons (1, 2, 4, 8)	These buttons add additional notch filters to filter out harmonics.
High Cut Freq (3 to 20 kHz, or Off)	Use this high-cut filter to eliminate high-frequency noise. The filter is off when the curve point is moved all the way to the right.
High Cut Slope pop-up menu	Allows you to choose a slope value for the high-cut filter.
High Cut Preview button	This button is located between the High Cut Freq button and the graphical display. Use it to switch the filter to a complementary low-cut filter. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.

StepFilter

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–



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 are set at the pointer position.
- The horizontal axis shows the pattern steps 1 to 16 from left to right, and the vertical axis determines the (relative) filter cutoff frequency and resonance settings.

  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.

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 settings are saved together in the 8 pattern slots.
- Use the Pattern Selector below the Resonance grid to select a new pattern.

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

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 slot, which is useful for creating variations on a pattern.

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

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

StepFilter Parameters

The following parameters are available:

Parameter	Description
Base Cutoff	Sets the base filter cutoff frequency. Values set in the Cutoff grid are relative to the Base Cutoff value.
Base Resonance	Sets the base filter resonance. Values set in the Resonance grid are relative to the Base Resonance value. Note that very high Base Resonance settings can produce loud ringing effects at certain frequencies.
Glide	Applies glide between the pattern step values, causing values to change more smoothly.
Filter mode	Use this slider to select a filter mode: low-pass (LP), band-pass (BP), or high-pass (HP) (from left to right).
Sync button	When the Sync button to the right of the Sync pop-up menu is activated (yellow), the pattern playback is synchronized with the project tempo.
Sync pop-up menu (1/1 to 1/32, straight, triplet, or dotted)	Use this pop-up menu to set the pattern beat resolution, that is, the note values the pattern plays in relation to the tempo.

Parameter	Description
Output slider	Sets the overall volume.
Mix slider	Adjusts the mix between dry and processed signal.

ToneBooster

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–

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 10), greatly enhancing the tonal varieties available.

The following parameters are available:

Parameter	Description
Tone	Sets the center filter frequency.
Gain	Adjusts the gain of the selected frequency range by up to 24 dB.
Width	Sets the resonance of the filter.
Mode selector	Sets the basic operational mode of the filter; Peak or Band Mode.

WahWah

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

WahWah is a variable slope band-pass filter that can be auto-controlled by a side-chain signal (if supported, see table above) or via MIDI modeling the well-known analog pedal effect. 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 lies at 50.

The following parameters are available:

Parameter	Description
Pedal	Controls the filter frequency sweep.
Pedal Control (MIDI) pop-up menu	Allows you to choose the MIDI controller that controls the plug-in. Set this to “Automation” if you do not want to use MIDI realtime control.
Freq Lo/Hi	Set the frequency of the filter for the Lo and Hi Pedal positions.
Width Lo/Hi	Set the width (resonance) of the filter for the Lo and Hi Pedal positions.
Gain Lo/Hi	Set the gain of the filter for the Lo and Hi Pedal positions.
Filter Slope selector	Allows you to choose between two filter slope values: 6dB or 12dB.

⇨ If side-chaining is supported, the Pedal parameter can also be controlled from another signal source via the side-chain input. The louder the signal, the more the filter frequency (Pedal) is raised so that the plug-in acts as an auto-wha effect. For a description of how to set up side-chain routing, see the Operation Manual.

MIDI Control

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

- Whenever WahWah has been added as an insert effect (for an audio track or an FX channel), it is available on the Output Routing pop-up menu for MIDI tracks.

  If WahWah is selected on the Output Routing menu, MIDI data is directed to the plug-in from the selected track.

Mastering Plug-ins

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

UV22HR

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–

The UV22HR is a dithering plug-in, based on an advanced algorithm developed by Apogee. For an introduction to the concept of dithering, see the Operation Manual.

The following parameters are available:

Option	Description
Bit Resolution	The UV22HR supports dithering to multiple resolutions: 8, 16, 20 or 24 bits. Select the resolution by clicking the corresponding button.
Hi	Try this first, it is the most all-round setting.

Option	Description
Lo	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.

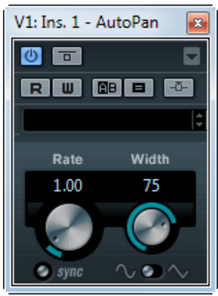
⚠ Dithering should always be applied post-fader on an output bus.

Modulation Plug-ins

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

AutoPan

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–



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

The following parameters are available:

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.
Sync button	The button below the Rate knob switches tempo sync on or off.
Width	Sets the depth of the auto-pan effect.
Waveform Shape selector	Allows you to select the modulation waveform. A sine and a triangle waveform are available.

⇨ If side-chaining is supported, the Width parameter can also be controlled from another signal source via the side-chain input. For a description of how to set up side-chain routing, see the Operation Manual.

Chopper

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–

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 following parameters are available:

Parameter	Description
Waveform buttons	Set the modulation waveform.
Depth	Sets the depth of the Chopper effect. This can also be set by clicking in the graphical 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.
Sync button	The button above the Speed knob switches tempo sync on (button lights up) or off.
Stereo/Mono button	Determines whether the Chopper works as an auto-panner (button set to “Stereo”) or a tremolo effect (button set to “Mono”).
Mix	Sets the level balance between the dry and the wet signal. If Chopper is used as a send effect, this should be set to the maximum value.

Chorus

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

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 77).

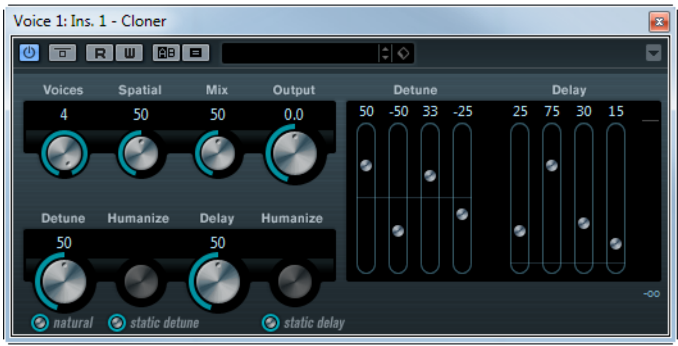
The following parameters are available:

Parameter	Description
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.
Sync button	The button below the Rate knob switches tempo sync on or off.
Width	Determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Waveform Shape selector	Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.
Spatial	Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
Mix	Sets the level balance between the dry and the wet signal. If Chorus is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Delay	Affects the frequency range of the modulation sweep by adjusting the initial delay time.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

⇨ If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal’s envelope. For a description of how to set up side-chain routing, see the Operation Manual.

Cloner

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–



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

The following parameters are available:

Parameter	Description
Voices	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	Spreads the added voices across the stereo spectrum. Turn clockwise for a deeper stereo effect.
Mix	Sets the level balance between the dry and the wet signal. If Cloner is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Output	Allows you to reduce or increase the output gain by up to 12 dB.
Detune slider 1 to 4	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 to 4	Controls the relative delay amount for each voice. A value of zero means no delay for that voice.
Detune	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.
Natural button	By clicking the Natural button below the Detune knob, you can change the pitch algorithm.
Detune – Humanize	Controls the amount of detune variation when Static Detune is deactivated. With Humanize, the detune is constantly modulated for a more natural effect. The value range is from 0 to 100 (strongest detune variation).
Static Detune button	Use this button to activate/deactivate the Static Detune function. If activated, the set detune amount is static, and the Humanize knob is grayed out.
Delay	Governs the overall depth of the delay for all voices. If set to zero, no delay takes place regardless of the Delay slider settings.

Parameter	Description
Delay – Humanize	Controls the amount of delay variation when Static Delay is deactivated. With Humanize, the delay is constantly modulated for a more natural effect. The value range is from 0 to 100 (strongest delay variation).
Static Delay button	Use this button to activate/deactivate the Static Delay function. If activated, the set delay amount is static, and the Humanize knob is grayed out.

Flanger

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

Flanger is a classic flanger effect with added stereo enhancement.

The following parameters are available:

Parameter	Description
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.
Sync button	The button below the Rate knob switches tempo sync on or off.
Range Lo/Hi	Set the frequency boundaries for the flanger sweep.
Feedback	Determines the character of the flanger effect. Higher settings produce a more metallic sounding sweep.
Spatial	Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
Mix	Sets the level balance between the dry and the wet signal. If Flanger is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Waveform Shape selector	Allows you to select the modulation waveform, altering the character of the flanger sweep. A sine and a triangle waveform are available.
Delay	Affects the frequency range of the modulation sweep by adjusting the initial delay time.

Parameter	Description
Manual knob	Allows you to change the sweep position manually when the Manual button is deactivated. The value range is from 0 to 100.
Manual button	Use this button to activate/deactivate the Manual function. If activated, the flanger sweep is static, that is, no modulation takes place.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

⇨ If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the Operation Manual.

Metalizer

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–

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

The following parameters are available:

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 a sharper sound and a more pronounced effect.
Tone	Governs the feedback frequency. The effect of this is more noticeable with high Feedback settings.
On button	Turns filter modulation on and off. When turned off, Metalizer works as a static filter.
Mono button	When this is activated, the output of Metalizer is 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.
Sync button	The button above the Speed knob switches tempo sync on (button lights up) or off.

Parameter	Description
Output slider	Sets the overall volume.
Mix slider	Sets the level balance between the dry and the wet signal. If Metalizer is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

Phaser

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

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

The following parameters are available:

Parameter	Description
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.
Sync button	The button below the Rate knob switches tempo sync on or off.
Width	Determines the width of the modulation effect between higher and lower frequencies.
Feedback	Determines the character of the phaser effect. Higher settings produce a more pronounced effect.
Spatial	When using multi-channel audio, the Spatial parameter creates a 3-dimensional impression by delaying modulation in each channel.
Mix	Sets the level balance between the dry and the wet signal. If Phaser is used as a send effect, set this to the maximum level as you can control the dry/effect balance with the send.
Manual knob	Allows you to change the sweep position manually when the Manual button is deactivated. The value range is from 0 to 100.

Parameter	Description
Manual button	Use this button to activate/deactivate the Manual function. If activated, the flanger sweep is static, that is, no modulation takes place.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

⇨ If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the Operation Manual.

RingModulator

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–

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.

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

The following parameters are available:

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 decreases the oscillator pitch, whereas right of center the oscillator pitch increases when fed a loud input.
Oscillator – Waveform buttons	Allows you to select the oscillator waveform; square, sine, saw, or triangle.
Oscillator – Range slider	Determines the frequency range of the oscillator in Hz.

Parameter	Description
Oscillator – Frequency	Sets the oscillator frequency ±2 octaves within the selected range.
Oscillator – Roll-Off	Attenuates high frequencies in the oscillator waveform, to soften the overall sound. This is best used when harmonically rich waveforms are selected (square or saw, for example).
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, at 0% no modulation is applied. With negative values, a loud input signal slows down the LFO, whereas positive values speed it up at loud input signals.
LFO – Waveform	Allows you to select the LFO waveform; square, sine, saw, or triangle.
LFO – Invert Stereo	Inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.
Envelope Generator section – Attack and Decay	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 controls 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 button	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 slider	Sets the overall volume.
Mix slider	Adjusts the mix between dry and processed signal.

Rotary

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–

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

The following parameters are available:

Parameter	Description
Speed selector (Stop/Slow/Fast)	Allows you to control the speed of the Rotary in three steps.
Speed Change Mode	Allows you to select whether the Slow/Fast setting is a switch (left) or a variable control (right). When switch mode is selected and Pitchbend is the controller, the speed switches with an up or down flick of the bender. Other controllers switch at MIDI value 64.
Speed Mod	When the Slow/Fast setting is set to variable control, this allows you to select the rotary speed, from 0 (Stop) to 100 (Fast).
MIDI controller pop-up menu	Allows you to choose the MIDI controller that controls the plug-in. Set this to “Automation” if you do not want to use MIDI realtime control.
Overdrive	Applies a soft overdrive or distortion.
CrossOver	Sets the crossover frequency (200 to 3000 Hz) between the low and high frequency loudspeakers.
Horn – Slow	Allows for a fine adjustment of the high rotor Slow speed.
Horn – Fast	Allows for a fine adjustment of the high rotor Fast speed.
Horn – Accel.	Allows for a fine adjustment of the high rotor acceleration time.
Horn – Amp Mod	Controls the high rotor amplitude modulation.
Horn – Freq Mod	Controls the high rotor frequency modulation.
Bass – Slow	Allows for a fine adjustment of the low rotor Slow speed.
Bass – Fast	Allows for a fine adjustment of the low rotor Fast speed.
Bass – Accel.	Allows for a fine adjustment of the low rotor acceleration time.
Bass – Amp Mod	Adjusts the modulation depth of the amplitude.
Bass – Level	Adjusts the overall bass level.
Microphones – Phase	Adjusts the phasing amount in the sound of the high rotor.
Microphones – Angle	Sets the simulated microphone angle. 0 = mono, 180 = a mic on each side.
Microphones – 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 signal.

Directing MIDI to the Rotary

For realtime MIDI control of the Speed parameter, MIDI must be directed to the Rotary.

- Whenever Rotary has been added as an insert effect (for an audio track or an FX channel), it is available on the Output Routing pop-up menu for MIDI tracks.

  If Rotary is selected on the Output Routing menu, MIDI is directed to the plug-in from the selected track.

StudioChorus

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–
Side-chain support	–	–	–	–	X	X	–

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).

For each stage the following parameters are available:

Parameter	Description
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.
Sync button	The button below the Rate knob switches tempo sync on or off.
Width	Determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Waveform Shape selector	Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.
Spatial	Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
Mix	Sets the level balance between the dry and the wet signal. If StudioChorus is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Delay	Affects the frequency range of the modulation sweep by adjusting the initial delay time.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

⇨ If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the Operation Manual.

Tranceformer

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–

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.

The following parameters are available:

Parameter	Description
Waveform buttons	Allow you to select a 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.
Sync button	The button above the Speed knob switches tempo sync on (button lights up) or off.
On button	Turns modulation of the pitch parameter on or off.
Mono button	Governs whether the output is stereo or mono.
Output slider	Adjusts the output level of the effect.
Mix slider	Sets the level balance between the dry and the wet signal.

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

Tremolo

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

Tremolo produces amplitude (volume) modulation. The following parameters are available:

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 modulation speed can be set freely with the Rate knob.
Sync button	The button below the Rate knob switches tempo sync on or off.
Depth	Governs the depth of the amplitude modulation.
Spatial	Adds a stereo effect to the modulation.
Output	Adjusts the output volume.

⇨ If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the Operation Manual.

Vibrato

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–
Side-chain support	–	–	–	X	X	X	–

The Vibrato plug-in produces pitch modulation. The following parameters are available:

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 modulation speed can be set freely with the Rate knob.
Sync button	The button below the Rate knob switches tempo sync on or off.
Depth	Governs the depth of the pitch modulation.
Spatial	Adds a stereo effect to the modulation.

⇨ If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the Operation Manual.

Pitch Shift Plug-ins

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

Octaver

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–

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 following parameters are available:

Parameter	Description
Direct	Adjusts the mix of the original signal and the generated voices. 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	Adjusts the level of the signal that is generated one octave below the original pitch. A setting of 0 means that the voice is muted.
Octave 2	Adjusts the level of the signal that is generated two octaves below the original pitch. A setting of 0 means that the voice is muted.

Pitch Correct

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–

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

Furthermore, you can use Pitch Correct creatively. You can create backing vocals, for example, by modifying the lead vocals or vocoder sounds by using extreme values. You can use an external MIDI controller, a MIDI 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 is 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 is displayed in blue while the changes are displayed in orange.

The following parameters are available:

Parameter	Description
Correction – Speed	Determines 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 (for example, the famous Cher effect).
Correction – Tolerance	Determines the sensitivity of analysis. A low Tolerance value lets Pitch Correct find pitch changes quickly. When the Tolerance value is high, pitch variations in the audio (for example, vibrato) are not immediately interpreted as note changes.
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 that the signal is not transposed.

Parameter	Description
Scale Source – Internal	If you choose the Internal option from the Scale Source pop-up menu, you can use the pop-up menu next to it to decide to which scale the source audio is adapted. The following options are available: Chromatic: The audio is pitched to the closest semitone. Major/Minor: The audio is pitched to the major/minor scale specified in the pop-up menu to the right. This is reflected on the keyboard display. Custom: The audio is pitched to the notes that you specify by clicking keys on the keyboard display. To reset the keyboard, click the orange line below the display.
Scale Source – External MIDI Scale	Select this option if you want the audio to be shifted to a scale of target pitches, using an external MIDI controller, the Virtual Keyboard or a MIDI track. Note that you have to assign the audio track as the output of your MIDI track and that the Speed parameter has to be set to a value other than Off.
Scale Source – External MIDI Note	Select this option if you want the audio to be shifted to a target note, using an external MIDI controller, the Virtual Keyboard or a MIDI track. Note that you have to assign the audio track as the output of your MIDI track and that the Speed parameter has to be set to a value other than Off.
Formant – Shift (-60 to 60)	Changes the natural timbre, that is, the characteristic frequency components of the source audio.
Formant – Optimize (General, Male, Female)	Allows you to specify the sound characteristics of the sound sources. While General is the default setting, Male is designed for low pitches and Female for high pitches.
Formant – Preservation (On/Off)	When this parameter is set to Off, formants are raised and lowered with the pitch, provoking strange vocal effects. Higher pitch correction values result in Mickey Mouse effects, lower pitch correction values in Monster sounds. When this parameter is set to On, the formants are kept, maintaining the character of the audio.
Master Tuning	Detunes the output signal. The default setting is 440 Hz.

PitchDriver

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	–	X	–

PitchDriver was created for sound design purposes in postproduction. This plug-in can be used for extreme up or down pitching of voices or effect samples (for example, to create eerie monster sounds). Shifting the pitch with this plug-in does not keep the formants.

The following parameters are available:

Parameter	Description
Detune	Lets you detune the pitch of the incoming audio. Positive and negative values can be set.
Mix	Sets the level balance between the dry and the wet signal.
Spatial	Creates an ambience effect. It introduces a light pitch offset to the incoming signal. Different offset values are used for the individual input channels in order to create a panorama effect. Note that the created panorama effect can be unstable. For a stable panorama, turn off the Spatial parameter. In that case the incoming signals are summed up to a mono signal.
Output	Adjusts the output volume.

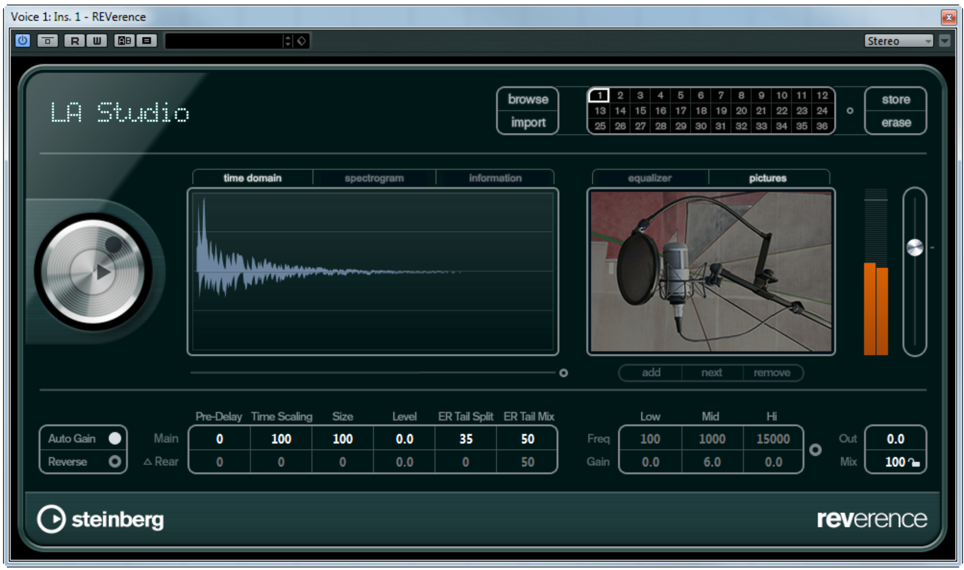
⇨ To avoid hearing artifacts, it is recommended to set the ASIO buffer for your audio card to at least 128 samples. The buffer size can be set on the card driver's control panel (opened via the Device Setup dialog).

Reverb Plug-ins

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

REVerence

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

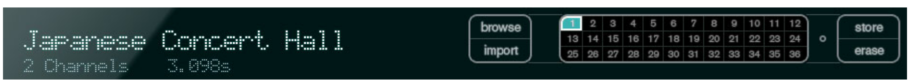


REVerence is a convolution tool that allows you to apply room characteristics (reverb) to the audio. This is done by processing the audio signal according to an impulse response – a recording of an impulse in a room or another location that recreates the characteristics of the room. As a result, the processed audio sounds 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 can be very demanding in terms of RAM. This is because the impulse responses that you load into the program slots are preloaded into RAM to guarantee artifact-free switching between programs. Therefore, you should always load only those programs that you need for a given task.

Using the Program Matrix

A program is the combination of an impulse response and its settings. These include reverb settings (see “Changing the Reverb Settings” on page 87), EQ settings (see “Making EQ Settings” on page 89), pictures (see “Loading Pictures” on page 90), and output settings (see “Making Output Settings” on page 90). The program matrix allows you to load programs and to view the name of the current program, that is, the impulse response (see “Working with Custom Impulse Responses” on page 91).



The following parameters are available:

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 is shown. After loading an impulse response, its number of channels and the length in seconds are displayed for a few seconds.
Browse button	This button opens a browser window showing the available programs. When you select a program in the browser, it is loaded into the active program slot. To be able to filter the list of impulse responses in the browser window, for example, by room type or the number of channels, you can activate the Filters section (by clicking the “Set Up Window Layout” button at the bottom left of the window).
Import button	Click this button to load your own impulse response files from disk. The files should have a maximum length of 10 seconds. Longer files are automatically cut. For more information, see “Working with Custom Impulse Responses” on page 91.
Program slots (1 to 36)	Use these slots to load all the impulse responses (programs) that you want to work with in a session. The selected program slot is indicated by a white frame. Used slots are shown in a different color. Double-clicking an empty program slot opens a browser window, showing the available programs. Clicking a used program slot recalls the corresponding program and loads it into REVerence. When you move the mouse over a used slot, the corresponding program name is displayed below the active program name.
Smooth Parameter Changes button	The “Smooth Parameter Changes” button is located between the program slots and the Store/Erase buttons. If it is activated, a crossfade is performed when switching programs. Leave this button deactivated while looking for a suitable program or an appropriate setting for an impulse response. Once you have set up the program matrix to your liking, activate the button to avoid hearing artifacts when switching between programs.
Store button	Stores the active impulse response and its settings as a program.
Erase button	Removes the selected program from the matrix.

Programs vs. Presets

You can save your REVerence settings as VST plug-in presets or programs.

Both presets and programs use the file extension .vstpreset and appear in the same category in the MediaBay (Plug-In Presets), but they are represented by different icons:

Icon	Description
	A REVerence preset contains all settings and parameters for the plug-in, that is, all the loaded impulse responses along with their parameter settings and positions in the program matrix. Presets are loaded via the Presets pop-up menu at the top of the plug-in panel.
	A REVerence program only contains the settings related to a single impulse response. Programs are loaded and managed via the program matrix.

Presets

Presets are useful in the following situations:

- When you want to save a complete setup with different impulse responses for later use (for example, different setups for explosion sounds that can be reused for other scenes or movies).
- When you want to save different parameter sets for the same impulse response so that you can later choose the set that best suits your needs.

Programs

Programs offer the following advantages:

- Up to 36 programs can be loaded into the program matrix for instant recall.
- A program provides a quick and easy way to save and recall a subset of the plug-in parameters (that is, the settings for a single impulse response), allowing for short loading times.
- When automating a project and loading a REVerence program, only one automation event is written.

  If you load a plug-in preset instead (which contains a lot more settings than a program), a lot of unnecessary automation data (for the settings that you did not use) is written.

Setting up Programs

Proceed as follows:

1. In the program matrix, click on a program slot to select it.

   A blinking white frame indicates that this program slot is selected.
2. Click the Browse button or click the empty slot again to load one of the included programs.

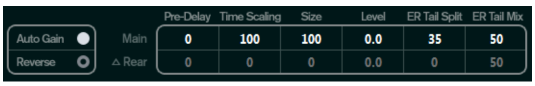
   You can also import a new impulse response file, see “Importing Impulse Responses” on page 91.
3. In the browser that appears, select the program containing the impulse response that you want to use and click OK.

   The name of the loaded impulse response is shown in the upper left corner of the REVerence panel.
4. Set up the REVerence parameters as needed and click the Store button to save the impulse response with the current settings as a new program.
5. Set up as many programs as you need (up to 36) by following the steps above.

⇨ If you want to use your set of programs in other projects, save your settings as a plug-in preset using the Presets pop-up menu at the top of the plug-in panel.

Changing the Reverb Settings

The reverb settings allow you to change the characteristics of the room.



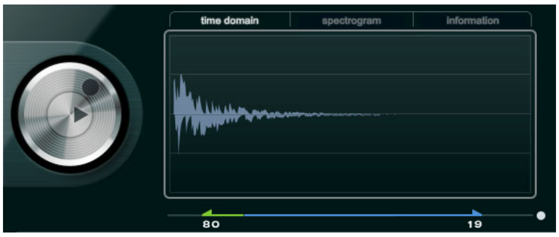
The following parameters are available:

Parameter	Description
Main	All values shown in the top row are for all speakers except the LFE.
Rear Δ	If you are working with surround tracks up to 5.1, you can use this row to set up an offset for the rear channels.

Parameter	Description
Auto Gain button	When this button is activated, the impulse response is automatically normalized.
Reverse button	Reverses the impulse response.
Pre-Delay	Controls the amount of time between the dry signal and the onset of the reverb. With higher pre-delay values you can simulate larger rooms.
Time Scaling	Controls the reverb time.
Size	Determines the size of the simulated room.
Level	A level control for the impulse response. This governs the volume of the reverb.
ER Tail Split	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 are 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 attenuate the tail.

The Impulse Response Display

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



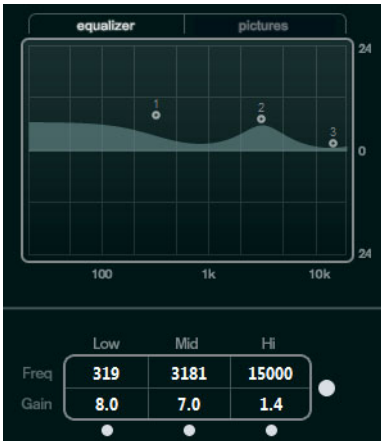
The following parameters are available:

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 display	Shows the waveform of the impulse response.
Spectrogram display	Shows the analyzed spectrum of the impulse response. Time is displayed along the horizontal axis, frequency along the vertical axis, and volume is represented by the color.
Information display	Shows additional information, for example, the name of the program and the loaded impulse response, the number of channels, the length, and Broadcast Wave File information.

Parameter	Description
Activate Impulse Trimming button	Use this button at the bottom right of the Impulse display section to activate impulse trimming. The Trim slider is shown below the Impulse display.
Trim slider	Allows you to trim the start and end of the impulse response. Drag the front handle to trim the start 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 is cut without any fading.

Making EQ Settings

In the Equalizer section you can tune the sound of the reverb.

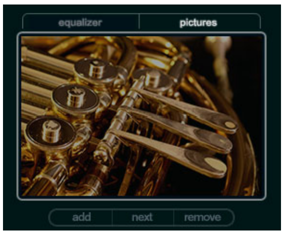


The following parameters are available:

Parameter	Description
EQ curve display	Shows the EQ curve. You can use the EQ parameters below the display to change the EQ curve, or modify the curve manually by dragging the curve points.
Activate EQ button	This button to the right of the EQ parameters activates the EQ for the effect plug-in.
Low Shelf On button	Activates the low shelf filter that boosts or attenuates frequencies below the cutoff frequency by the specified amount.
Low Freq (20 to 500)	Sets the frequency of the low band.
Low Gain (-24 to +24)	Sets the amount of attenuation/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)	Sets the center frequency of the mid band.
Mid Gain (-12 to +12)	Sets the amount of attenuation/boost for the mid band.
Hi Shelf On button	Activates the high shelf filter that boosts or attenuates frequencies above the cutoff frequency by the specified amount.
Hi Freq (5000 to 20000)	Sets the frequency of the high band.
Hi Gain (-24 to +24)	Sets the amount of attenuation/boost for the high band.

Loading Pictures

In the Pictures section you can load graphics files to illustrate the setting, for example, the recording location or microphone arrangement of the loaded impulse response. Up to five pictures can be loaded.



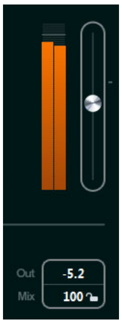
The following parameters are available:

Parameter	Description
Add button	Opens a file dialog where you can navigate to the graphics file that you want to import. JPG, GIF, and PNG file formats are supported.
Next button	If several pictures are loaded, you can click this button to display the next image.
Remove button	Deletes the active picture. Note that this does not remove the graphics file from your hard disk.

⇨ Pictures are only referenced by the plug-in and are not copied to the project folder.

Making Output Settings

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



The following parameters are available:

Parameter	Description
Output activity meter	Indicates the overall level of the impulse response and its settings.
Output slider	Adjusts 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.

- To lock the dry/wet balance while browsing through the available presets and programs, activate the Lock button (padlock symbol) next to the Mix parameter.

Working with Custom Impulse Responses

In addition to working with the impulse responses included with REVerence, you can import your own impulse responses and save these as programs or presets. WAVE and AIFF files with a mono, stereo, true-stereo, or multi-channel (up to 5.0) configuration are supported. If a multi-channel file contains an LFE channel, this channel is ignored.

REVerence uses the same channel width as the track it is inserted on. When importing impulse response files with more channels than the corresponding track, the plug-in only reads as many channels as needed. If the impulse response file contains fewer channels than the track, REVerence generates the missing channels (for example, the center channel as a sum of the left and right channels). If the rear channels are missing (when importing a stereo response file onto a 4.0 track, for example), the left and right channels are also used for the rear channels. In this case you can use the Rear offset parameter to create more spatiality.

Importing Impulse Responses

To import impulse responses, proceed as follows:

1. In the program matrix, click the Import button.
2. Navigate to the file that you want to import, and click Open.
   The file is loaded into REVerence. The channels from an interleaved file are imported in the same order as in other areas of the program (for example, the VST Connections window), see below.
3. Make the appropriate settings and add a picture, if available.
   Pictures residing in the same folder as the impulse response file or in the parent folder are automatically found and displayed.
4. Click the Store button to save the impulse response and its settings as a program. That way you can recall the setup at any time.
   The program slot turns blue, indicating that a program is loaded.

⇨ When saving a program, the impulse response file itself is only referenced. It still resides in the same location as before and is not modified in any way.

5. Repeat these steps for any impulse response files that you want to work with.

REVerence reads input channels in the following order:

No. of input channels	Channel order in REVerence
1	L
2	L/R
3	L/R/C
4	L/R/LS/RS (if inserted on a track with a 4.0 channel configuration, see below)
4	LL/LR/RL/RR (if inserted on a track with a stereo configuration, see below)
5	L/R/C/LS/RS
6	L/R/C/LFE/LS/RS (LFE is being ignored.)

True Stereo

Impulse responses recorded as trues-stereo files enable you to create a very realistic impression of the corresponding room. REVerence can only process true-stereo impulse response files with the following channel configuration (in exactly that order): LL, LR, RL, RR.

The channels are defined as follows:

Channel	The signal from this source…	…was recorded with this microphone
LL	left source	left microphone
LR	left source	right microphone
RL	right source	left microphone
RR	right source	right microphone

⇨ If your true-stereo impulse responses are only available as separate mono files, you can use the Export Audio Mixdown function to create REVerence compliant interleaved files (see the Operation Manual).

By default, REVerence automatically works in true-stereo mode when the plug-in is inserted on a stereo track and you load a 4-channel impulse response.

Therefore, if you are working with surround files, that is, 4-channel impulse responses recorded with a Quadro configuration (L/R, LS/RS), you need to insert the plug-in on an audio track with a 4.0 configuration. On a stereo track these files would be processed in true-stereo mode, too.

So how can you prevent REVerence from unintentionally processing surround files in true-stereo mode? The answer is a “Recording Method” attribute that can be written to the iXML chunk of the corresponding impulse response file. Whenever you load an impulse response with a 4-channel configuration on a stereo track, REVerence searches the iXML chunk of the file. If the plug-in finds the Recording Method attribute, the following happens:

- If the attribute is set to “TrueStereo”, the plug-in works in true-stereo mode.
- If the attribute is set to “A/B” or “Quadro”, the plug-in works in normal stereo mode and processes only the L/R channels of the surround file.

⇨ You can use the Attribute Inspector in the MediaBay to tag your own impulse response files with the Recording Method attribute. For more information, see the Operation Manual.

Relocating Content

Once you have imported your own impulse responses in REVerence you can comfortably work with them on your computer. But what if you need to transfer your content to another computer, for example because you work sometimes with a PC and sometimes with a notebook, or you need to hand over a project to a colleague in the studio?

The factory content is not a problem since it is also present on the other computer. For these impulse responses you just need to transfer your REVerence programs and presets to be able to access your setups.

User content is a different matter, though. If you have transferred your audio files to an external drive or a different hard disk location on the other computer, REVerence cannot access the impulse responses any more since the old file paths have become invalid.

To access your impulse responses again, proceed as follows:

1. Transfer you audio files to a location that you can access from the second computer (for example, an external hard disk).

   If you keep the files in the same folder structure as on the first computer, REVerence automatically finds all files contained in this structure.

2. Transfer any REVerence presets or programs that you need to the second computer.

   If you are unsure where the presets need to be stored, you can find the paths in the MediaBay (see the Operation Manual).

3. Open REVerence on the second computer and try to load the preset or program that you want to work with.

   The Locate Impulse Response dialog opens.

4. Navigate to the folder that contains your impulse responses. Click Open.

   REVerence is now able to access all the impulse responses stored in this location.

⚠ The new path to these audio files has not been saved yet. To make the files permanently available without having to use the Locate dialog, you need to save your programs or presets under a different name.

RoomWorks

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

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.

The following parameters are available:

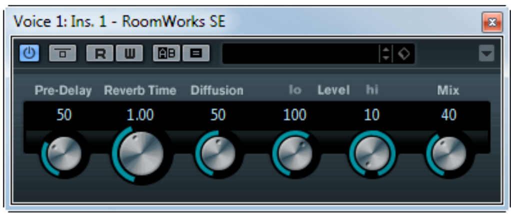
Parameter	**Description**
Input – Lo Freq	Determines the frequency at which the low-shelving filter takes effect. Both the high and low settings filter the input signal prior to reverb processing.
Input – Hi Freq	Determines the frequency at which the high-shelving filter takes effect. Both the high and low settings filter the input signal prior to reverb processing.
Input – Lo Gain	Controls the amount of boost or attenuation for the low-shelving filter.
Input – Hi Gain	Controls the amount of boost or attenuation for the high-shelving filter.

Parameter	Description
Reverb – Pre-Delay	Controls how much time passes before the reverb is applied. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.
Reverb – Reverb Time	Allows you to set the reverb time in seconds.
Reverb – Size	Alters the delay times of early reflections to simulate larger or smaller spaces.
Reverb – Diffusion	Affects the character of the reverb tail. Higher values lead to more diffusion and a smoother sound, while lower values lead to a clearer sound.
Reverb – Width	Controls the width of the stereo image. 100% gives you full stereo reverb. At 0%, the reverb is all in mono.
Reverb – Variation button	Clicking this button generates 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 often solves these issues. There are 1000 possible variations.
Reverb – Hold button	Clicking this button freezes the reverb buffer in an infinite loop (yellow circle around button). You can create some interesting pad sounds using this feature.
Damping – Lo Freq	Determines the frequency below which low-frequency damping occurs.
Damping – High Freq	Determines the frequency above which high-frequency damping occurs.
Damping – Low Level	Affects the decay time of low frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes low frequencies to decay quicker. Values above 100% cause low frequencies to decay more slowly than the mid-range frequencies.
Damping – High Level	Affects the decay time of high frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes high frequencies to decay quicker. Values above 100% cause high frequencies to decay more slowly than the mid-range frequencies.
Envelope – Amount	Determines how much the envelope attack and release controls affect the reverb itself. Lower values have a more subtle effect while higher values lead to a more drastic sound.
Envelope – Attack	The envelope settings in RoomWorks control how the reverb follows the dynamics of the input signal in a fashion similar to a noise gate or downward expander. Attack determines how long it takes for the reverb to reach full volume after a signal peak (in milliseconds). This is similar to a pre-delay but the reverb is ramping up instead of starting all at once.
Envelope – Release	Determines how long after a signal peak the reverb can be heard before being cut off, similar to a gate’s release time.
Surround – 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.

Parameter	Description
Surround – Rotate button	This button is only available for surround configurations. When active, the perspective of the room is shifted 90°.
Surround – 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. When the Rotate option is activated, these relationships shift 90°.
Output – Mix	Determines the balance of dry (unprocessed) and wet (processed) signal. When RoomWorks is used as an insert for an FX channel, you most likely want to set this to 100% or use the wet only button.
Output – Wet only button	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 for an FX or group channel.
Output – Efficiency	Determines how much processing power is used for RoomWorks. The lower the value, the more CPU resources are used, and the higher the quality of the reverb. Interesting effects can be created with very high Efficiency settings (>90%). Experiment for yourself.
Output – Export button	Determines if during audio export RoomWorks uses the maximum CPU power for the highest quality reverb. During export you may wish to keep a higher efficiency setting to achieve a specific effect. If you want the highest quality reverb during export, make sure this button is activated.
Output – Output meter	Indicates the level of the output signal.

RoomWorks SE

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	X	–



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.

The following parameters are available:

Parameter	Description
Pre-Delay	Controls how much time passes before the reverb is applied. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.
Reverb Time	Allows you to set the reverb time in seconds.

Parameter	Description
Diffusion	Affects the character of the reverb tail. Higher values lead to more diffusion and a smoother sound, while lower values lead to a clearer sound.
Hi Level	Affects the decay time of high frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes high frequencies to decay quicker. Values above 100% cause high frequencies to decay more slowly than the mid-range frequencies.
Lo Level	Affects the decay time of low frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes low frequencies to decay quicker. Values above 100% cause low frequencies to decay more slowly than the mid-range frequencies.
Mix	Determines the balance of dry (unprocessed) and wet (processed) signal. When using RoomWorks SE inserted in an FX channel, you most likely want to set this to 100% or use the wet only button.

Spatial + Panner Plug-ins

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

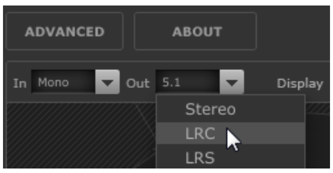
Anymix Pro

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	–	X	–

The Anymix Pro plug-in from IOSONO is a sophisticated surround panner and a powerful upmix/fold down processor that converts any given audio material into output formats ranging from mono to 8.1.

Input/Output Configuration

The input/output configuration of the plug-in can be selected from the In and Out pop-up menus in the top left corner of the plug-in panel.



If Anymix Pro is used as an insert effect, the maximum input and output configuration cannot exceed the track width of the current track.

If Anymix Pro is used as a panner, the maximum input configuration cannot exceed the track width of the current track. The maximum output configuration cannot exceed the width of the output bus that the track is currently routed to.

Channel Order

The plug-in uses the channel order of the host application unless the selected output configuration differs from the track configuration.

⚠ Choosing an output configuration that differs from the current track configuration results in channel oddities.

If the output configuration of the track is not a subset of the plug-in output configuration, for example, track = 6.1 cine and plug-in output = 7.0 music, the channels are routed as follows:

1	2	3	4	5	6	7	8	9
L	R	C	LFE	LS	RS	RSS/RC	LSS/LC	CS

⇨ Channels that are missing in the output configuration are automatically skipped.

Example:

	1	2	3	4	5	6	7	8	9
Track configuration: 6.1 Cine	L	R	C	LFE	LS	RS	CS		
Plug-in output configuration: 7.0 Music	L	R	C	LS	RS	LSS	RSS		
Result		OK			Mismatch				

Latency Compensation

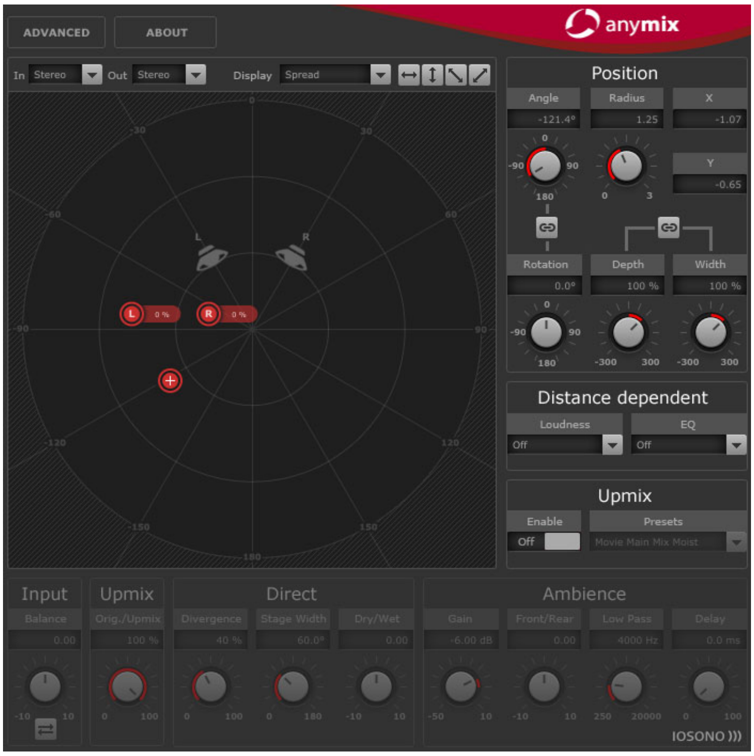
Anymix Pro causes a processing delay. The amount of latency depends on the buffer size of the audio card and the processing mode of the plug-in, that is, panning or upmix. Steinberg host applications can compensate this delay automatically.

The Plug-In Panel

The panel of Anymix Pro is divided into several sections, with the stage view taking the most space to display the position and movement of the input channels, output configuration, and distance-dependent filter values. On the right side, there are the controls for position and movement, and the bottom section of the plug-in panel contains the upmix controls.

⇨ The plug-in panel has two different display modes, panning and upmix.

Panning mode



Panning

In the stage view, input channels are represented by red icons, output channels by gray speakers in the background.

Moving the input channels outside the loudspeaker setup results in panning between the two nearest output speakers. The input channels that are placed at smaller distances are distributed to several output speakers.

- To change the position of the input group, click and drag anywhere in the stage view, or right-click in the stage view. Right-clicking causes the channels to jump to the new position.

⇨ The distance between the input channels automatically shrinks when these are moved to the border of the stage. This lets you create the illusion of depth when moving stereo or multi-channel material.

Position Section

In addition to using the stage view, the input channels can also be moved using the controls at the top right of the plug-in panel. The following options are available:

Option	Description
Angle/Radius	Control the position of the center point of the input group, relative to the center of the stage view.
X/Y	Move the center point horizontally and vertically.

Option	Description
Rotation	Rotates the input group around its center point.
Link Angle & Rotation	Changes the rotation of the input group from self-centered to stage-centered.
Depth	Scales the input group vertically.
Width	Scales the input group horizontally.
Link Depth & Width	Keeps the aspect ratio between Depth and Width scaling.

⇨ By pressing [Shift] while using the controls, you can fine-adjust all parameters.

Individual Channel Adjustment

You can change the positions of the input channels individually by double-clicking the corresponding input icon in the stage view. A separate panel with channel-specific parameters opens.

The following options are available:

Option	Description
Radius/Angle	Control the position of the selected input channel, relative to the center of the input group.
X/Y	Move the selected input channel horizontally and vertically.
Volume	Applies gain to the selected input channel.
LFE Volume	Controls the amount of LFE for the selected input channel.
Spread	Distributes the audio from the selected input channel to more than two output channels. At 0% the audio source is rendered where the channel icon is placed. At 100% the audio is evenly distributed to all speakers of the output configuration.
Manual Delay	Adds a delay to the selected input channel.
Link buttons	Activate these buttons to link the corresponding parameters in the current plug-in instance. Adjusting the value of a linked parameter changes the other linked parameters, too.

⚠ The individual input channel parameters cannot be automated from the host application, but the adjustments you make for each input channel are saved for each plug-in instance and panner in the session.

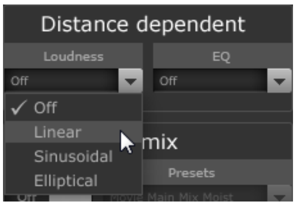
Restricting Movement

You can use the double-arrow buttons at the top right of the stage view to restrict the direction of movement of the object in the stage view to orthogonal or diagonal, for easy automation.

⇨ In most cases objects move on very simple routes around the audience. By restricting the direction of movement, you can quickly create accurate movements.

Distance-Dependent Filters

To create immersive mixes even faster, Anymix Pro is equipped with a distance-dependent filter unit that lets you adjust the volume and air damping of moving objects automatically.



The following filters are available:

Option	Description
Loudness	Lowers the volume for objects that are further away.
EQ	Dampens the high frequencies of objects that are further away.

For both of the filters you can choose an attenuation curve from the corresponding pop-up menu:

Option	Description
Off	Deactivates the distance-dependent filter.
Linear	The filtering starts right from the center point and is applied linearly. Select this curve type if even tiny movements should have an impact on the distance-dependent filter.
Sinusoidal	The filtering starts approximately at loudspeaker distance and increases exponentially with distance. Select this curve type if movements in the center circle should have no audible impact on the distance-dependent filter.
Elliptical	The filtering starts approximately at two thirds of the stage with an exponential attenuation curve. Select this curve type if only movements along the border of the stage should have an impact on the distance-dependent filter.

⇨ The current values can be shown in the speaker icon labels, using the Display pop-up menu above the stage view.

The distance-dependent filters can be further adjusted using the advanced options, see “Advanced Options” on page 103.

Upmix

The upmix feature of Anymix Pro is very useful if rearranging tracks with fewer input channels into a specific surround format is not enough.

The upmix algorithm analyzes the incoming audio signal and separates it into parts of direct sound and ambient sound. While the direct sound parts are sent to the direct sound stream and can be placed at the virtual front speaker configuration, the ambient sound parts can be modified and arranged around the virtual stage. Note that this does not add any additional information to the audio stream. All sound parts that you hear from the ambient sound were already part of the original audio material.

⚠ If your audio does not contain spatial information, there cannot be an ambient sound stream. For example, you cannot extract ambient sound from a dry recording of a narrator sitting in a recording booth.

⚠ Lossy compression, such as in MP3 files, or other deficiencies of the incoming audio cannot be remedied using the upmix mode. For example, compression artifacts can easily be misinterpreted and redistributed to the ambient sound stream.

Switching to Upmix Mode

- To switch to upmix mode, activate the Enable option in the Upmix section to the right of the stage view.

⇨ The upmix algorithm is very sophisticated and can cause a high CPU load. Therefore, you cannot automate the Enable option.

Stage View

In upmix mode, the parameters are represented by segments of a circle in the stage view.

⇨ The position parameters for the input group and any created automation are preserved when the upmix is enabled. In upmix mode, the sound image created by the upmix algorithm can be moved around the stage and is also fully automatable. The parameters that you have adjusted for a single channel have no influence on the upmix, but they are kept and automatically reloaded when the upmix is disabled.

Upmix Presets

Anymix Pro comes with a set of preconfigured upmix presets. When a preset is loaded, the upmix and advanced parameters are set accordingly and can be further adjusted.

An upmix preset contains settings for the following upmix parameters: “Divergence”, “Stage Width”, “Direct Dry/Wet”, “Ambience Gain”, “Ambience Front/Rear”, “Ambience Low Pass”, and “Ambience Delay”. Furthermore, the following parameters on the Advanced panel are affected by the preset: “LFE Gain”, “LFE Low Pass Enable”, “LFE Low Pass Order”, “LFE Low Pass Cutoff Frequency”, and “Output Gain”.

⇨ Upmix presets from the “Cinema” category are designed for the use with X-curve tuned speaker systems. The other presets are designed for listening environments with a flat speaker tuning.

Input – Balance

Adjusts the balance of the input signal if the input signal is stereo or higher.

Upmix – Orig./Upmix

Adjusts the plug-in output between original and processed signal.

Direct Sound Stream Parameters

In the Direct section, the following parameters are available:

Option	Description
Divergence	Controls the strength of the center signal. At 0% the mono components of the direct sound stream are distributed to the center channel. At 100% the mono components of the direct sound stream are distributed to the front left and right channels.
Stage Width	Controls the position of the front channels to adjust the stereo base.
Dry/Wet	Controls the amount of ambience that remains in the direct sound stream after the ambience extraction.

Ambient Sound Stream Parameters

In the Ambience section, the following parameters are available:

Option	Description
Gain	Applies gain to the ambient sound stream to either emphasize (high gain) or dampen (low gain) the amount of ambience in the mix.
Front/Rear	Adjusts the front/rear balance of the ambient sound stream.
Low Pass	Controls the ambient sound stream with a low-pass filter to prevent hissing.
Delay	Adds extra delay to the ambient sound stream to create the illusion of a very large space.

Advanced Options

The Advanced options can be opened using the top left button in the main plug-in panel. These options can be set for the plug-in instance by adjusting the values as needed.

Distance Dependent Parameters

In the “Distance dependent” section, the following parameters are available:

Option	Description
Loudness	Allows you to select whether the volume change that is to be applied depends on the position of the center point, or if the volume change is calculated for each input channel separately. Sets the maximum gain reduction that is applied when the group input or channel reaches the stage border.
EQ Gain	Allows you to select whether the filtering that is to be applied depends on the position of the center point, or if the amount of filtering is calculated for each input channel separately. Sets the maximum gain reduction of the filter that is applied when the group or input channel reaches the stage border.
EQ Cutoff	Sets the cutoff frequency of the distance-dependent EQ.

Upmix – Matrix

Activates matrix decoding for matrix-encoded input signals.

⇨ Matrix decoding is only applied in upmix mode.

LFE Parameters

In the LFE section, the following parameters are available:

Option	Description
LFE Gain	Sets a separate gain level for the LFE channel that is applied to the plug-in output.
LP Enable	Enables a low-pass filter that is applied to the LFE output channel after summing the signals from the input channels.

Option	Description
LP Cutoff	Sets the cutoff frequency for the generated LFE channel.
LP Order	Allows you to select the order, or slope, of the low-pass filter: 2nd order = 12 dB/octave 3rd order = 18 dB/octave 4th order = 24 dB/octave

⇨ Keep in mind that the amount of LFE can be adjusted for each input channel individually, see “Individual Channel Adjustment” on page 99.

⇨ If the selected input configuration includes an LFE channel, but the selected output configuration does not, the LFE input channel is distributed to front left and front right at a level of -3 dB automatically. The low-pass filter is applied to the incoming LFE signal before it is distributed to the front speakers.

MonoToStereo

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–

This effect turns a mono signal into a pseudo-stereo signal. The plug-in must be inserted on a stereo track playing a mono file.

The following parameters are available:

Parameter	Description
Width	Controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.
Delay	Increases the amount of differences between the left and right channels to further increase the stereo effect.
Color	Generates additional differences between the channels to increase the stereo effect.
Mono button	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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	X	–

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

The following parameters are available:

Parameter	Description
Width	Controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.
Delay	Increases the amount of differences between the left and right channels to further increase the stereo effect.
Color	Generates additional differences between the channels to increase the stereo enhancement.
Mono button	Switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when enhancing the stereo image.

SurroundPanner V5

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

Note that in Cubase the SurroundPanner V5 is only available as a channel panner, but not as an insert plug-in. For a description of the SurroundPanner V5 plug-in, see the Operation Manual.

Surround Plug-ins

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

MatrixDecoder

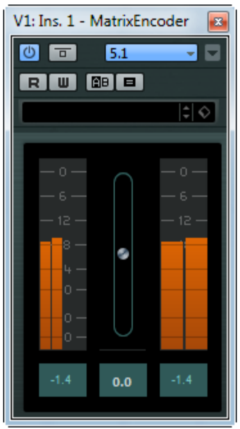
	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	–	X	–

The MatrixDecoder reverses the Encoder process performed by the MatrixEncoder. It is used for monitoring how an encoded mix sounds when played back on a Pro Logic compatible system. When an encoded mix is played back via the decoder, the Lt/Rt channels are again converted to four outputs (LRCS).

⚠ This manual does not attempt to explain the full background on how Pro Logic works, but focuses on how you can use the MatrixEncoder/Decoder to produce a mix that is compatible with this standard.

MatrixEncoder

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	–	X	–



The MatrixEncoder is intended for the Pro Logic compatible encoding of multi-channel files. This is a process where a 4-channel surround mix is packed into two channels for broadcasting or a two-channel version for DVDs, for example. The MatrixEncoder takes four separate inputs (LRCS = Left, Right, Center, and Surround) and creates two final outputs: Left-total and Right-total (Lt and Rt).

Setting Up

1. In the VST Connections window, create an output bus with the LRCS channel configuration and route it to the physical outputs of your audio hardware.
   This is if you want to make a four-channel surround mix. If you want to make a five-channel mix, see "Using the MatrixEncoder with the 5.0 Surround Format" on page 108.
2. Place the MatrixEncoder in the first post-fader insert slot (#7) for the output bus, followed by the MatrixDecoder (#8).

Using the MatrixEncoder/Decoder

1. Set up the mix roughly the way you want it.
   Use the SurroundPanner V5 to place channels in the surround mix, or assign channels to the individual LRCS outputs.
2. Activate the MatrixEncoder.
   What you now hear is the encoded stereo mix, the way it sounds when played back on a normal stereo reproducer. On the MatrixEncoder control panel, you can adjust the Gain of the Lt/Rt output by using the fader.

3. Activate the MatrixDecoder, open the control panel and click the Steering Mode button.

   Now you can hear how the mix is reproduced in surround on a Pro Logic compatible system.

- The Steering display shows an x within the surround field. The position of this x sign indicates the dominant direction of the mix, sometimes referred to as the dominance vector. Part of the processing that is applied for various technical reasons results in the dominant channel being enhanced and the non-dominant channels being reduced in gain.

4. By activating and deactivating the Bypass button in the MatrixDecoder, you can compare the decoded mix with the encoded stereo mix, and make adjustments in the MixConsole as necessary.

   The main goal is to produce a mix that sounds good in both the encoded and the decoded version. To compare the encoded or decoded mix with the unprocessed mix, switch off both the MatrixEncoder and the Decoder.

⚠ The encoding/decoding process produces significant signal loss compared to the unprocessed mix. This is normal, and does not indicate that something is not working properly. However, with careful tweaking of the mix you can decrease the signal degradation to a much more acceptable level. You have to adjust levels and other settings before the signal runs through the MatrixEncoder, since neither the encoder or decoder can control the mix in any way.

5. When you are satisfied with the result, bypass the MatrixDecoder, or remove it from its effect slot.

6. Connect a master recording device to the stereo mix output and perform a mixdown as usual.

   The resulting encoded stereo mix is compatible with common home systems that use the Pro Logic standard.

Using the MatrixEncoder with the 5.0 Surround Format

There are situations when you may want to mix for several surround formats. For example, you might need to mix the same material for 5.1 and LRCS.

5.1 is similar to LRCS. Omitting the LFE channel is easy, but more of a problem is that LRCS only has one surround channel whereas 5.1 has two.

For this reason the MatrixEncoder sums up the surround channels to a mono signal.

Proceed as follows:

1. Create your mix for 5.1.
2. In the VST Connections window, create an output bus with a 5.0 channel configuration and route it to the physical outputs of your audio hardware.
3. Run the mix through the MatrixEncoder.

First, the two surround channels are merged to make the mix compatible with LRCS. Then the four resulting signals are encoded as usual. This way, far fewer adjustments are necessary when working with 5.1 and LRCS at the same time.

Using the MatrixDecoder with the 5.0 Surround Format

Normally two surround speakers are used even when playing back LRCS. The two speakers then simply use the same material. The MatrixDecoder simulates this by delivering the surround channel to two outputs. This allows you to move between formats and listening situations with less repatching of speaker channels.

Mix6To2

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

Mix6To2 lets you quickly mix down your surround mix format to stereo. You can control the levels of up to six surround channels and decide for each channel up to which level it is included in the resulting mix.

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

For each of the surround channels the following parameters are available:

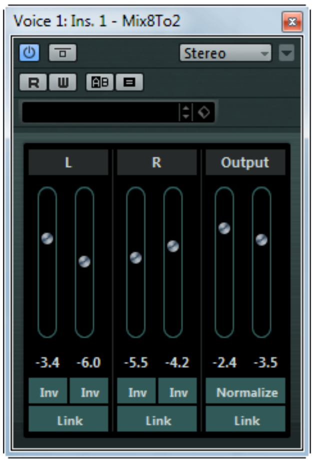
- Two volume faders that govern how much of the signal is included in the left and/or right channel of the output bus.
- A Link button that links the two volume faders.
- Two Invert buttons that allow you to invert the phase of the left and right channel of the surround bus.

For the Output bus the following parameters are available:

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

Mix8To2

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	–	X	–



Mix8To2 lets you quickly mix down your surround mix format to stereo. You can control the levels of up to eight surround channels and decide for each channel up to which level it is included in the resulting mix.

⇨ Mix8To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in should be placed in one of the post-fader insert effect slots for the output bus.

For each of the surround channels the following parameters are available:

- Two volume faders that govern how much of the signal is included in the left and/or right channel of the output bus.
- A Link button that links the two volume faders.
- Two Invert buttons allow you to invert the phase of the left and right channel of the surround bus.

For the Output bus the following parameters are available:

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

MixConvert V6

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

The MixConvert V6 plug-in can be used to quickly convert a multi-channel mix to a format with a different channel configuration, for example, to mix down a 7.1 cinema surround format to a 5.1 home theater format.

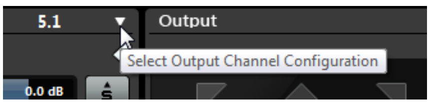
MixConvert V6 can be used as an insert effect like other plug-ins but it also has special functions. The MixConvert V6 plug-in is also used to convert an audio-related channel into a different format if the corresponding input/output configuration is not handled by the SurroundPanner V5. The sequencer application places MixConvert V6 automatically where needed. Furthermore, MixConvert V6 is also in place of any send panner where necessary.

Input/Output Channel Configuration

The input configuration is determined by the channel width of the track, group, or output bus on which MixConvert V6 is inserted.

If MixConvert V6 replaces the panner, the output configuration is determined by the destination of the channel or cue send.

If MixConvert V6 is used as an insert effect, the output configuration can be modified using the "Select Output Channel Configuration" pop-up menu. You can select any configuration from the VST 3 specification that contains speakers that are already present in the input configuration.



⇨ You can also change the output configuration by loading a preset.

Parameters

The plug-in panel is divided into three sections: By default, only the center section is visible, the input and output channel displays can be shown to the left and right of the center section.

- To show/hide the input or output channel configuration, click the corresponding arrow button.

Controls in the Center Section

The center section contains the main plug-in parameters as well as buttons for soloing several speakers in one go.

The following parameters are available:

Parameter	Description
Solo Channel buttons	Solos all front channels, the LFE channel, or all surround channels in the input or output display. All other channels are muted.
Listen to Solo Channels on Center Channel	Routes all soloed channels to the center channel. If no center channel is present, the signal from the soloed channels is distributed equally to the left and right speakers.
Global Gain	Control the level of all output channels.
Listen to Surround Channels on Front Channels	Solos all surround channels, including the side channels, and routes or downmixes them to the front speakers.
Center Level	Controls the level of the front center channel.
LFE fader	Controls the level of the LFE channel.
Surround Level	Controls the level of the surround channels. The level of the surround channels cannot be adjusted individually.
Activate/Deactivate Low-Pass Filter	Activates the low-pass filter (120 Hz) that is applied to the LFE channel.

Soloing Channels in the Channel Displays

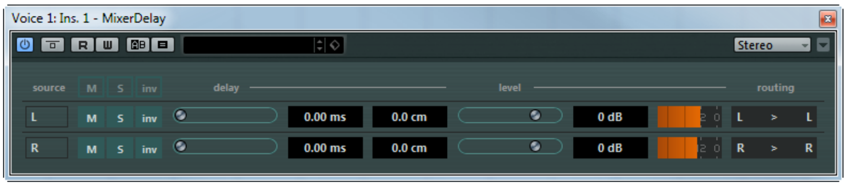
- To solo a channel, click the corresponding speaker icon.
  You can solo several channels at the same time. All other channels are muted.
- To deactivate the solo state of a channel, click the corresponding speaker icon again.
- To solo a channel exclusively, that is, mute all other channels, [Ctrl]/[Command]-click the corresponding speaker icon.
- To solo the LFE channel, click the corresponding cross-hair icon in the center of the channel display.
  This corresponds to the “Solo Input/Output LFE Channel” button in the center section.
- To mute a channel, [Shift]-click the corresponding speaker icon.

Depending on whether you solo a channel in the input or output configuration, the following applies:

- For output configurations, you can only hear the soloed speaker's channel in the downmix.
- For input configurations, you can hear the influence of the soloed speaker's channel on the downmix.

MixerDelay

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–



MixerDelay allows you to adjust and manipulate each individual channel in a surround track, group or bus.

- Above the individual channel controls you find global buttons for turning off Mute, Solo and Invert Phase switches for all channels.

For each channel the following controls are available:

Parameter	Description
Mute button	Allows you to mute individual channels.
Solo button	Allows you to solo individual channels.
Inv button	Lets you invert the phase or polarity for individual channels.
Delay slider	Allows you to delay individual speaker channels. The delay times are shown in milliseconds and centimeters, making this feature very useful for distance compensation when playing back surround mixes on different speaker setups, etc.
Level slider	Allows you to fine-tune the volume balance between the surround channels.
Volume meter	Shows the level of the input signal.
Routing section	Lets you select/switch the outputs for the channels quickly. You can assign the same output to several channels by holding down the [Alt]/[Option] key while selecting. Note that there are also several channel routing presets available.

⇨ It is common for the center channel in a 5.1 speaker configuration to be closer to the mix position in order to accommodate large video monitors or projection screens. In cases like this, MixerDelay can be used to compensate for the center channel being too close. Simply adjust the delay for the center channel by the difference in distance (in cm) between it and the other speakers to the mix position. You must delay the closer speaker so that the sound from it arrives at the same time as the sound from the more distant speakers. Note that MixerDelay has a wide range (up to 1000 ms) and fine adjustments are best made by numerically entering the delay time in centimeters for speaker alignment.

⚠ The MixerDelay is not a mixer – the number of outputs is the same as the number of inputs. If you need to mix down a surround signal to stereo, use the Mix6to2, Mix8to2 or MixConvert V6 plug-ins.

Tools Plug-ins

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

MultiScope

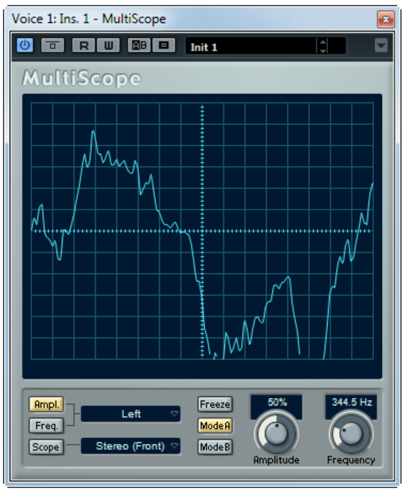
	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–

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.)

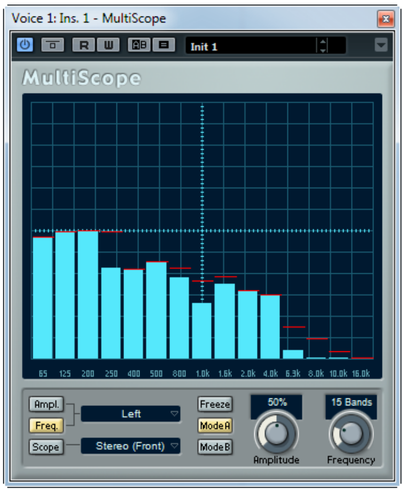
⇨ The Freeze button can be used to freeze the display in all three modes. Click it again to exit freeze mode.

Oscilloscope Mode (Ampl.)



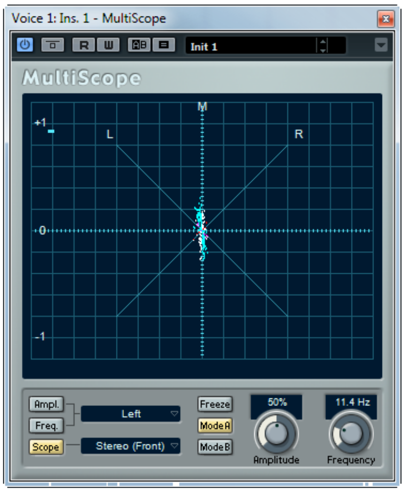
- To view a signal waveform, open the MultiScope control panel and make sure that the “Ampl.” button in the lower left corner is lit.
- If the source signal is stereo you can now select either the Left or Right channel for viewing, or Stereo for both channels to be shown in the window. If it is a mono signal, this does not matter.
- If 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.

Frequency Spectrum Analyzer Mode (Freq.)



- Click the Freq button so that it lights up.
  MultiScope now divides 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 does not matter.
- If 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 set it to “Spectrum”, which gives you 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 do not have any effect if you have set the Frequency knob to “Spectrum”.

Phase Correlator Mode (Scope)



- 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:

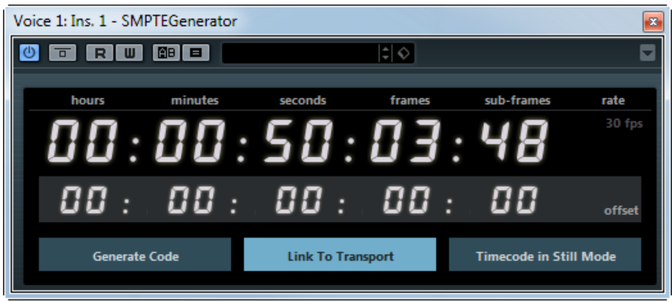
- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but fairly round shape indicates a well balanced stereo signal. If the shape leans to the left, 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 shows 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 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 indicates the phase and amplitude relationship between the front stereo channels.
- If “Surround” is selected, the display indicates the energy distribution in the surround field.

SMPTEGenerator

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–



This plug-in is not a real audio effect. It sends out SMPTE timecode to an audio output, allowing you to synchronize other equipment to your host application (provided that the equipment can sync directly to SMPTE timecode). This can be very useful if you do not have access to a MIDI-to-timecode converter.

The following parameters are available:

Parameter	Description
Main timecode display	This display shows the current timecode. When "Link to Transport" is deactivated, the generator is in free run mode. You can then use the timecode display to set the SMPTE start time. When "Link to Transport" is activated, you cannot change any of the values. This display shows the current timecode in sync with the Transport panel. Where applicable, the offset defined in the offset timecode display is taken into account.
Frame rate display and pop-up menu	The frame rate shown to the right of the timecode display defaults to the frame rate set in the Project Setup dialog. To generate timecode in a different frame rate (for example, to stripe a tape), select another format on the pop-up menu (only available if "Link to Transport" is deactivated). Note that for another device to synchronize correctly to your host, the same frame rate has to be set in the Project Setup dialog, the SMPTE Generator and the receiving device.
Offset timecode display	This display is only available if "Link to Transport" is activated. It allows you to set an offset with regard to the timecode used by your host application. The offset affects the generated SMPTE signal, the current cursor position remains unaffected. For example, use this when playing back video using an external device, where the video starts at a different timecode position than in your host. A scenario could be as follows: Your have placed the same video several times on the timeline, in order to record different audio versions for that video one after the other. However, since video playback is done via an external machine (replaying the same video) you need an offset to match the different timecode positions in your host with the (unchanging) start position on the external machine.
Generate Code button	When you activate this button, the plug-in generates SMPTE timecode in free run mode, meaning that it outputs continuous timecode independent from the Transport panel. Use this mode if you want to stripe tape with SMPTE.

Parameter	Description
Link to Transport button	When you activate this button, the timecode is synchronized to the Transport panel.
Timecode in Still Mode button	When you activate this button, the plug-in also generates SMPTE timecode in stop mode. However, note that this is not continuous timecode, but timecode generated at the current cursor position. For example, this can be useful when working with video editing software that interprets the absence of timecode as a stop command. By using this option, the video software can enter still mode instead so that a still frame is shown instead of a blank screen.

⇨ To change one of the timecode values (main and offset timecode displays), double-click on any of the timecode fields and enter a new value.

Example – Synchronizing a Device to Your Host

1. Use the SMPTE Generator as an insert effect on an audio track, and route that track to a separate output.

   Make sure that no other insert or send effect is used on this track. You should also disable any EQ.

2. Connect the corresponding output on the audio hardware to the timecode input on the device you wish to synchronize to your host application.

   Make all necessary settings for the external device so that it synchronizes to incoming timecode.

3. If needed, adjust the level of the timecode, either in your host application or in the receiving device.

   Activate the Generate Code button (make the device send the SMPTE timecode 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 to Transport” button.

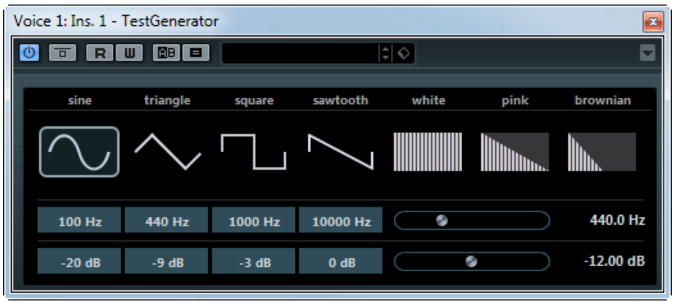
   The SMPTE Generator now outputs timecode that corresponds to the time display of your host application.

6. On the Transport panel, click Play.

   The external device is now synchronized and follows any position changes set with the transport controls.

TestGenerator

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	–	X	X	–



This utility plug-in 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, such as calibrating tape recorders.
- For testing signal processing methods.
- For educational purposes.

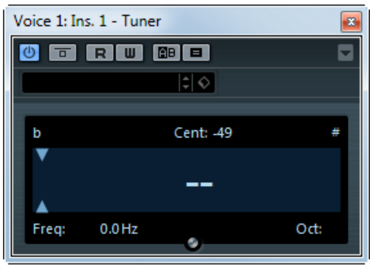
The TestGenerator is based on a waveform generator which can generate a number of basic waveforms such as sine and saw as well as various types of noise. Furthermore, you can set the frequency and amplitude of the generated signal.

As soon as you add the TestGenerator as an effect on 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 and noise section	Allows you to set the basis for the signal generated by the waveform generator. You can select between four basic waveforms (sine, triangle, square, and sawtooth) and three types of noise (white, pink, and brownian).
Frequency section	Allows you to set the frequency of the generated signal. You can select one of the preset values (100, 440, 1000, or 10000 Hz), or use the slider to set a value between 1 and 20000 Hz.
Gain section	Allows you to set the amplitude of the signal. The higher the value (up to 0 dB), the stronger the signal. You can select one of the preset values (for example, -20 dB), or use the slider to set a value between -81 and 0 dB.

Tuner

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	X	X	X	X	X	–



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 you play a note, the pitch 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.

The two arrows indicate any deviation in pitch. If the pitch is flat, they are positioned in the left half of the display, if the pitch is sharp they are in the right half. The deviation is also shown (in Cent) in the upper area of the display.

- If a string is out of tune (for example, if the pitch for the E string is shown as Eb), tune the string so that the correct pitch is shown and the two arrows are in the middle.

  Repeat this procedure for each string.

- To mute the output signal so that you can tune the strings in silence, activate the Mute button at the bottom middle of the plug-in panel.

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 Operation Manual.

Arpache 5

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

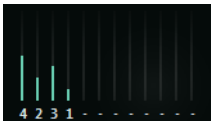
A typical arpeggiator accepts a chord (a group of MIDI notes) as input, and plays back each note in the chord separately, with the playback order and speed set by the user. The Arpache 5 arpeggiator does just that, and more. Before describing the parameters, let's look at how to create a simple, typical 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 Step Size setting to set the arpeggio speed.
   The speed is set as a note value, relative to the project tempo. For example, setting Step Size to 16 means the arpeggio is a pattern of sixteenth notes.
4. Use the Length setting to set the length of the arpeggio notes.
   This allows you to create staccato arpeggios (Length value smaller than the "Step Size" setting) or arpeggio notes that overlap each other (Length value greater than "Step Size").

5. Set the “Key Range” parameter to 12.
   This makes the notes arpeggiate within an octave.
6. Play a chord on your MIDI instrument.
   Now, instead of hearing the chord, you 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. The settings are described below.

Parameters

Arpache 5 has the following settings:

Setting	Description
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.
Step Size	Determines the speed of the arpeggio, as a note value related to the project tempo.
Length	Sets the length of the arpeggio notes, as a note value related to the project tempo. The range is the same as for the Step Size 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 are 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 are added to the arpeggio (as many octaves as fit within the range).
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 means C3, 2 means E3, and 3 means G3. Note that you can use the same number in several slots, creating arpeggio patterns that are not possible using the standard play modes. 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, that is, the chord you play, pass through the plug-in and are sent out together with the arpeggiated notes.

Arpache SX

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

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.

Classic Mode

The following parameters are available:

Parameter	Description
Direction	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 have 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 is looped.
Transpose	When a setting other than “Off” is selected, the arpeggio is expanded upwards, downwards, or both (depending on the mode). This is done by adding transposed repeats of the basic arpeggio pattern.
Repeats	Sets the number of transposed repeats.
Pitch Shift	Determines the transposition of each repeat.
MIDI Thru	If this is activated, the notes sent to the arpeggiator, that is, the chord you play, pass through the plug-in and are sent out together with the arpeggiated notes.
Step Size	Determines the resolution of the arpeggio, that is, 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.

Parameter	Description
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 sorts 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 is the first note, E is 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 are 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 reads 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) generates 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 are 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 is 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 remain empty.
Sort First	As above, but if there are fewer notes in the MIDI input, the missing notes are replaced by the first note.
Sort Any	As above, but if there are fewer notes in the MIDI input, the missing notes are replaced by any (random) note.

Option	Description
Arp. Style	As above, but if there are fewer notes in the MIDI input, the missing notes are replaced by the last valid note in the arpeggio.
Repeat	In this mode, the chords played are not separated into notes. Instead, they are 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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

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:

Option	Description
Waveform	Determines the shape of the controller curves that are sent out. You can click a waveform symbol or choose a value from the pop-up menu.
Wavelength	Sets the speed of the Auto LFO, or rather the length of a single controller curve cycle. 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 is 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.

Option	Description
Density	Determines the density of the controller curves that are sent out. The value can be set to “small”, “medium”, or “large”, or to rhythmically exact note values. The higher the note value, the smoother the controller curve. For example, if you set this to 1/16, a new controller event is sent out at every 1/16 note position.
Value Range	These sliders determine the range of controller values that are sent out, in other words, the bottom and top of the controller curves.

Beat Designer

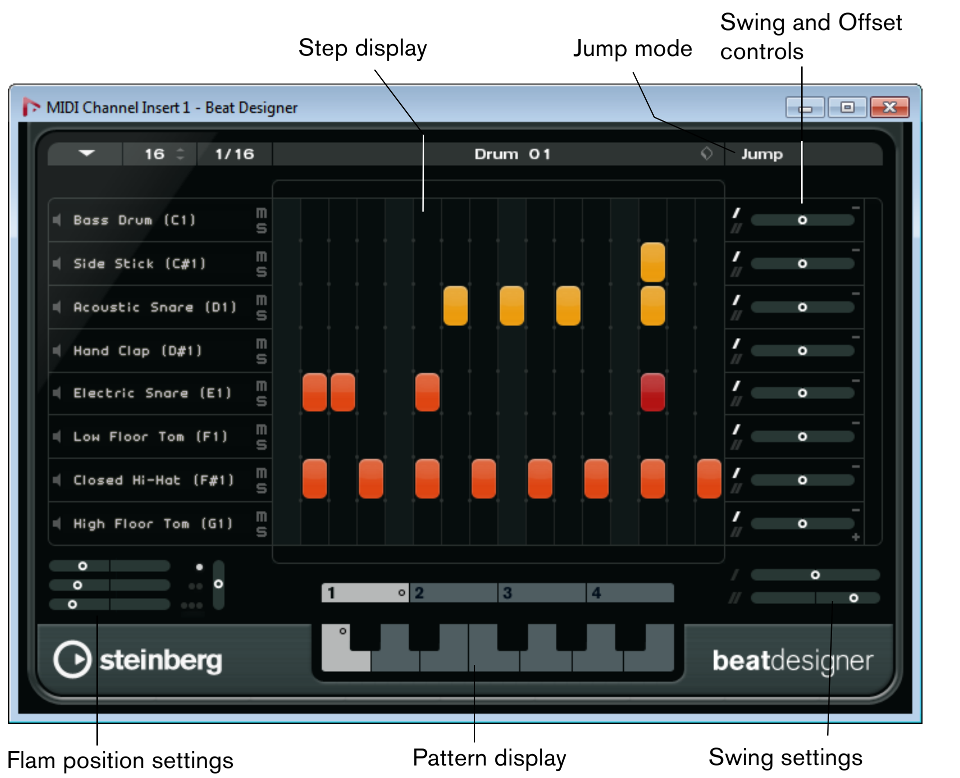
	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	–	X

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 work on a short sequence, adjusting and modifying it while playing it back in a loop. 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 132 and “Triggering Patterns” on page 132.

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 a value.
  The maximum number of steps is 64.
- The playback length, that is, the note value for the steps, can be specified on the “Step Resolution” pop-up menu.
  On this menu, you can also set triplet values. These also affect the Swing setting, see “The Swing Setting” on page 130.

Selecting Drum Sounds

You can click in the drum name field for a lane and select a 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

You can enter a drum step by clicking on the step field where you want to add a beat. For example, add a snare drum on each downbeat for a lane and a bass drum on a second lane.

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.

- To fine-tune the velocity setting for an existing drum step, click on it and drag up or down.
  The current velocity is indicated numerically while you drag.
- To fine-tune the velocity for a range of drum steps, click on the first step, drag up or down to enter velocity edit mode, and then drag sideways and up or down to modify the velocity for all the steps.
  If you change the velocity for several steps at the same time, the relative velocity differences are kept for as long as possible (until the minimum or maximum setting is reached). The velocity for the steps is increased or decreased by the same amount.

⇨ If you hold down [Shift] while dragging up or down, you can change the velocity for all steps on a lane.

- To create a crescendo or decrescendo for an existing range of drum steps, hold down [Alt]/[Option], click on the first step, drag up or down, and then drag to the left or right.

Editing Operations

- To move all drum steps on a lane, hold down [Shift], click on the lane, and drag to the left or right.
- To invert a lane, that is, add drum sounds for all steps that were empty while removing all existing drum steps, hold down [Alt]/[Option] and drag the mouse over the lane. This lets you create unusual rhythmic patterns.
- To copy the content of a lane onto another lane, hold down [Alt]/[Option], click in the section to the left of the lane that you want to copy, and drag.

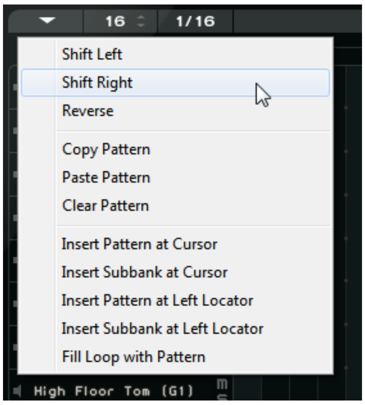
Lane Handling

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 the “Add Instrument Lane” button at the bottom right of the last lane.
- To remove a lane, click the “Remove Instrument Lane” button in the controls section at the far right of the lane.
- To change the order of the drum lanes, click in an empty area in the section to the left of a lane, and drag it to another position.
- To mute or solo a lane, click the corresponding buttons to the left of the step display.

⚠ Lane operations always affect all patterns in the Beat Designer instance.

The Pattern Functions Menu



This menu contains the following editing functions:

Option	Description
Shift Left	Moves all steps of the current pattern on all lanes to the left.
Shift Right	Moves all steps of the current pattern on all lanes to the right.
Reverse	Reverses the pattern, so that it plays backwards.
Copy Pattern	Copies the pattern to the clipboard. Copied patterns can be pasted into another pattern subbank and even directly into the project.
Paste Pattern	Allows you to paste a complete pattern, for example, 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.
Clear Pattern	Resets the current pattern.
Insert Pattern at Cursor	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 132).
Insert Subbank at Cursor	Creates a MIDI part for each used pattern in the subbank and inserts the parts one after the other, starting at the project cursor. For more information, see “Converting Patterns into MIDI Parts” on page 132.
Insert Pattern at Left Locator	Creates a MIDI part for the current pattern and inserts it in the Project window, at the left locator. For more information, see “Converting Patterns into MIDI Parts” on page 132.
Insert Subbank at Left Locator	Creates a MIDI part for each used pattern in the subbank and inserts the parts one after the other, starting at the left locator. For more information, see “Converting Patterns into MIDI Parts” on page 132.
Fill Loop with Pattern	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, that is, the space between the left and right locators. For more information, see “Converting Patterns into MIDI Parts” on page 132.

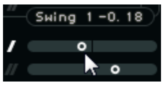
⇨ In the Key Commands dialog, you can set up key commands for the Insert options and the Fill Loop command. How to set up and use key commands is described in the Operation Manual.

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 is offset instead.

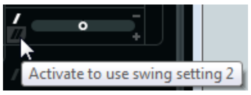
In the lower right section of the Beat Designer panel, you can find two Swing sliders. Dragging a slider to the right delays every second or third drum step in the pattern. Dragging to the left makes them play a little earlier.

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 activated for all lanes, but the slider is set to zero (middle position).



- 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 the buttons to select the corresponding swing setting, or click on a selected button to deactivate swing for this lane.

Adding Flams

The Flam parameter lets you add flams, that is, 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 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.

For example, playing the bass drum or snare a little earlier allows you to add more urgency to the drums, while delaying these drum sounds results in a more relaxed drum pattern. Experiment with the settings to find out which fit best in your project.

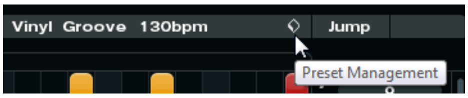
Note that this function can also be used to correct faulty drum samples: If a drum sound has an attack that is slightly late, simply adjust the Offset slider for the lane.

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 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 is now available on the Presets browser, in the MediaBay and on the Load Track Preset pop-up menu in the Inspector.

Pattern banks are handled much like track presets in the MediaBay. For further information, see 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.

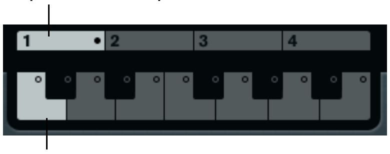
Converting Patterns into MIDI Parts

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 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 is 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. If you did not enter drum steps in a pattern, this is not converted into a MIDI part.

You can also use the Pattern Functions menu to insert patterns or subbanks into the project, see "The Pattern Functions Menu" on page 129.

⚠ 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 continues 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 is lengthened accordingly.

The inserted MIDI parts can now be edited as usual in the project. For example, you can 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 is triggered depends on the pitch of the MIDI notes. The trigger range is four octaves starting with C1 (that is, 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 the Jump field to activate Jump mode.

   In this mode, a MIDI note-on event triggers a new pattern.

Jump mode is activated.

- When you want to trigger the patterns using a MIDI part containing trigger events, you can specify whether the pattern is switched directly (at the moment the event is received) or at the next bar: Click on the field to the right (where it says “Now”) to activate the immediate switching of patterns. When Now is deactivated, patterns 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 immediately 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 starts 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 is played directly, or starts 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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

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 136.

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 in the keyboard display are used to record your chords.

All Keys

In this mode, you can assign chords to each key on the keyboard display. When you play any of these keys, you 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 on a different octave, you 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 C3) are then played by all keys on the keyboard, but transposed according to the note you play.

The Chord Indicator Lane

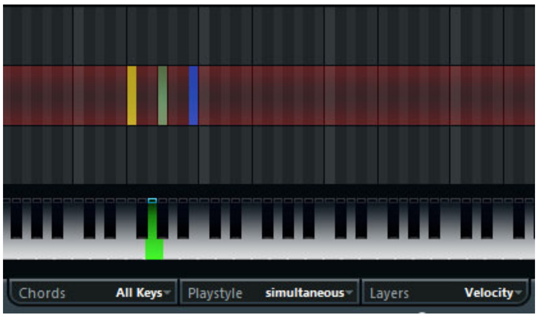
At the top of the keyboard display you find a thin lane with a small rectangle for each key that you can use to record a chord. These rectangles are shown in blue for all keys that already have chords assigned to them.

⇨ 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 example, the keyboard display is shown in red.

The keyboard display in Learn mode



The second layer in Learn mode

Proceed as follows:

1. Click the Learn button at the top 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 keyboard display, or by pressing the key on a connected MIDI keyboard.
   The red bar now moves 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 learns 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 jumps automatically to the next layer where you can record another chord.
  When all the layers for a key are filled, the red bar jumps back to the keyboard display 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 does 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 keyboard. 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:

Mode	Description
Velocity	The full velocity range (1 to 127) is divided into zones, according to the number of layers you specified. For example, if you are using two variations (Number of Layers is set to 2) there are two velocity zones: 1 to 63 and 64 to 127. Playing a note with velocity 64 or higher triggers the second layer, while playing a softer note triggers the first layer. Using the “Velocity spread” slider at the bottom right of the window, you can change the velocity ranges of the layers so that a different layer is activated using the same velocity value.
Interval	In this mode, the Chorder plays 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 a layer, with the lower key determining the base note for the chord. The layer number is the difference, that is 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 are filled automatically when you end the Learn mode.

This works according to the following rules:

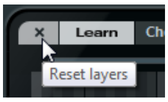
- Empty layers are filled from bottom to top.
- If there are empty layers below the first layer with a chord, these are filled from top to bottom.

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

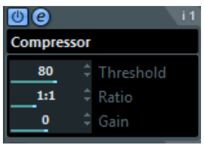
From the Playstyle pop-up menu at the bottom of the pane you can choose one of seven different styles that determine in which order the individual notes of the chords are played back.

The following options are available:

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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



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 Compressor plug-in presents the controls in a manner more like regular audio compressors.

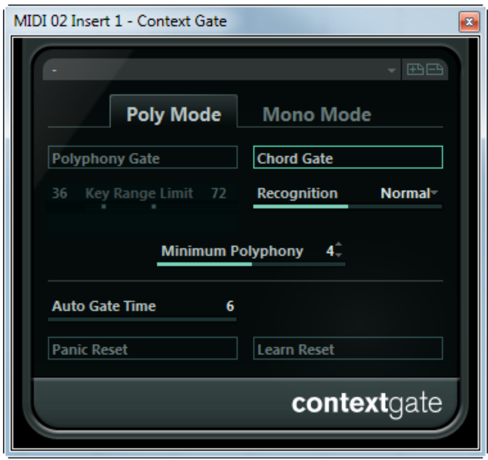
The following parameters are available:

Parameter	Description
Threshold	Only notes with velocities above this value are affected by the compression/expansion.

Parameter	Description
Ratio	Determines the rate of compression applied to the velocity values above the threshold level. Ratios greater than 1:1 result in compression, that is, less difference in velocity, while ratios lower than 1:1 result in expansion, that is, 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	Adds or subtracts a fixed value from the velocities. Since the maximum range for velocity values is 0 to 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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



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 set the key range.
  Only notes within this range are 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 to 16), or to “Any”, that is, no channel gating.

Mono Mode – Velocity Gate

This can be used independently or in conjunction with the Channel Gate function. Played notes 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 set the key range.
  Only notes within this range are let through.
- Notes below the Minimum Velocity threshold value are 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

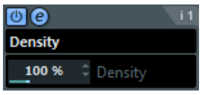
In Poly mode, you could use the Context Gate to accompany yourself during a live guitar performance using a VST instrument. To do this, you might use a guitar-to-MIDI converter: You could then program the Context Gate, for example, to allow only those notes to pass the gate that are part of a four-note chord. During your performance you would then play a four-note chord every time that you want to trigger the VST instrument. The instrument plays until the Auto Gate Time is reached and fades 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 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 (for example, beginning at the 12th band). When you now play one of these notes, the note-off command is not send out and the corresponding note sounds 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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



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% randomly filters out or mutes notes. Raising the setting above 100% instead randomly adds notes that were played before.

Micro Tuner

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

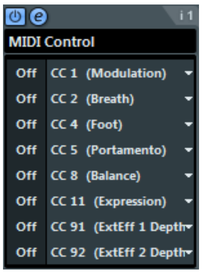
The Micro Tuner lets you set up a different microtuning scheme for the instrument, by detuning each key.

- 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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

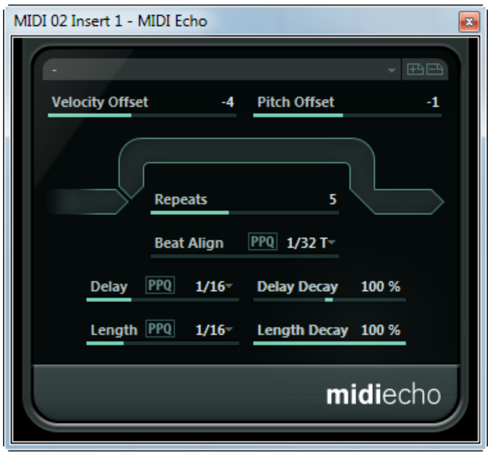


This generic control panel allows you to select up to eight different MIDI controller types, and use the value fields or sliders (which are displayed when you click on a value field while holding down the [Alt]/[Option] key) to set values for these. A typical use for this would be if you are using a MIDI instrument with parameters that can be controlled by MIDI controller data (for example, 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 your host application, at any time.

- To select a controller type, use the pop-up menus on the right.
- To deactivate a controller slider, set it to “Off” (drag the slider all the way down).

MIDI Echo

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



This is an advanced MIDI Echo, which generates 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 does not echo the actual audio, but the MIDI notes which 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 are 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 causes 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, causes 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 is always played together with the note event itself.

Delay

The echoed notes are 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 is the same for all repeats (as set with the Delay parameter).
- If you raise the value above 100%, the echoing notes play with gradually longer intervals, that is, the echo becomes slower.
- If you lower the value below 100%, the echoing notes 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 to 100), the longer the echoed notes, compared to their original notes.

About Ticks and Note Values

The timing and position-related parameters (Delay, Length and Beat Align) 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 the 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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

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 the “Scale Transpose” function that is not 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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

The MIDI Monitor monitors 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, for example, 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 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 are 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

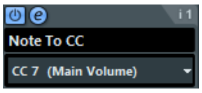
Use this button to the left of the Inputs section to start or stop the monitoring of MIDI events.

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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



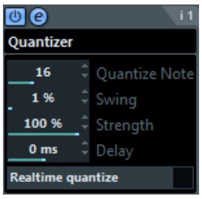
This effect generates a MIDI continuous controller event for each incoming MIDI note. The value of the controller event corresponds to the velocity of the MIDI note, which is then used to control the selected MIDI controller (by default CC 7, Main Volume). For each note end, another controller event 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 that are played control something else. For example, if Main Volume (CC 7) is selected, notes with low velocity lower the volume in the MIDI instrument, while notes with a high velocity 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

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



Quantizing is a function that changes the timing of notes by moving them towards a quantize grid. For example, this grid may consist of straight sixteenth notes, in which case the notes all get perfect sixteenth note timing.

⇨ The main Quantize function in your host application is described in the Operation Manual.

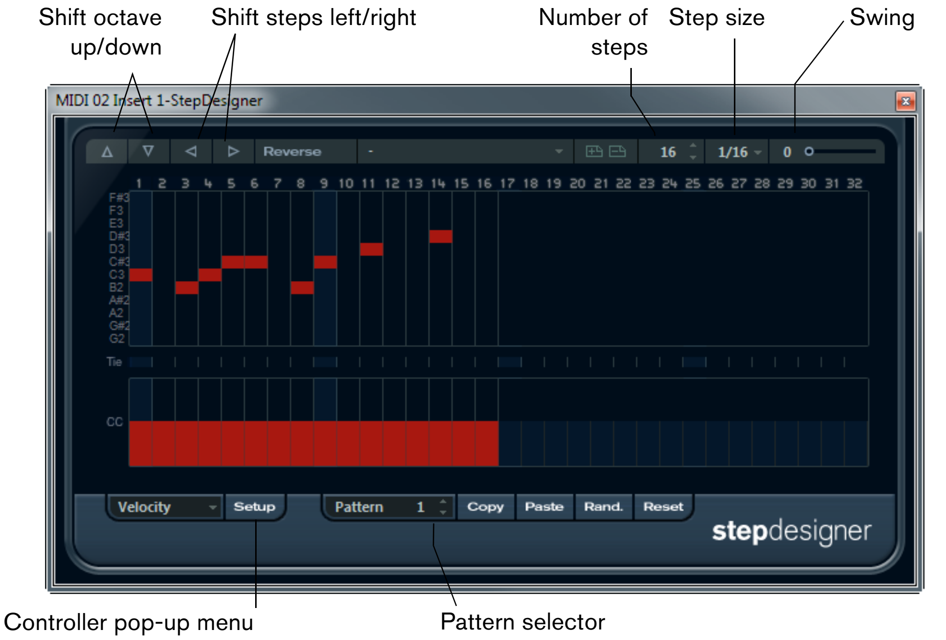
While the Quantize function on the Edit menu applies the timing change to the actual notes on a track, the Quantizer effect allows you to apply quantizing on the fly, changing the timing of the notes in realtime. This makes it easier to try out different settings when creating grooves and rhythms. Note 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	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	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	Determines how close the notes should be moved to the quantize grid. When this is set to 100%, all notes are forced to the closest grid position; lowering the setting gradually loosens the timing.
Delay	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.

StepDesigner

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



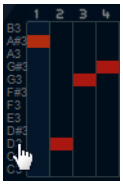
The StepDesigner is a MIDI pattern sequencer that sends out MIDI notes and additional controller data according to the pattern you set up. It does not make use of the incoming MIDI, other than automation data (such as recorded pattern changes).

Creating a Basic Pattern

1. Use the Pattern selector to choose which pattern to create.
   Each StepDesigner 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 is 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 StepDesigner only plays back the number of steps set with the Step size parameter.

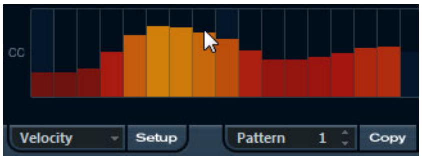
- The display spans one octave (as indicated by the pitch list to the left). You can scroll the displayed octave up or down by clicking in the pitch list and dragging up or down.
  This way you can insert notes at any pitch. Note that each step can contain one note only – the StepDesigner is monophonic.

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 is 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 is not triggered – the previous note is lengthened instead. Also, the tied (second) note automatically gets 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 your host application, the pattern plays as well, sending out MIDI notes on the track’s MIDI output and channel (or, if you activated the StepDesigner 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, that is, it applies to all patterns.
- To insert controller information in a pattern, select a controller from the pop-up menu and click in the controller display to draw events.
  The MIDI controller events are 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	Shift the entire pattern up or down in octave steps.
Shift Steps left/right	Move the pattern one step to the left or right.
Reverse	Reverses the pattern, so that it plays backwards.
Copy/Paste	Allow you to copy the current pattern and paste it in another pattern location (in the same StepDesigner 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	Offsets 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	The handling of presets is described in the Operation Manual. Note that a stored preset contains all 200 patterns in the StepDesigner.

Automating Pattern Changes

You can create up to 200 different patterns in each StepDesigner – 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 realtime by activating the Write automation and switching patterns during playback or by drawing in the automation track for the StepDesigner'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 StepDesigner 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 a MIDI track or create a new one and activate the StepDesigner as an insert effect.
2. Set up several patterns as described above.
3. Press the Record button and press keys on your keyboard to select the corresponding patterns.
   The pattern changes are recorded on the MIDI track.
4. Stop recording and play back the MIDI track.
   You now hear the recorded pattern changes.

⇨ This only works for the first 92 patterns.

Track Control

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–

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 your host application.

Selecting a Control Panel

At the top of the Track Controls effect window you 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.

Control panel	Description
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 find two buttons labeled “Off” and “Reset” at the top of the control panel:

- Clicking the Off button sets all controls to their lowest value, without sending out any MIDI messages.
- Clicking the Reset button sets all parameters to their default values, and sends out the corresponding MIDI messages.

  For most parameters, the default values are 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.

Control	Description
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

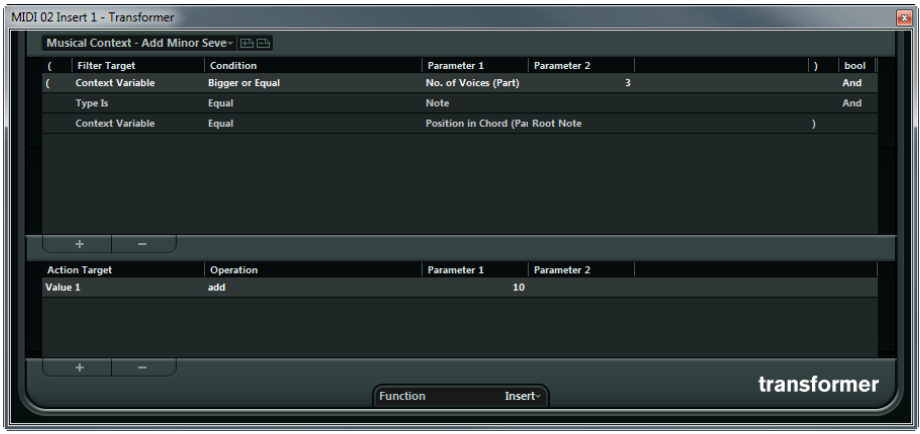
In this mode, the parameters affect global settings in the instrument(s). Changing one of these settings for a track affects all MIDI instruments connected to the same MIDI output, regardless of the MIDI channel setting of the track. Therefore, to avoid confusion it might be a good idea to create an empty track and use this only for these global settings.

The following controls are available:

Control	Description
Eff. 1	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	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	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	Controls 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 MixConsole or in the Inspector).

Transformer

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



The Transformer is a realtime 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 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.

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 VST 3. For further information, see the Operation Manual.

Groove Agent ONE

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	–	X

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 155.
- 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, for example, 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 157).
  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 lights 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 turns light green to indicate that it is selected for editing. Unselected pads not playing back any samples are gray.
- To select multiple pads for sound editing, [Ctrl]/[Command]-click on the pads.
  The pad that has been selected first lights up light green, the rest of the selected pads turn dark green (see “Editing Sounds” on page 157).
- To mute or solo a pad, click the corresponding icon in the upper left corner of a pad.
  The icon lights up to indicate that the pad is muted or soloed. If you solo a pad, all other pads are muted automatically. To unmute or unsolo the pad, click once more on the icon.
- 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 pad.

Velocity

- The velocity is determined by where on the pad you click: it 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 can 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 has 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 your host application onto Groove Agent ONE. Samples can either be mapped to the same pad, or to different pads.

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

- MediaBay
- Project window
- Pool
- Sample Editor (regions)
- Audio Part Editor
- LoopMash slices (if LoopMash is supported)

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) automatically creates 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, select samples, drag them to the Groove Agent ONE window, 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 MIDI pitches of the pads.

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.

Replacing Individual Samples

To replace a sample mapped to one pad with another sample, proceed as follows:

- Drag the new sample to the pad, press [Alt]/[Option] and drop it.

To replace a sample in a pad layer with another sample, proceed as follows:

- Drag the new sample to the Layer indicator, press [Alt]/[Option] and drop it onto the required layer.

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 pointer at an empty pad and let go of the mouse button.
   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 onto the Project window.
   Dropping the file onto the Project window creates a new MIDI track. You can also drop the MIDI file to an existing MIDI or instrument track.
6. Play back the MIDI file.
   The unedited MIDI file plays the same groove as the original audio loop. By editing the MIDI file you can change the original groove.

⇨ If LoopMash is supported, you can use it to slice an audio loop, and drag an individual slice directly from LoopMash to a Groove Agent One sample pad. For further information about LoopMash, see “LoopMash” on page 161.

Saving the Groove Agent ONE Setup

You can save the current configuration of Groove Agent ONE either as a plug-in preset or as a combination of a Groove Agent ONE archive (.gak) and a plug-in preset.

These presets and archives are useful in cases where you want to use your current settings and samples on a different computer.

Saving Plug-in Presets

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

1. At the top of the Groove Agent ONE window, click the button to the right of the Presets pop-up menu and select “Save Preset”.
   The Save Preset dialog opens.
2. Enter a name for the new preset and click OK.
   The preset is saved in the MediaBay under User Content.

Loading Plug-in Presets

To load an existing plug-in preset, proceed as follows:

1. At the top of the Groove Agent ONE window, click the button to the right of the Presets pop-up menu and select “Load Preset”.
   The Presets browser opens.
2. The Presets browser shows all presets it finds in the VST 3 Presets folder for Groove Agent ONE. Double-click a preset to load it.
   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 prompts you to locate the missing files. You can click either Ignore to skip this message, click Locate File to navigate to a specific folder containing the missing file(s), or click Search Folder to browse a specific folder and any subfolders that might contain the missing file(s).

Saving a GAK Archive

You can save all Groove Agent ONE settings, and the sample files referenced by the current configuration, as a Groove Agent ONE kit. The file name extension of these kit files is .gak. Proceed as follows:

1. Set up Groove Agent ONE the way you want it.
2. In the Exchange section, click the Export button.
   The “Export Groove Agent ONE kit” dialog opens in which you can specify a location and a name for the new archive.
3. Click Save.
   The archive is created and the dialog is closed.

⚠ Note that a plug-in preset file is created alongside the .gak file. This plug-in preset references the samples inside the .gak file. It can be browsed in the MediaBay, giving you access to all Groove Agent ONE settings (including all samples) from within your host application.

Loading a GAK Archive

To load the GAK file, proceed as follows:

1. In the Exchange section, click the Import button.
   Navigate to the GAK file.
2. Click Open.
   The saved settings and all samples are imported into Groove Agent ONE.

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. The Polyphony counter shows the number of pads currently playing.

- Click on a pad for sound editing.
  It turns light green and the display shows its sample parameters.
- To adjust a parameter, either use one of the quick controls below the display, or click on the parameter in the display and adjust it by dragging your mouse.
- You can select multiple pads for sound editing by [Ctrl]/[Command]-clicking on them, and adjust their parameters in one go with the quick controls below the display.
  The first selected pad lights up light green, all other selected pads turn dark green. The display shows the parameters of the first selected pad.

- By default, the parameters of the selected samples are adjusted in relation to their previous settings. If you want to set a specific value for all selected samples, [Ctrl]/[Command]-click the quick control to set an initial value, release [Ctrl]/[Command] and adjust the value.

  The parameter is set to the same value for all selected sample pads.

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

Parameter	Description
Brightness slider	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 and dragging it onto the trash icon.
MIDI input off	When the MIDI symbol button in the top right corner of the LCD display is activated, the LCD display shows the waveform and parameter values of the currently playing sample. When this button is deactivated, the display shows 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 fine-tune the sample by up to ±100 cents.
Volume	Sets the sample volume.
Waveform display	The waveform of the current sample.
s/e locators in waveform display	You can define the sample start and end points by dragging the s and e locators in the waveform display. When you click on a locator and press [Ctrl], this zooms in on the waveform and centers the display around the locator. Note that the locators automatically snap to zero crossings.

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 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 fine-tune 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 never play back simultaneously. New notes 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 your host application.

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.
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 sounds louder the higher the velocity. When set to 0%, velocity has 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 has 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.

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).

⇨ Groove Agent ONE imports only the mapping data from the PGM file. Any additional information (on MPC effects, etc.) cannot be imported into 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 156.

The function of the Export button is described in detail in the section “Saving a GAK Archive” on page 157.

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.

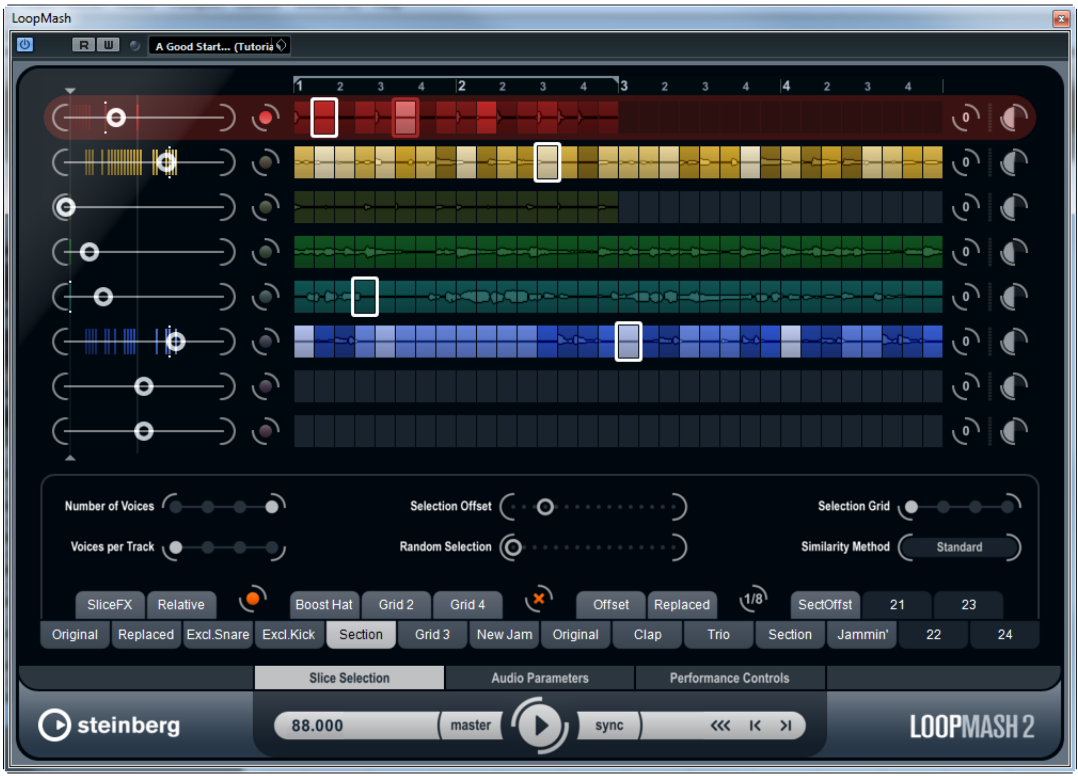
HALion Sonic SE

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	X	X	X	X	X	–	X

This VST instrument is described in detail in the separate PDF document “HALion Sonic SE”.

LoopMash

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	–	X



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 from up to seven other loops.

LoopMash provides dozens of possibilities to influence the way the slices are re-assembled, thus giving you full control over the results of your performance. You can choose from a variety of effects and apply them to single slices or to your overall performance. Finally, you can store your configuration as scenes on scene pads, and trigger these scene pads with your MIDI keyboard. All this turns LoopMash into a really powerful instrument for live performances and recordings!

LoopMash is fully integrated into your host application, which allows you to drag and drop audio loops from the MediaBay or Project window directly onto the LoopMash panel. Furthermore, you can drag and drop slices from LoopMash to the sample pads of Groove Agent One. This allows you to extract certain sounds that you like from

LoopMash and use them with Groove Agent One. LoopMash supports the undo and redo functionality, so that you can see and modify your steps in the Edit History dialog, as long as the LoopMash panel is open (for further information about the Edit History dialog, see the Operation Manual).

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 your host application, create an instrument track with LoopMash as the associated VST instrument.
2. In the Inspector for the new track, click the Edit Instrument button to open the LoopMash panel.

   It has two main areas: the track section in the upper part of the panel, and the parameter section at the bottom.
3. At the top of the plug-in panel, click on the icon to the right of the preset field and select Load Preset from the pop-up menu.
4. The Presets browser opens, showing presets found in the VST 3 Presets folder for LoopMash.
5. Select the preset called “A Good Start…(Tutorial) 88”.

   The preset is loaded into LoopMash.
6. At the bottom of the panel, make sure that the sync button in the transport controls is off, and start playback by clicking 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 – that is, what you hear is the rhythmic pattern of this loop.

7. Look at the 24 pads below the track section: the pad labeled “Original” is selected. Select the pad named “Clap”.

   A new loop is displayed on the second track in the track display, and you hear that the snare drum sound of the first loop has been replaced with a handclap sound.
8. Select the pad labeled “Trio”, and then the pad labeled “Section”. Each time you click, 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.
9. Select other pads to find out how different parameter settings influence the LoopMash output. For a detailed description of the available parameters, see the section “LoopMash Parameters” on page 164.

   Some of the pads have the same label, for example, “Original” and “Replaced”. The scenes that are associated with these pads form the basis for variations of that scene. The variations of a scene are associated with the scene pads to the right of the original scene, that is, the scene labeled “SliceFX” is a variation of the scene labeled “Original” and shows an example for the usage of slice effects (see “Applying Slice Selection Modifiers and Slice Effects” on page 167).

On the left of each track, you find the similarity gain sliders. 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 slices are played back from this track.

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 knows 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.

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.

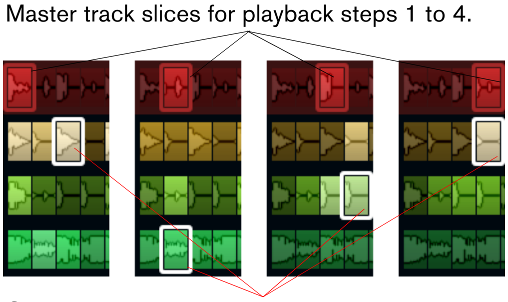
The similarity is shown by the brightness of each slice on each track, and also by the position of each slice on the similarity gain slider to the left of each track (when you click on a slice, its position is highlighted on the similarity gain slider). The brighter a slice, the more similar a slice is to the current master track slice, and the further to the right it is displayed on the similarity gain slider. Darker slices are less similar 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 rectangle in the track's color around the current slice) and automatically selects four slices from these tracks to replace the slices of the master track. The currently playing slice is indicated by a white rectangle around the slice.

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

Master track slices for playback steps 1 to 4.



Slices 1 to 4 selected for playback.

For best performance, use audio files that have the same sample rate as your project (to avoid sample rate conversion when loading presets or storing scenes).

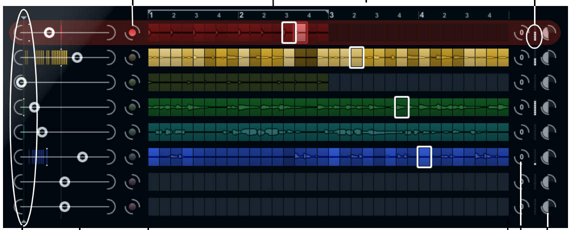
Experiment with the provided LoopMash presets, and with your own loops of different lengths and with different rhythms, containing many different sounds – LoopMash is like an instrument, and we very much encourage you to play it!

LoopMash Parameters

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

⇨ Note that many of LoopMash's parameters can be automated. The automation of VST instrument parameters is described in the Operation Manual.

The Track Section



The track section contains the track display with the track controls for setting the track volume and a transposition value to the right of each track. To the left of the track display you find the similarity gain sliders. With the button between the similarity gain slider and the track, you can define the master track that serves as the reference for rhythm and timbre. At the top of the track display you find a ruler that shows bars and beats and the loop range selector.

Importing and Removing Loops

You can import up to eight audio loops onto the eight tracks in the track display. Proceed as follows:

1. Locate the audio loop that you want to import in one of the following locations: MediaBay and the MediaBay related browsers (for example, the Mini Browser), Project window, Pool, Sample Editor (regions), Audio Part Editor, or the Explorer/Finder.

   The quickest way to find the LoopMash content is to use the MediaBay: Navigate to the LoopMash content via the VST Sound node.

2. Drag the loop file onto a track in LoopMash.

   Dragging a loop to a track already occupied replaces the original loop.

LoopMash separates the loop into slices, analyzes them, and displays them as a waveform on the track. One track can hold up to 32 slices. Even if a long loop were to contain more than 32 slices, LoopMash imports only the first 32. Ideally, you would use a loop file cut at bar boundaries. When you import your file from the MediaBay, LoopMash uses the tempo information supplied by the MediaBay for the slicing of the loop.

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

Defining the Master Loop

One track is always selected. This is the master track: it provides the rhythmic pattern that 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.

- To make a track the master track, activate the button to the left of the track display.

Auditioning Slices

To audition the slices, proceed as follows:

- Click on the slice that you want to hear.
- Use the Step function in the transport controls (see “Transport Controls” on page 168) to step through the slices.

Playback and Master Slice Indicators

A rectangle in the track color around a slice indicates the current position within the master loop, that is, the master slice. The slice currently selected for playback is indicated by a white rectangle.

Setting a Loop Range

At the top of the track display, a ruler showing bars and beats (using the project’s time signature) is displayed. In the ruler, you also find the loop range selector (the bracket) that defines the play length.

- To shorten the play length, click and drag the handles of the loop range selector (the bracket) at the top of the track display.

  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 “Storing Your Configuration as Scenes” on page 169).
- To change the playback range, click the loop range selector and drag it to a different position as a whole.

Setting Track Transposition Value and Track Volume

The track controls to the right of each track allow you to set a track transposition value and the track volume for each track individually.

- To set a track transposition value, click the button to the right of the track and select a transposition interval from the pop-up menu.

  The set value is displayed on the button.

⇨ This function is tied to the setting for the Slice Timestretch parameter (see “Audio Parameters” on page 171). 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, that is, there is no change in playback speed.

- You can change the relative volumes of your tracks with the volume controls on the far right of each track.

  This is useful for level adjustments between tracks. A VU meter to the left of the volume control provides visual feedback of the current volume.

Setting the Similarity

With the similarity gain slider (to the left of each track) you can determine how important a particular track is for the mashing up of the master loop. By moving the slider, you specify that a track is more/less similar to the master track, thus overruling the result of the LoopMash analysis. As a result, more/less slices from this track are included in the current mash.

- Move the slider to the right to select more slices from the corresponding track for playback, and to the left to reduce the number of slices for playback (set to middle position by default).

  The brightness of the slices changes when moving the similarity gain slider. The further to the right, the lighter the color, and the higher the playback priority for these slices.

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.

- Drag the similarity threshold control (the thin line with handles at the top and bottom intersecting all similarity gain sliders) to the left or right to determine a minimum similarity that slices must match to be considered for playback.

  Slices with a similarity below (to the left of) this threshold are not played.

On the Slice Selection page at the bottom of the LoopMash panel, you can make further settings for influencing which slices are played (see "Slice Selection" on page 169).

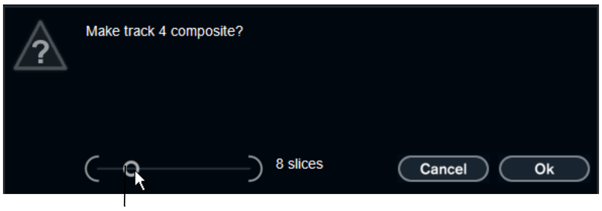
Creating Composite Tracks

LoopMash allows you to build composite tracks, that is, as soon as you drag a slice to a different position on the same track or another track, you are asked if you want to create a composite track.

To build a composite track, proceed as follows:

1. Import the loop that you want to extract sounds from.
2. Audition the slices and drag the slices that you want to use onto an empty track.

   A dialog opens asking you to confirm that you want to create a composite track, and to determine the number of slices that the track contains. If you enter a higher number of slices than the track actually contains, the track is filled up with empty slices.

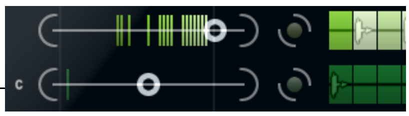


Move this slider to specify the number of slices that the track includes.

3. Click OK.

   The destination track of the dragged slice becomes composite, indicated by a "C" to the left of the track.

Composite track



You can use this feature in a very versatile way:

- You can assemble a combination of sounds that you like most on one track.
- You can define a certain rhythmic pattern by combining slices from different loops on a composite track and making this track the master loop.

- You can use a composite track as a clipboard, allowing you to include sounds from more than eight loops into your mash.
  You can use one track for importing and removing the loops that you want to search for sounds, and use the remaining seven tracks as composite tracks. This allows for including up to 32 sounds from up to 32 different loop files on each of the seven composite tracks.

⇨ Composite tracks are quantized according to the set tempo (see “Transport Controls” on page 168).

Applying Slice Selection Modifiers and Slice Effects

Right-clicking a slice opens a context menu where you can influence the selection of individual slices and which effect is applied to them. The upper part of the context menu shows the slice selection modifiers. The following options are available:

Option	Description
Always	Only available for master track slices. The slice is played always.
Always Solo	Only available for master track slices. The slice is played always and exclusively (independent of the Voices para-meter that you set on the Slice Selection page, see “Slice Selection” on page 169).
Exclude	The slice is never selected for playback.
Boost	Increases the similarity for this particular slice, so that it is played back more often.

Below the selection modifiers, the context menu shows the slice effects. The following options are available:

Option	Description
Mute	Mutes the slice.
Reverse	Plays the slice in reverse.
Staccato	Shortens the slice.
Scratch A, B	Plays the slice as if scratched.
Backspin 4	Simulates a turntable backspin lasting over 4 slices.
Slowdown	Applies a slowdown.
Tapestart	Simulates a tapestart, that is, speeds the slice up.
Tapestop 1, 2	Simulates a tapestop, that is, slows the slice down.
Slur 4	Stretches the slice over 4 slice lengths.
Slur 2	Stretches the slice over 2 slice lengths.
Stutter 2, 3, 4, 6, 8	Plays only the initial portion of a slice, and repeats it 2, 3, 4, 6, or 8 times during one slice length, respectively.

The best way to hear the results of the effects is to try them out!

⇨ You can also apply effects to your overall performance (see “Performance Controls” on page 171).

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 left/right	Clicking the Step left/right button steps 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 your host application, or can follow its own tempo setting:

- Click the sync button (to the right of the Play button) to activate or deactivate synchronization to the project tempo set in your host application.

  When sync is on, you can start playback using the transport controls. With sync off, LoopMash starts 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 to the left of the master 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 tempo field) to copy the tempo of the current master loop into the tempo field.

  The sync on/off parameter can be automated. This is useful to control LoopMash in a project – with sync off, the playback of LoopMash within a project is paused.

Controlling Transport Functions with Your MIDI Keyboard

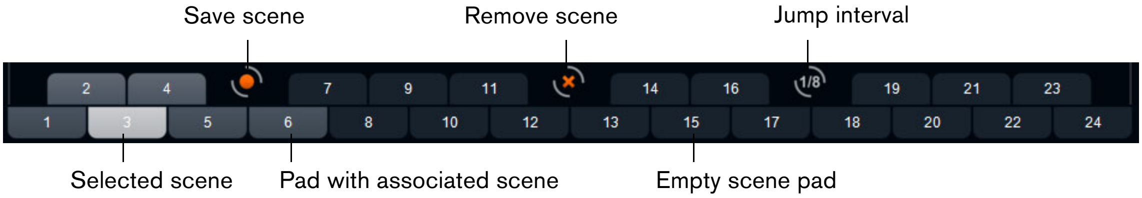
You can control the start, stop, sync on, and sync off function with your MIDI keyboard.

Function	Key
Start	C2
Stop	D2
Sync on	E2
Sync off	F2

⇨ If you do not have a MIDI keyboard connected to your computer, you can make use of the Virtual Keyboard feature (see the Operation Manual).

Storing Your Configuration as Scenes

On the Slice Selection and the Audio Parameters pages you find a row of 24 pads. To each of these pads, you can save one scene, that is, a combination of up to eight tracks with all parameter settings. By triggering the pads, you can quickly change between different scenes during your performance.



- To save the current settings as a scene, click the round button and then a pad. This saves your setup to that pad.
- To recall a scene, click on the corresponding scene pad.
- To remove a scene from a pad, click the x button and then a pad.
- To edit a scene pad label, double-click on the scene pad and enter a name.
- To rearrange the scene pads, click on a scene pad and drag it to a new position.

⚠ 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.

Setting a Jump Interval

You can determine a point at which LoopMash changes to the next scene during playback when you trigger a pad. Proceed as follows:

- Click the Jump interval button and select an option from the pop-up menu that opens.

⇨ The option “e: End” means that the current loop is played to the end before switching scenes. When you set up a short loop range, you may need to set the interval to “e: End” to ensure that the jump point is reached.

Triggering Scene Pads with Your MIDI Keyboard

As you can see, the scene pads are arranged according to the keys on a MIDI keyboard. You can trigger the 24 scene pads with a connected MIDI keyboard starting from C0 and ending with B1. You can also make use of the Virtual Keyboard for triggering the scene pads (see the Operation Manual).

Slice Selection

Click the Slice Selection button (above the transport controls) to open the Slice Selection page. The options on this page allow you to further influence which slices are selected for playback.

The following parameters are available:

Parameter	Description
Number of Voices	Here you can set the total number of slices from all tracks that replace the master slice (according to the current similarity gain settings). The range is from one (left) to four (right) voices, that is, sounds from up to four loops can play simultaneously. Increasing the number of voices increases the CPU load.

Parameter	Description
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 get in the LoopMash output.
Selection Offset	Move this slider to the right to allow slices that are less similar to be selected for playback. This setting affects all tracks of this scene (see "Storing Your Configuration as Scenes" on page 169).
Random 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 "Storing Your Configuration as Scenes" on page 169).
Selection Grid	Here you can determine how often LoopMash looks for similar slices during playback: always (left position), or only every 2nd, 4th, or 8th (right position) step. For example, if you set the Selection Grid to every 8th step (right position), LoopMash replaces similar slices every 8th step. Between two replacement steps it plays back the tracks of the slices that have been selected in the last replacement step, resulting in longer playback sequences on one track.
Similarity Method	Here you can modify the criteria that LoopMash considers when comparing the slices for similarity. There are three similarity methods: Standard – This is the standard method, where all slices on all tracks are compared and various characteristics regarding rhythm, tempo, spectrum, etc. are taken into account. Relative – This method does not only consider the overall similarity of all slices on all tracks, but also takes the relation to the other slices on the same track into account, for example, LoopMash may replace the loudest, lowest sound on one track with the loudest, lowest sound on another track. Harmonic – This method only takes the analyzed tonal information into account, so that a slice is replaced by a harmonically similar slice, rather than by a rhythmically similar slice. With this method, also the track transposition value is considered, that is, a master slice with a C major chord is not replaced by a slice with a D major chord. But it is indeed replaced if you set the transposition value of the track of the slice with the D major chord to -2. It is advisable to keep the similarity gain sliders in a low position when you work with this method, because otherwise you may produce disharmonies. You can modify the transposition values to play back more slices of a specific track.

Audio Parameters

Click the Audio Parameters button (above the transport controls) to open the Audio Parameters page. With the options on this page you can influence the sound of the LoopMash audio output.

The following parameters are available:

Option	Description
Adapt Mode	With Adapt Mode, you can adapt the sound of the selected slice to the sound of the master slice. The available options are: Volume – changes the overall volume of the selected slice. Envelope – modifies volume changes within the slice. Spectrum – modifies the spectrum of the slice (equalization). Env + Spectrum – this is a combination of the Envelope and Spectrum modes.
Adapt Amount	Move this slider to the right to increase the adaptation specified with the Adapt Mode parameter.
Slice Quantize	Move this slider to the right to apply quantizing to the slices, that is, the slices are aligned to an eighth-note grid. When the slider is all the way to the left, the slices follow the rhythmic pattern defined by the original master loop.
Slice Timestretch	Use this option to apply realtime timestretching to the slices, filling gaps or avoiding overlaps between slices that are not played back at their original tempo, or when combining slices with different original tempos. Applying timestretch increases 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 (see "Setting Track Transposition Value and Track Volume" on page 165).
Staccato Amount	When you move this slider to the right, the length of the slices is gradually reduced, giving the output a staccato feel.
Dry/Wet Mix	This sets the balance between the volumes of the master loop and the selected slices from the other tracks.

Performance Controls

Click the Performance Controls button to open the Performance Controls page. On this page you find a row of buttons that are arranged according to the keys on a MIDI keyboard.

- By clicking these buttons during playback, you can apply effects to your overall performance.

  An effect is applied as long as you keep the button activated.

Most of the available effects correspond to the effects that you can apply to single slices, with the green buttons corresponding to the stutter and slur effects and the red buttons to the Mute, Reverse, Staccato effects, etc. (see "Applying Slice Selection Modifiers and Slice Effects" on page 167).

⇨ Effects triggered with the Performance Controls buttons override the slice effects.

With the blue buttons and the yellow button, you can apply additional effects that cannot be applied to single slices:

Button	Description
Cycle 4, 2, 1	Temporally sets up a short cycle over 4, 2, 1 slice, respectively. This short cycle is always set up within the loop range that is set in the ruler (see “Setting a Loop Range” on page 165). Setting up a cycle over 1 slice means that this slice is repeated until you release the button.
Continue	Plays back the tracks of the currently selected slices continuously until you release the button.

⇨ You cannot save these global effects in scenes. To apply effects and save them in scenes, you should use slice effects.

Triggering the Performance Controls with Your MIDI Keyboard

You can trigger the Performance Controls with your MIDI keyboard starting from C3 upwards. You can also make use of the Virtual Keyboard for triggering the Performance Controls (for information about the Virtual Keyboard see the Operation Manual).

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 opens.
2. Enter a name for the new preset and click OK.
   You find the preset in the MediaBay under User Content. Make sure that 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 opens.
2. The Presets browser shows all presets it finds in the VST 3 Presets folder for LoopMash. Double-click a preset.
   The Presets browser is closed and the preset is loaded into LoopMash.

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

⇨ The “Empty” preset clears all settings of the current LoopMash instance.

Loading VST Presets Saved with a Former Version of LoopMash

When you load a VST preset that was saved with a previous version of LoopMash, all new parameters are automatically set to values that match the behavior of the previous LoopMash version.

⇨ To ensure MIDI control compatibility, the saved scenes are shifted to the scene pads 13–24, that is, the scene on pad 1 is shifted to pad 13, the scene on pad 2 to pad 14, and so on.

Mystic

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	–	X

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) causes a resonating tone – this tone is basically what the Mystic produces. 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 largely affects 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 results in a resonating feedback tone.

- When the signal is fed back into the comb filter, it goes via a separate, variable low-pass filter.
  This filter corresponds to the damping of high frequencies in a physical instrument – when set to a low cutoff frequency it causes 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 simulates 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 212.

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

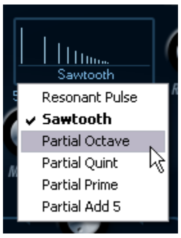
The displays allow you to draw a filter contour with your mouse for spectrum filters A and B.

- To set up the contour, click in one of the displays and drag the mouse to draw a curve. Note that this produces 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 menu.

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

Waveform Pop-up Menu



The pop-up menu 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.

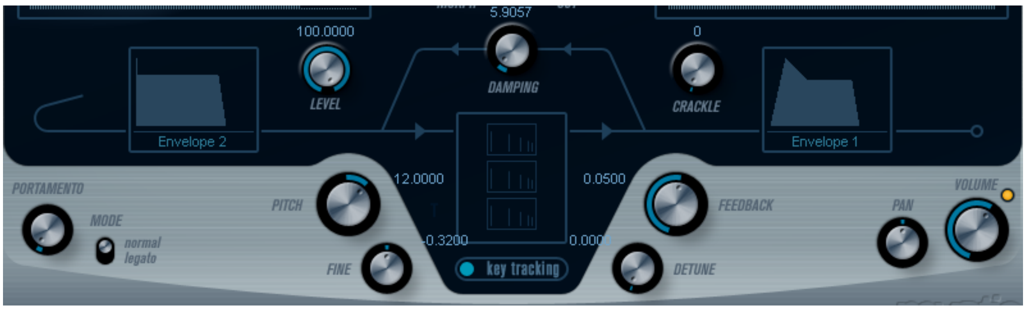
Coarse

This offsets the pitch for the impulse sound. In a typical string setup, when the impulse sound is very short, this does 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 changes the final timbre.

Comb Filter Sound Parameters



Damping

This is a 6 dB/oct low-pass filter that affects the sound being fed back into the comb filters. This means the sound becomes gradually softer when decaying, that is, high harmonics 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 is static – the sound does 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 produce a crackling, erratic effect; higher amounts 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) effectively turns off the comb filter sound, as no feedback tone is produced.
- Setting Feedback to a positive value creates a feedback tone, with higher settings generating longer decays.
- Setting Feedback to a negative value creates 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 affects 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 knob controls the position in the stereo spectrum for the instrument. You can use Pan as a modulation destination.

Modulation and Controllers

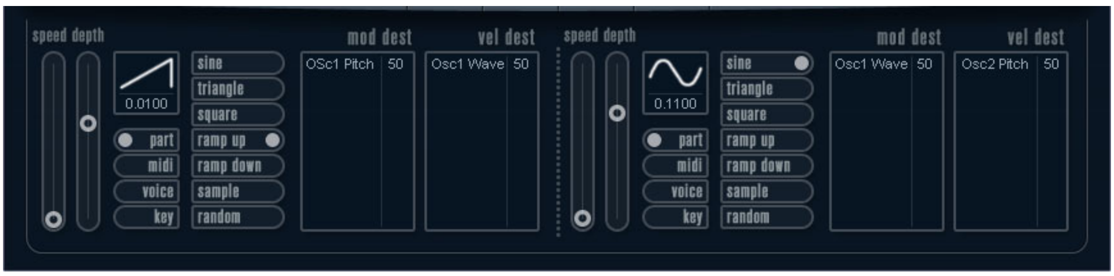
The lower half of the control panel displays the various modulation and controller assignment pages available as well as the effect page. You switch between these pages using the buttons above this section.

The following pages are available:

- The LFO page has two low frequency oscillators (LFOs) for modulating parameters.
- The Envelope page contains the four Envelope generators which can be assigned to control parameters – see “Envelope Page” on page 180.
- The Event page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments – see “Event Page” on page 182.
- The Effect page has three separate effect types available; Distortion, Delay and Modulation – see “Effects (EFX) Page” on page 183.

LFO Page

The LFO page 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 179.

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 required.

The two LFOs have identical parameters:

Parameter	Description
Speed	This governs the rate of the LFO. If the sync mode is set to MIDI, the available rate values are selectable as note values, so the rate is synced to the sequencer tempo 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.

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 affects all the voices in sync. “Free running” means that the LFO cycles continuously, and does not 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.

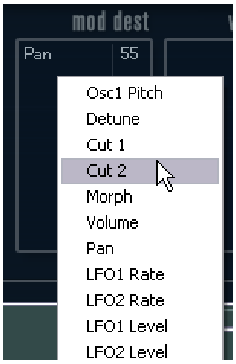
  For example, if LFO 2 is set to use Sample the resulting effect also depends on the speed and waveform of LFO 1.

Assigning LFO Modulation Destinations

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

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

   A pop-up menu appears in which 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, for example, 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 are all 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 menu.

Assigning LFO Velocity Destinations

You can also assign velocity-controlled LFO modulation, that is, the modulation is governed by how hard or soft you strike a key. Proceed as follows:

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

   A pop-up menu appears in which 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 are all 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 menu.

LFO modulation velocity control – an example:

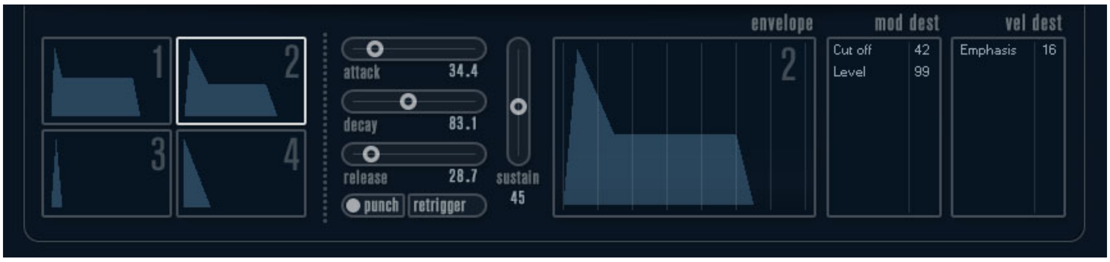
If you follow the steps above and select the Cut parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the Cut parameter is 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 changes when a key is pressed, when a key is held and finally when a key is released.



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 selects it and displays 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).
- 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 adjusts how the volume of the sound changes 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 176.

The Envelope parameters are as follows:

Attack

The attack phase is the time it takes from zero to the maximum value. How long this takes is governed by the Attack setting. If the Attack is set to 0, the maximum value is reached instantly. If this value is raised, it takes 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 takes 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 rests 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 re-triggers 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 menu appears in which 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, for example, 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 are all 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 menu.

Assigning Envelope Velocity Destinations

You can also assign velocity-controlled Envelope modulation, that is, the modulation is governed by how hard or soft you strike a key. Proceed as follows:

1. Click in the “Vel Dest” box for one of the envelopes.
   A pop-up menu appears in which 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 are all 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 menu.

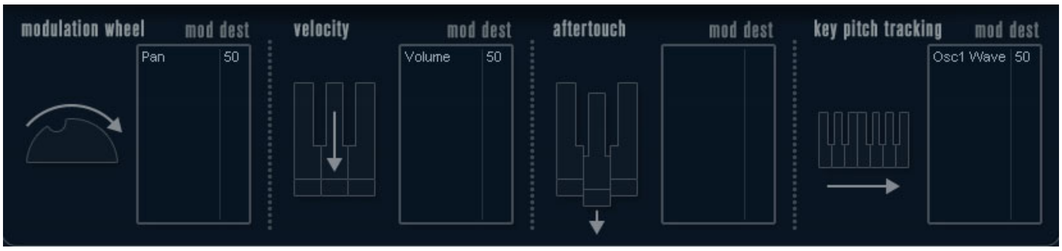
Envelope modulation velocity control – an example:

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 is 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 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 assignments.



The following controllers are available:

Controller	Description
Modulation Wheel	The modulation wheel on your keyboard can be used to modulate parameters.
Velocity	Velocity controls 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.

Controller	Description
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 menu appears in which 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.

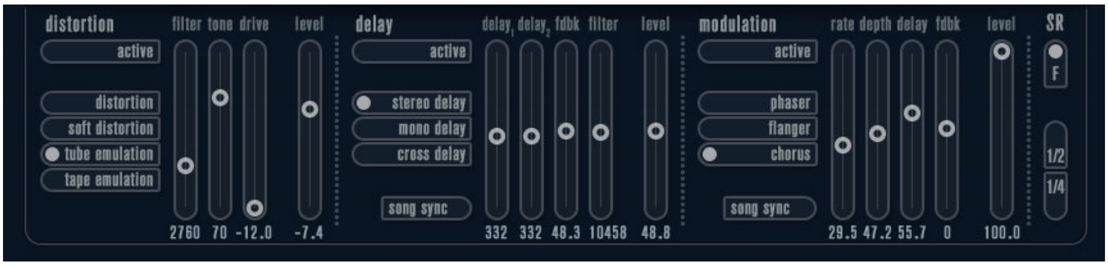
- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the [Enter] key.
  To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of modulation destinations for the controllers.
   They are all listed in the “Mod Dest” box for each controller.

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

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 low-pass filter and a high-pass filter with a cutoff frequency equal to the crossover frequency.
Tone	This parameter controls the relative amount of low-pass 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 low-pass 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 all-pass 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. 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 is 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 reduce the high frequency content and sound quality, but the pitch is not 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 plays back with the sample rate set in the host application.
- If button “1/2” is active, the selected Part’s program plays back with half the original sample rate.
- If button “1/4” is active, the selected Part’s program plays 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.

Padshop

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	–	X

Padshop is described in detail in a separate PDF document, which can be accessed via the “?” button on the plug-in interface.

Prologue

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	X	X	X	–	X

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

- Multimode filter
  Variable slope low-pass and high-pass, plus band-pass and notch filter modes – see “About the Filter Types” on page 192.
- Three oscillators, each with 4 standard waveforms plus an assortment of specialized waveforms.
  See “Selecting Waveforms” on page 187.
- Frequency modulation.
  See “About Frequency Modulation” on page 190.
- Ring Modulation.
  See “Ring Modulation” on page 190.
- Built-in effects.
  See “Effects (EFX) Page” on page 199.
- Prologue receives MIDI in Omni mode (on all MIDI channels).
  You do not have 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 212.

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.

The following waveforms are available:

Waveform	Description
Sawtooth	This waveform contains all harmonics and produces a bright and rich sound.
Parabolic	This can 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.
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 can 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 controls 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	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 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 produces a pulse waveform. By modulating the WM parameter with for example an LFO, classic PWM (pulse width modulation) is produced. However, wave modulation can be applied to any waveform.
Phase button (On/Off)	When Phase synchronization is activated, all oscillators 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 starts from a random phase in the cycle, adding warmth to the sound. But when synthesizing bass sounds or drum sounds, it is often required that the attack of every note played sounds the same, so for these purposes activate Phase sync. Phase sync also affects the noise generator.
Tracking button (On/Off)	When Tracking is activated, the oscillator pitch tracks 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 menu (see “Selecting Waveforms” on page 187)	Sets the basic waveform for the oscillator.

OSC 2 Parameters

Oscillator 2 has the following parameters:

Parameter	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	Description
Wave Mod (±50)	This parameter 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 produces a pulse waveform. By modulating the WM parameter with for example an LFO, classic PWM (pulse width modulation) is produced. However, wave modulation can be applied to any waveform.
Ratio (1–16)	This parameter (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 190. 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 an 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 tracks 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 pop-up menu (see “Selecting Waveforms” on page 187)	Sets the basic waveform for the oscillator.

OSC 3 Parameters

Oscillator 3 has the following parameters:

Parameter	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 (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 190. Is normally referred to as FM index.

Parameter	Description
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 an 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 tracks 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 pop-up menu (see "Selecting Waveforms" on page 187)	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 is 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, 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 happens.

  However, if you change the pitch of Osc 2, 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 sounds harmonic, other intervals produce inharmonious, complex timbres.
- Deactivate Oscillator Sync 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, 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 195 for a description of the Envelope generators.

Filter Section

The circle in the middle contains the filter parameters. The central control sets the filter cutoff parameter and the outer ring the filter type:

Parameter	**Description**
Filter type	Sets the filter type to either low-pass, high-pass, band-pass or notch, see “About the Filter Types” on page 192.
Cutoff	This knob controls the filter frequency or cutoff. If a low-pass 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.
Emphasis	This is the resonance control for the filter. For low-pass and high-pass filters, raising the Emphasis value emphasizes 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 band-pass or notch filters, the Emphasis setting adjusts the width of the band. When you raise the value, the band where frequencies are let through (band-pass), or cut (notch) becomes narrower.
Drive	This can be used to adjust the filter input level. Levels above 0 dB gradually introduce a soft distortion of the input signal, and a decrease of the filter resonance.

Parameter	Description
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 low-pass and high-pass filter types it changes the filter slope. For band-pass and notch filter types it changes the bandwidth. The Shift parameter has no effect if either the 12dB LP or 12dB HP filter type is selected.
Tracking	If this parameter is set to values over the 12 o'clock position, the filter cutoff frequency increases the further up on the keyboard you play. Negative values invert this relationship. If the Tracking parameter is set fully clockwise, the cutoff frequency tracks the keyboard by a semitone per key.

About the Filter Types

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

Type	Description
12dB LP	Low-pass filters let low frequencies pass and cut out the high frequencies. This low-pass filter has a gentler slope (12dB/Octave above the cutoff frequency), leaving more of the harmonics in the filtered sound.
18dB LP	This low-pass filter also has a cascade design, attenuating frequencies above the cutoff frequency with a 18dB/Octave slope, as used in the classic TB 303 synth.
24dB LP	This filter type attenuates frequencies above the cutoff frequency with a 24dB/Octave slope, which produces a warm and fat sound.
24dB LP II	This low-pass filter has a cascade design which attenuates frequencies above the cutoff frequency with a 24dB/Octave slope, which produces a warm and dark sound.
12dB Band	This band-pass filter cuts both high and low frequencies above and below the cutoff frequency with a 12dB/Octave slope, producing a nasal and thin sound.
12dB Notch	This notch filter cuts off frequencies near the cutoff frequency by 12dB/Octave, letting the frequencies below and above through. This produces a phaser-like sound.
12dB HP	A high-pass filter is the opposite of a low-pass filter, cutting out the lower frequencies and letting the high frequencies pass. This high-pass filter has a 12dB/Octave slope, giving a bright and thin sound.
24dB HP	This filter has a 24dB/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 knob controls the position in the stereo spectrum for the instrument. You can use Pan as a modulation destination.

Modulation and Controllers

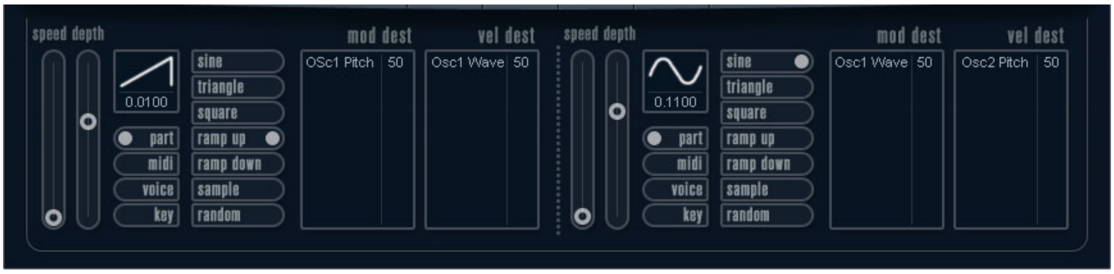
The lower half of the control panel displays the various modulation and controller assignment pages available as well as the effect page. You switch between these pages using the buttons below the Filter section.

The following pages are available:

- The LFO page has two low frequency oscillators (LFOs) for modulating parameters.
- The Envelope page contains the four Envelope generators which can be assigned to control parameters – see “Envelope Page” on page 195.
- The Event page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments – see “Event Page” on page 198.
- The Effect page has three separate effect types available; Distortion, Delay and Modulation – see “Effects (EFX) Page” on page 199.

LFO Page

The LFO page 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 194. 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 required.

The two LFOs have identical parameters:

Parameter	Description
Speed	This governs the rate of the LFO. If the sync mode is set to MIDI, the available rate values are selectable as note values, for example, beat increments of the sequencer tempo.
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.

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 affects all the voices in sync. “Free running” means that the LFO cycles continuously, and does not 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

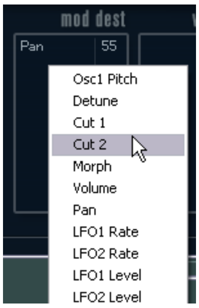
Most standard LFO waveforms are available for LFO modulation. You use Sine and Triangle waveforms for smooth modulation cycles, Square and Ramp up/down for different types of stepped modulation cycles and Random or Sample for random modulation. The Sample waveform is different. In this mode, one LFO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

Assigning LFO Modulation Destinations

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

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

   A pop-up menu appears in which 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, for example, 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 cutoff being modulated by the LFO.

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

   They are all 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 menu.

Assigning LFO Velocity Destinations

You can also assign velocity-controlled LFO modulation, that is, the modulation is governed by how hard or soft you strike a key. Proceed as follows:

1. Click in the "Vel Dest" box for one of the LFOs.
   A pop-up menu appears in which 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 are all 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 menu.

LFO modulation velocity control – an example:

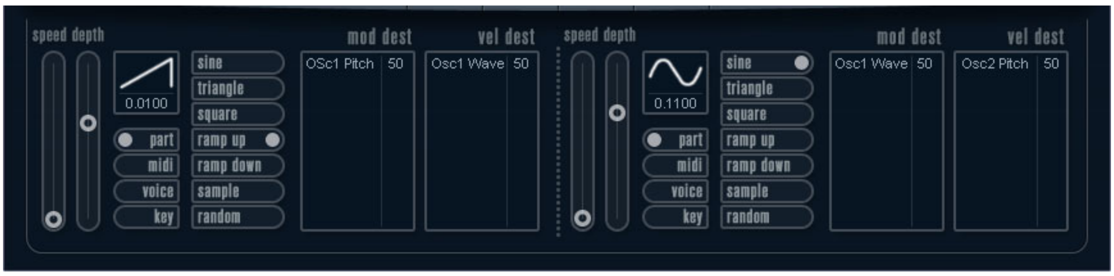
If you follow the steps above and select the filter cutoff parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the filter cutoff parameter is 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 cutoff 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 changes when a key is pressed, when a key is held and finally when a key is released.



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 selects it and displays 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).

- 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 adjusts how the volume of the sound changes 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 takes is governed by the Attack setting. If the Attack is set to 0, the maximum value is reached instantly. If this value is raised, it takes 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 takes 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 rests 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, that is, 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 re-triggers 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 menu appears in which 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, for example, 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 cutoff 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 are all 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 menu.

Assigning Envelope Velocity Destinations

You can also assign velocity-controlled Envelope modulation, that is, the modulation is governed by how hard or soft you strike a key. Proceed as follows:

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

   A pop-up menu appears in which 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 are all 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 menu.

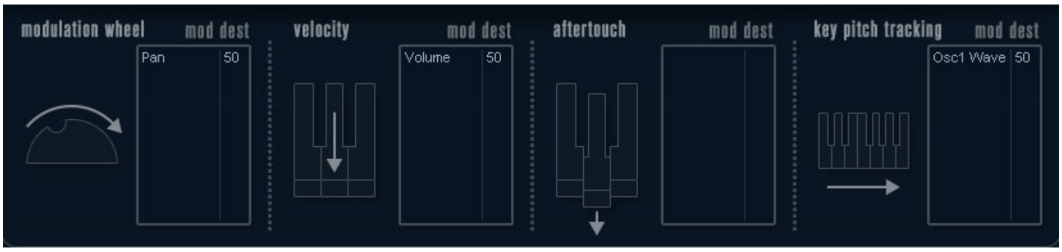
Envelope modulation velocity control – an example:

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

- The harder you strike the key, the more the filter cutoff parameter is 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 cutoff 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 assignments.



The following controllers are available:

Controller	Description
Modulation Wheel	The modulation wheel on your keyboard can be used to modulate parameters.
Velocity	Velocity controls 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 menu appears in which 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.

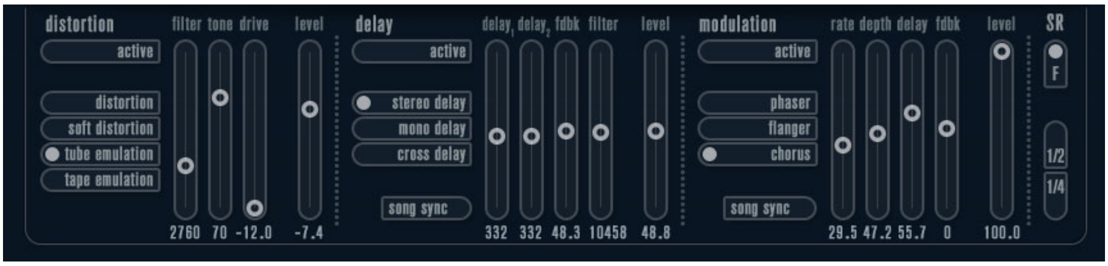
- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the [Enter] key.
  To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of modulation destinations for the controllers.
   They are all listed in the “Mod Dest” box for each controller.

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

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 low-pass filter and a high-pass filter with a cutoff frequency equal to the crossover frequency.
Tone	This parameter controls the relative amount of low-pass 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.

Parameter	Description
Filter	A low-pass 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 all-pass 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 is 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 reduce the high frequency content and sound quality, but the pitch is not 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 plays back with the sample rate set in the host application.
- If button “1/2” is active, the selected Part’s program plays back with half the original sample rate.
- If button “1/4” is active, the selected Part’s program plays 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.

Retrologue

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	–	X

Retrologue is described in detail in a separate PDF document, which can be accessed via the “?” button on the plug-in interface.

Spector

	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	–	X

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, labeled 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 204.

⇨ The signal flow of the Spector synth is illustrated in the section “Diagrams” on page 212.

Sound Parameters

Oscillator Section

A/B Waveform Pop-up Menus

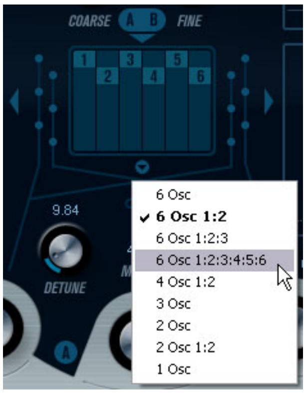
This is where you select basic waveforms for the A and B output of the oscillators. The options are especially suited for use with the spectrum filter.

Coarse and Fine

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

Oscillator Pop-up Menu

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



The pop-up menu 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.

Option	Description
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 give gentle chorus-like detuning; raising the control detunes 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:

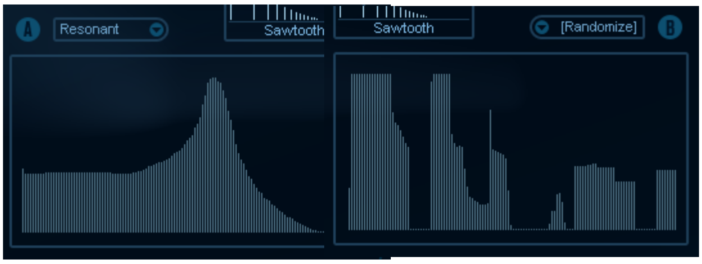
Setting	Description
0	All harmonics present.
1	Only every second harmonic present.
2	Only every third harmonic present.
…	…and so on.

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 Sections



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 is 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 menu.

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

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 is used for the spectrum filter; lowering the Cut controls gradually moves 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 follow each other and are 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 is heard; when it is turned right only the "B" sound is 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 knob 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:

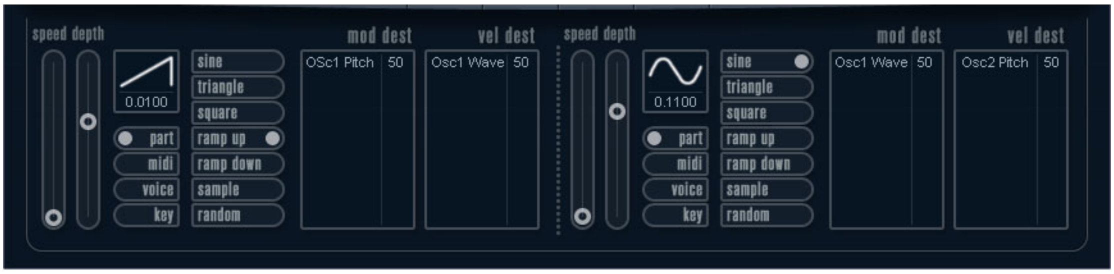
- The LFO page has two low frequency oscillators (LFOs) for modulating parameters.

- The Envelope page contains the four Envelope generators which can be assigned to control parameters – see "Envelope Page" on page 207.

- The Event page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments – see “Event Page” on page 209.
- The Effect page has three separate effect types available; Distortion, Delay, and Modulation – see “Effects (EFX) Page” on page 210.

LFO Page

The LFO page 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 206. 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 required.

The two LFOs have identical parameters:

Parameter	Description
Speed	This governs the rate of the LFO. If the sync mode is set to MIDI, the available rate values are selectable as note values, so the rate is synced to the sequencer tempo 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.

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 affects all the voices in sync. “Free running” means that the LFO cycles continuously, and does not 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.

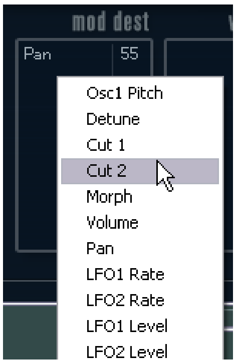
  For example, if LFO 2 is set to use Sample the resulting effect also depends on the speed and waveform of LFO 1.

Assigning LFO Modulation Destinations

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

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

   A pop-up menu appears in which 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, for example, 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 are all 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 menu.

Assigning LFO Velocity Destinations

You can also assign velocity-controlled LFO modulation, that is, the modulation is governed by how hard or soft you strike a key. Proceed as follows:

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

   A pop-up menu appears in which 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 are all 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 menu.

LFO modulation velocity control – an example:

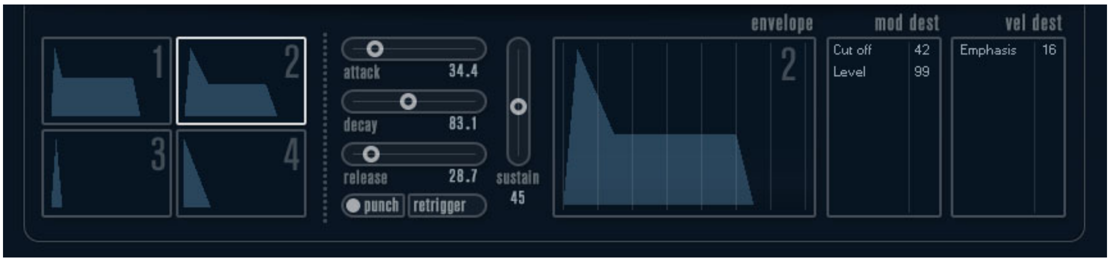
If you follow the steps above and select the Cut parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the Cut parameter is 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 changes when a key is pressed, when a key is held and finally when a key is released.



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 selects it and displays 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).
- 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 adjusts how the volume of the sound changes 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 takes is governed by the Attack setting. If the Attack is set to 0, the maximum value is reached instantly. If this value is raised, it takes 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 takes 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 rests 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 re-triggers 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 menu appears in which 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, for example, 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 are all 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 menu.

Assigning Envelope Velocity Destinations

You can also assign velocity-controlled Envelope modulation, that is, the modulation is governed by how hard or soft you strike a key. Proceed as follows:

1. Click in the “Vel Dest” box for one of the envelopes.
   A pop-up menu appears in which 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 are all 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 menu.

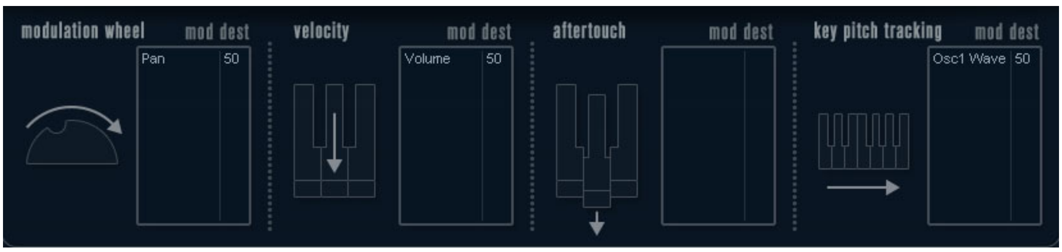
Envelope modulation velocity control – an example:

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 is 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 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 assignments.



The following controllers are available:

Controller	Description
Modulation Wheel	The modulation wheel on your keyboard can be used to modulate parameters.

Controller	Description
Velocity	Velocity controls 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 menu appears in which 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.

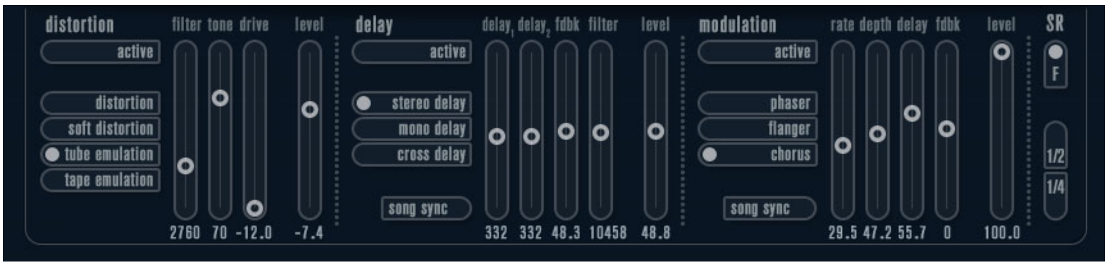
- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the [Enter] key.
  To enter negative values type a minus sign followed by the value.

3. Using the same basic method, you can add any number of modulation destinations for the controllers.
   They are all listed in the “Mod Dest” box for each controller.

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

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 low-pass filter and a high-pass filter with a cutoff frequency equal to the crossover frequency.
Tone	This parameter controls the relative amount of low-pass 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 low-pass 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 all-pass 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 is 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

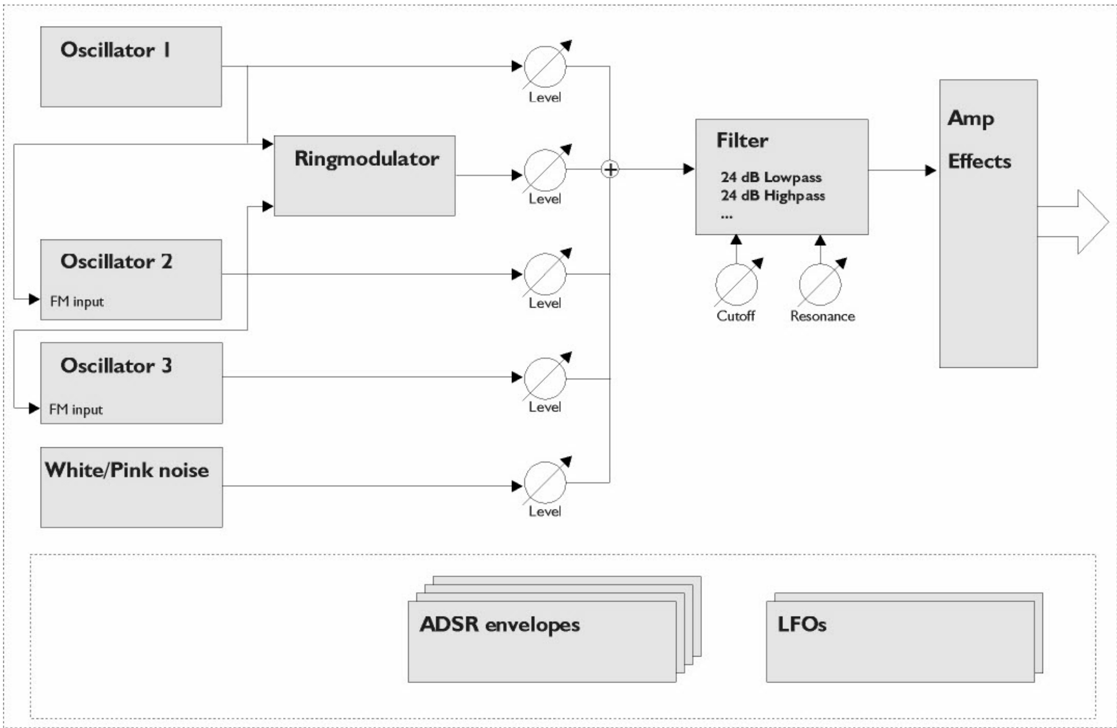
With these buttons you can change the sample rate. Lower sample rates basically reduce the high frequency content and sound quality, but the pitch is not altered. This is a great way to emulate the lo-fi sounds of older digital synths!

- If button “F” is active, the selected Part’s program plays back with the sample rate set in the host application.
- If button “1/2” is active, the selected Part’s program plays back with half the original sample rate.
- If button “1/4” is active, the selected Part’s program plays back with a quarter of the original sample rate.

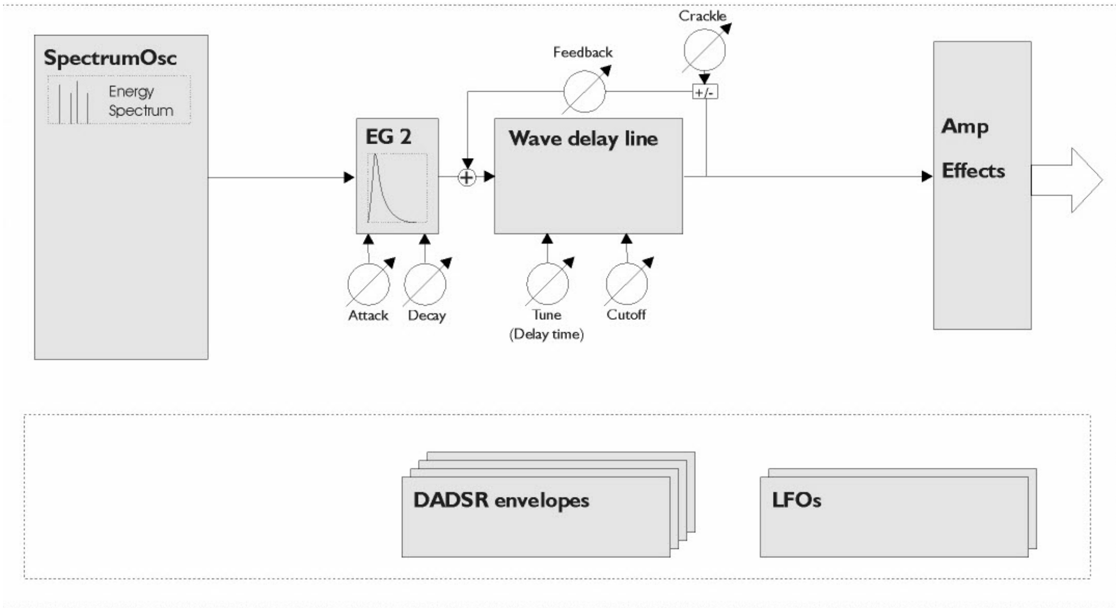
▪ A bonus effect of using lower sample rates is that it reduces the load on the computer CPU, allowing more simultaneous voices to be played, etc.

Diagrams

Prologue



Mystic



Spector