

Plug-in Reference



Cristina Bachmann, Heiko Bischoff, Christina Kaboth, Insa Mingers, Sabine Pfeifer, Kevin Quarshie, Benjamin Schütte

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Included Effect Plug-ins

The included plug-in effects are arranged according to their categories. This chapter is arranged in the same fashion, with the plug-ins listed in separate sections for each effect category.

NOTE

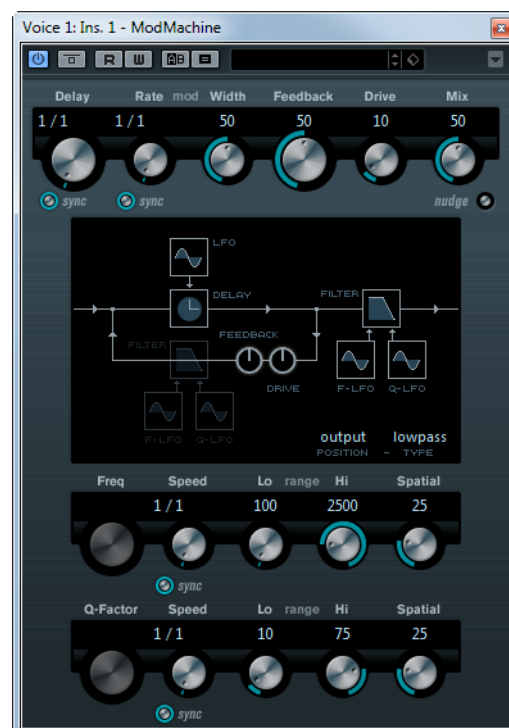
Most of the included effects are compatible with VST 3. For more information, see the Operation Manual.

Delay Plug-ins

ModMachine

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.

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



Delay

If tempo sync is on, this sets the base note value for the delay. If tempo sync is off, the delay time can be set freely.

Delay – Sync

Activates/Deactivates tempo sync for the **Delay** parameter.

Rate

Sets the base note value for tempo-syncing the delay modulation. If tempo sync is off, the rate can be set freely.

Rate – Sync

Activates/Deactivates tempo sync for the **Rate** parameter.

Width

Sets the amount of delay pitch modulation. This allows you to create 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 are distorted over time.

Mix

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

Nudge

Clicking this button once momentarily speeds up the audio coming into the plug-in, simulating the nudge command of analog tape machines.

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 in 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. You can choose between a **low-pass**, **band-pass**, and **high-pass** filter.

Filter Frequency LFO Modulation

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. If using tempo sync, this parameter sets the base note value for tempo syncing the modulation. If tempo sync is off, the speed can be set freely.

Speed – Sync

Activates/Deactivates tempo sync for the **Speed** parameter.

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 **Speed** is set to zero, these parameters are inactive and the filter frequency is controlled by the **Freq** parameter instead.

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.

Filter Resonance LFO Modulation

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. If tempo sync is activated, the resonance is controlled by the **Speed** and **Range** parameters.

Speed

Sets the speed of the filter resonance LFO modulation. If tempo sync is activated, this parameter sets the base note value for tempo syncing the modulation. If tempo sync is deactivated, the speed can be set freely.

Speed – Sync

Activates/Deactivates tempo sync for the **Speed** parameter.

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 deactivated and **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

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

	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	–



Delay

If tempo sync is activated, this is where you specify the base note value for the delay. If tempo sync is deactivated, the delay time can be set freely in milliseconds.

Sync

Activates/Deactivates tempo sync.

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. 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. The button below the knob activates/deactivates the filter.

Mix

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

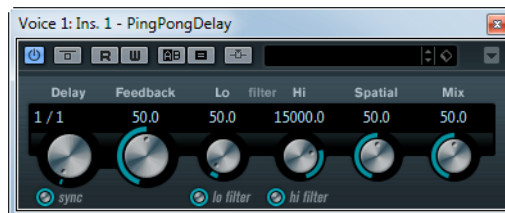
NOTE

If side-chaining is supported, the delay can also be controlled from another signal source via the side-chain input. If the side-chain signal exceeds the threshold, the delay repeats are silenced. If 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

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.

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



Delay

If tempo sync is activated, this is where you specify the base note value for the delay. If tempo sync is deactivated, the delay time can be set freely in milliseconds.

Sync

Activates/Deactivates tempo sync.

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. 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. The button below the knob activates/deactivates the filter.

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 this effect is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

NOTE

If side-chaining is supported, the delay can also be controlled from another signal source via the side-chain input. If the side-chain signal exceeds the threshold, the delay repeats are silenced. If 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

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

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



Delay 1 and 2

If tempo sync is activated, this is where you specify the base note value for the delay. If tempo sync is deactivated, the delay time can be set freely in milliseconds.

Sync

Activates/Deactivates tempo sync for the corresponding delay.

Feedback 1 and 2

Set the number of repeats for each delay.

Filter Lo 1 and 2

Affect the feedback loop of the effect signal and allow you to roll off low frequencies. The buttons below the knobs activate/deactivate the filter.

Filter Hi 1 and 2

Affect the feedback loop of the effect signal and allow you to roll off high frequencies. The buttons below the knobs activate/deactivate the filter.

Pan 1 and 2

Set the stereo position for each delay.

Mix 1 and 2

Set the level balance between the dry and the wet signal. If this effect is used as a send effect, set these controls to the maximum value as you can control the dry/effect balance with the send.

NOTE

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

AmpSimulator

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

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



Amplifier pop-up menu

Click the amplifier name shown at the top of the amp section to open this pop-up menu. It allows you to select an amplifier model. This 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.

Cabinet pop-up menu

Click the cabinet name shown at the top of the cabinet section to open this pop-up menu. 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 the values, enter a new value and press [Enter].

BitCrusher

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.

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



Mode

Allows you to select one of the four operating modes. 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, 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

Sets the output level.

Mix

Sets the balance between the output and the original audio signal.

DaTube

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

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



Drive

Sets the pre-gain of the amplifier. Use high values if you want an overdriven sound just on the verge of distortion.

Balance

Sets the balance between the signal processed by the **Drive** parameter and the dry input signal. For maximum drive effect, set this to its highest value.

Output

Adjusts the post-gain, or output level, of the amplifier.

Distortion

Distortion adds crunch to your tracks.

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



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.

Spatial

Changes the distortion characteristics of the left and right channels, thus creating a stereo effect.

Output

Sets the output level.

Grungelizer

Grungelizer adds noise and static to your recordings – like listening to a radio with bad reception, or a worn and scratched vinyl record.

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



Crackle

Adds crackle to create that old vinyl record sound.

RPM

When emulating the sound of a vinyl record, this switch lets you set the speed of the record in RPM (revolutions per minute).

Noise

Sets the amount of added static noise.

Distort

Adds distortion.

EQ

Cuts the low frequencies, and creates a hollow, lo-fi sound.

AC

Emulates a constant, low AC hum.

Frequency

Sets the frequency of the AC current (50 or 60 Hz), and thus the pitch of the AC hum.

Timeline

Sets the amount of overall effect.

Magneto II

This effect simulates the saturation and compression of recording on analog tape machines.

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



Saturation

Determines the amount of saturation and the generation of overtones. This leads to a small increase in input gain.

Saturation On/Off

Activates/Deactivates the saturation effect.

Dual Mode

Simulates the use of two tape machines.

Frequency Range Low/Hi

These parameters set the frequency range of the spectrum band to which the tape effect is applied.

For example, to avoid the saturation of lower frequencies, set the **Freq Low** value to 200Hz or 300Hz. To avoid the saturation of very high frequencies, set the **Freq Hi** parameter to values below 10kHz.

Tape Solo

Allows you to hear only the set frequency range including the tape simulation effect. This helps you to determine the appropriate frequency range.

HF-Adjust

Sets the amount of high frequency content of the saturated signal.

HF-Adjust On/Off

Activates/Deactivates the **HF-Adjust** filter.

VU Input/Output

The buttons determine whether the VU meter shows the level of the input or of the output signal. To set the input/output level, use the knob or enter a value in the value field.

SoftClipper

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

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



Input

Sets the pre-gain. Use high values if you want an overdriven sound just on the verge of distortion.

Mix

Sets the level balance between the dry and the wet signal.

Output

Sets the output level.

VST Amp Rack

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.

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



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**, **Master**, and **Configuration**.

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 **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.

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.

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.

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.

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.

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.

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.

Feedback – Sets the number of repeats for the delay.

Mix – Sets the level balance between the dry and the wet signal.

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.

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.

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 signal stays rather dry during loud or intensely played passages.

Overdrive

Drive – Overdrive creates a tube-like overdrive effect. The higher this value, the more harmonics are 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 this value, the more distortion is created.

Tone – Works as a filter effect on the added harmonics.

Level – Adjusts the output level.

Gate

Threshold – Determines the level at which the 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. This 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

Some parameters can be synchronized to the tempo of the host application.

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 **+** button that appears if 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. If an effect is active, the LED next to the button is lit.

NOTE

- Pre-effects and post-effects can be mono or stereo, depending on the track configuration.
- Using quick controls you can 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 are 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.

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 if 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, but you can also combine different amps and cabinets.

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.

Microphones

On the Microphones page, you can choose between 7 microphone positions. 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. If the equalizer is active, the LED next to the button is lit.
- To activate/deactivate an equalizer band, click the corresponding **Gain** knob. If 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. If 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. If 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.

Configuration

On the Configuration Page, you can specify whether you want to use VST Amp Rack in stereo or in mono mode.

- 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.
- To use the effect in mono-mode, make sure that the plug-in is inserted on a mono track, and activate the **Mono** button.

NOTE

In stereo mode, the effect requires more processing power.

View Settings

For the VST Amp Rack, you can choose between 2 views: 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.

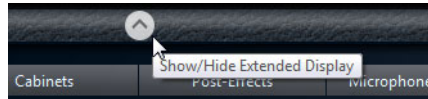
In the compact view, the page display is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

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

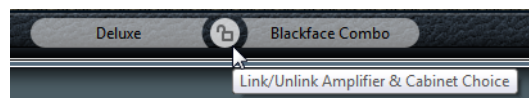
To toggle between the different views, click the down/up arrow button (Show/Hide Extended Display) at the top center of the plug-in frame.



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.

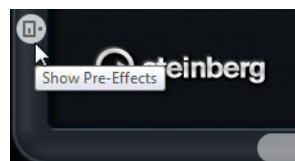
- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the **Link/Unlink Amplifier & Cabinet Choice** button. If you now select another amp model, the cabinet selection follows. However, 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

Brickwall Limiter

Brickwall Limiter ensures that the output level never exceeds a set limit.

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



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.

Threshold (-20 to 0 dB)

Only signal levels above the set threshold are processed.

Release (ms)

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

If this button is activated, Brickwall Limiter uses the channel with the highest level to analyze the input signal. If the button is deactivated, each channel is analyzed separately.

Detect Intersample Clipping

If this option is activated, Brickwall Limiter detects and limits signal levels between two samples to prevent distortion when converting digital signals into analog signals.

NOTE

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

This plug-in reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both.

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



Compressor features separate controls for **Threshold**, **Ratio**, **Attack**, **Hold**, **Release**, and **Make-Up Gain** parameters. It also 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.

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)

Sets the amount of gain reduction applied to signals over the set threshold. A ratio of 3:1 means that for every 3 dB the input level increases, the output level increases by only 1 dB.

Soft Knee

If this button is deactivated, signals above the threshold are compressed instantly according to the set ratio (hard knee). If **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.

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, 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 works better for percussive material with a lot of transient peaks.

Live

If 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. If **Live** mode is activated, there is no latency, which might be better for live processing.

DeEsser

This effect 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.

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



Close proximity microphone placement and equalizing can lead to situations where the overall sound is just right, but there is a problem with sibilants.

Reduction

Controls the intensity of the de-essing effect.

Threshold

If 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.

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

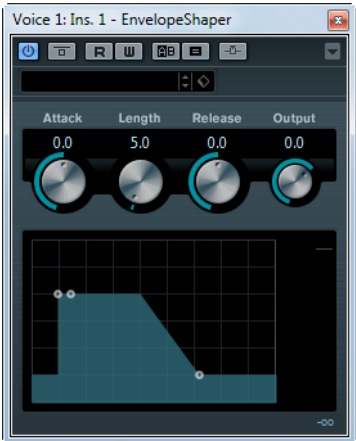
When recording a voice, the de-esser's position in the signal chain is usually located after the microphone pre-amp and before a compressor/limiter. This keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics.

EnvelopeShaper

This plug-in can be used to attenuate or boost the gain of the attack and release phase of audio material.

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

You can 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.



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.

NOTE

If side-chaining is supported, the effect can also be controlled from another signal source via the side-chain input. If 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

Expander reduces the output level in relation to the input level for signals below the set threshold. This is useful if 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.

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



Threshold (-60 to 0dB)

Determines the level where expansion kicks in. Signal levels below the threshold are affected. Signal levels above the threshold are not processed.

Ratio (1:1 to 8:1)

Determines the amount of gain boost applied to signals below the threshold.

Soft Knee

If this button is deactivated, signals below the threshold are expanded instantly according to the set ratio (hard knee). If **soft knee** is activated, the onset of expansion is more gradual, producing a less drastic result.

Attack (0.1 to 100ms)

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 2000ms)

Sets the time the applied expansion affects the signal below the threshold.

Release (10 to 1000ms or Auto mode)

Sets the time after which the gain returns to its 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.

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

If 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. If **Live** mode is activated, there is no latency, which might be better for live processing.

NOTE

If side-chaining is supported, the expansion can also be controlled from another signal source via the side-chain input. If 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

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

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



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 buttons (LP, BP, and HP)

If the **Side-Chain** button is activated, you can use these buttons to set the filter type to either low-pass, band-pass, or high-pass.

Side-Chain

This button activates the internal side-chain filter. The input signal can then be shaped according to the filter parameters. Internal side-chaining can be useful for tailoring how the gate operates.

Center (50 to 20000Hz)

If the **Side-Chain** button is activated, this sets the center frequency of the filter.

Q-Factor (0.01 to 10000)

If the **Side-Chain** button is activated, this sets the resonance of the filter.

Monitor

Allows you to monitor the filtered signal.

Attack (0.1 to 1000ms)

Sets the time after which the gate opens after being triggered.

NOTE

Deactivate the **Live** button to make sure that the gate is already open when a signal above the threshold is played back.


Hold (0 to 2000ms)

Determines how long the gate stays open after the signal drops below the threshold.

Release (10 to 1000ms or Auto mode)

Sets the time after which the gate closes (after the set **Hold** time). If the **Auto** button is activated, Gate automatically finds the best release setting for the audio material.

Range

Adjusts the attenuation of the gate when it is shut. If **Range** is set to , the gate is completely shut. The higher the value, the higher the level of the signal that passes through the shut gate.

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

If 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. If **Live** mode is activated, there is no latency, which is better for live processing.

NOTE

If side-chaining is supported, the gate can also be controlled from another signal source via the side-chain input. If 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

This plug-in is designed to ensure that the output level never exceeds a set output level, to avoid clipping in following devices.

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



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).

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 amount of time it takes for the gain to return to its original level. If the **Auto** button is activated, Limiter automatically finds an optimal release setting that varies depending on the audio material.

Maximizer

This plug-in 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.

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



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

If 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

This plug-in gates audio signals. The gate is triggered by MIDI notes.

	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. If a signal rises above the set level, the gate opens to let the signal through. Signals below the set level are silenced. MIDI Gate, however, is not triggered by threshold levels, but MIDI notes. Hence it needs both audio and MIDI data to function.

Attack

Determines how long it takes for the gate to open after receiving a signal that triggers it.

Hold

Determines 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 time. The higher the value, the more the attack time increases with high note velocities. Negative values result in shorter attack times with high velocities. If you do not want to use this parameter, set it to 0.

Note To Release

Determines to which extent the velocity values of the MIDI notes affect the release time. The higher the value, the more the release time increases. If you do not want to use this parameter, set it to 0.

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

Sets 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. The **Hold** and **Release** parameters are applied once a note-off signal has been received.

Setting Up MIDI Gate

To use MIDI Gate for your audio, you need an audio track and a MIDI track.

PROCEDURE

1. Select the audio track to which you want to apply MIDI Gate.
This can be recorded or realtime audio material from any audio track.
 2. Select **MIDI Gate** as an insert effect for the audio track.
 3. Select a MIDI track to control the MIDI Gate effect.
You can either play notes on a connected MIDI keyboard or use recorded MIDI notes.
 4. Open the **Output Routing** pop-up menu for the MIDI track and select **MIDI Gate**.
-

Applying MIDI Gate

PREREQUISITE

Set up the plug-in properly.

How to apply MIDI Gate depends on 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.

PROCEDURE

1. If you use realtime MIDI to trigger the plug-in, make sure the MIDI track is selected.
 2. Start playback.
 3. If you use realtime MIDI, play a few notes on your MIDI keyboard.
-

RESULT

The MIDI notes trigger the Gate. The plug-in gates the audio signals.

MultibandCompressor

MultibandCompressor allows a signal to be split into a maximum of four frequency bands, each with its own freely adjustable compressor characteristic.

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



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.

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. 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 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.

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 of the frequency bands.

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.

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.

Threshold (-60 to 0dB)

Determines the level where Multiband Compressor kicks in. Signal levels above the set threshold are affected. 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 1 dB.

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 passes through unprocessed.

Release (10 to 1000ms or Auto mode)

Sets the time after which the gain returns to its 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.

Output (-24 to 24dB)

Controls the total output level of MultibandCompressor.

Tube Compressor

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.

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



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

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 passes through unprocessed.

Release (10 to 1000ms 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 wet 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

Activates/Deactivates the internal side-chain filter. The input signal can then be shaped according to the filter parameters. Internal side-chaining is useful for tailoring how the compressor operates.

Filter section (LP, BP, and HP)

If 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.

Q-Factor

Sets the resonance or width of the filter.

Monitor

Allows you to monitor the filtered signal.

VintageCompressor

VintageCompressor is modeled after vintage type compressors.

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

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.



Input (-24 to 48dB)

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 24dB)

Sets the output gain.

Attack (0.1 to 100ms)

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)

If 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 1000ms or Auto mode)

Sets the time after which the gain returns to its original level. If the **Auto** button is activated, Vintage Compressor automatically finds an optimal release setting that varies depending on the audio material.

VU Meter

Shows the amount of gain reduction.

In/Out Meters

Show the highest peaks of all available input and output channels.

NOTE

If side-chaining is supported, the compression can also be controlled from another signal source via the side-chain input. If 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

VSTDynamics is an advanced dynamics processor. It combines three separate processors: Gate, Compressor, and Limiter, covering a variety of dynamic processing functions.

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



The window is divided into three sections, containing controls and meters for each processor. Activate the individual processors using the buttons **Gate**, **Compressor**, and **Limiter** at the bottom of the plug-in panel.

Gate Section

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

The following parameters are available:

Threshold (-60 to 0dB)

Determines the level where the 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

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 do not want it to, or to boost frequencies you want to accentuate, allowing for more control over the gate function.

LP (low-pass), BP (band-pass), HP (high-pass)

These buttons set the basic filter mode.

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.

Attack (0.1 to 100ms)

Sets the time after which the gate opens after being triggered.


Hold (0 to 2000ms)

Determines how long the gate stays open after the signal drops below the threshold level.

Release (10 to 1000ms or Auto mode)

Sets the time after which the gate closes (after the set **Hold** time). If the **Auto** button is activated, Gate automatically finds the best release setting for the audio material.

Range

Adjusts the attenuation of the Gate when it is shut. If **Range** is set to , the gate is completely shut. The higher the value, the higher the level of the signal that passes through the shut gate.

Input Gain Meter

Shows the input gain.

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.

Threshold (-60 to 0dB)

Determines the level where the compressor kicks in. Signal levels above the set threshold are affected. 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 1 dB.

Make-Up (0 to 24 dB)

Compensates for output gain loss, caused by compression. If the **Auto** button is activated, gain loss is compensated automatically.

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 amount of time after which the gain returns to the original level if 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 gain reduction in dB.

Limiter Section

The limiter ensures 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.

Output (-24 to 6dB)

Determines the maximum output level. Signal levels above the set threshold are affected. Signal levels below are left unaffected.

Soft Clip

If this button is activated, the limiter acts differently. If the signal level exceeds -6dB, 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 1000ms or Auto mode)

Sets the time after which the gain returns to the original level if 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).

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

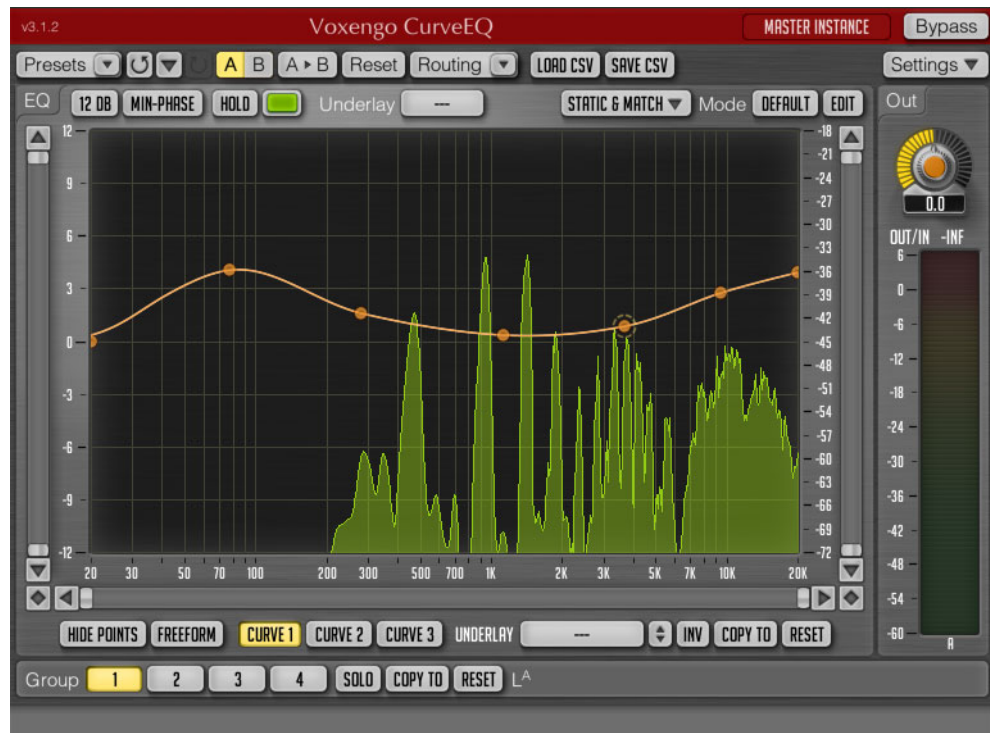
CurveEQ

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.

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

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



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.

General Control Bar



Presets selector

Allows you to store and restore custom settings.

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

By pressing the **A/B** button, you can switch between two plug-in states (A and B).

A>B (B>A)

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.

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.

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 20000Hz.

Settings

Allows you to change general settings.

EQ Top Control Bar



Equalizer dB gain range

Lets you change the maximum boost/decrease of frequencies per band.

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

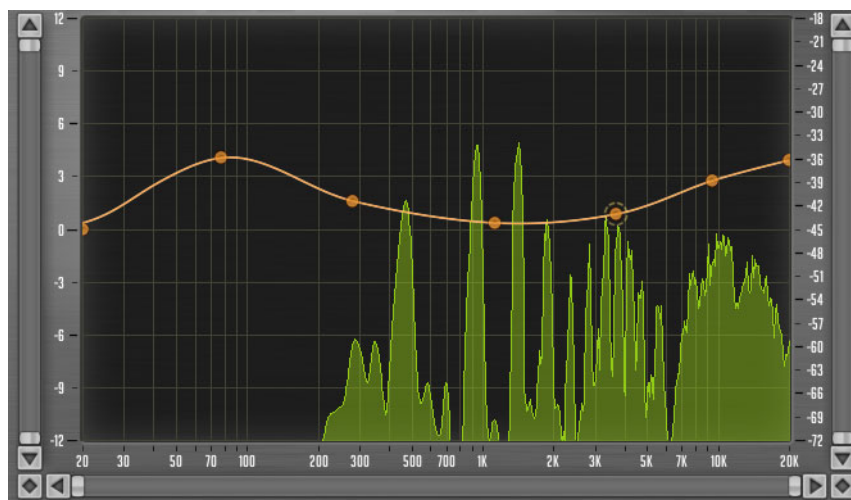
Allows you to select a mode for spectrum matching.

Edit

Opens the Spectrum Mode Editor.

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 to adjust the filter's gain and frequency.

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.

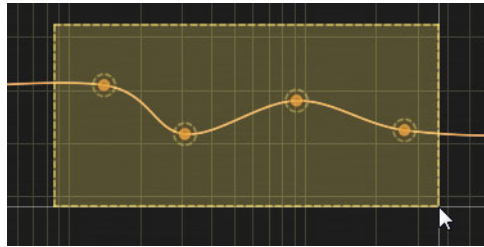
1.28K HZ -2.8 DB D#6 49 CENTS

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 mouse, 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 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 controls are available:

Up/down arrow button

Allows you to change the relationship 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.

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 [Shift] and dragging in the control surface with the mouse. 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



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 if 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 200Hz. 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.

DJ-EQ

This plug-in 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.

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



- To set the **Low**, **Mid**, and **Hi** frequency bands, click and drag the EQ points.
- To fine-adjust the **Low**, **Mid**, and **Hi** frequency bands, press [Shift] and drag the EQ points.
- To set a parameter to zero, press [Ctrl]/[Command] and click it.
- To change the **Gain** values, click them and move the mouse up or down.

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.

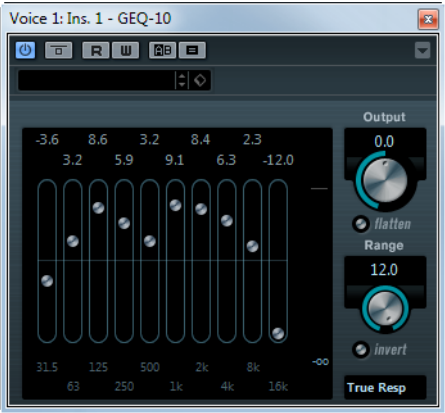
Output meter

Shows the overall output level.

GEQ-10/GEQ-30

These graphic equalizers are identical, except for the number of available frequency bands (10 and 30).

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



Each band can be attenuated or boosted by up to 12 dB, allowing for fine control of the frequency response. In addition, there are several preset modes available that can add color to the sound of GEQ-10/GEQ-30.

You can draw response curves in the main display by clicking and dragging with the mouse. Note that you have to click one of the sliders first before dragging across 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.

Output

Controls the overall gain of the equalizer.

Flatten

Resets all the frequency bands to 0 dB.

Range

Allows you to adjust how much a set curve cuts or boosts the signal.

Invert

Inverts the current response curve.

Mode pop-up menu

This pop-up menu allows you to set the filter mode that determines how the various frequency band controls interact to create the response curve, see below.

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.

True Response

Applies serial filters with an accurate frequency response.

Digi Standard

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.

VariableQ

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.

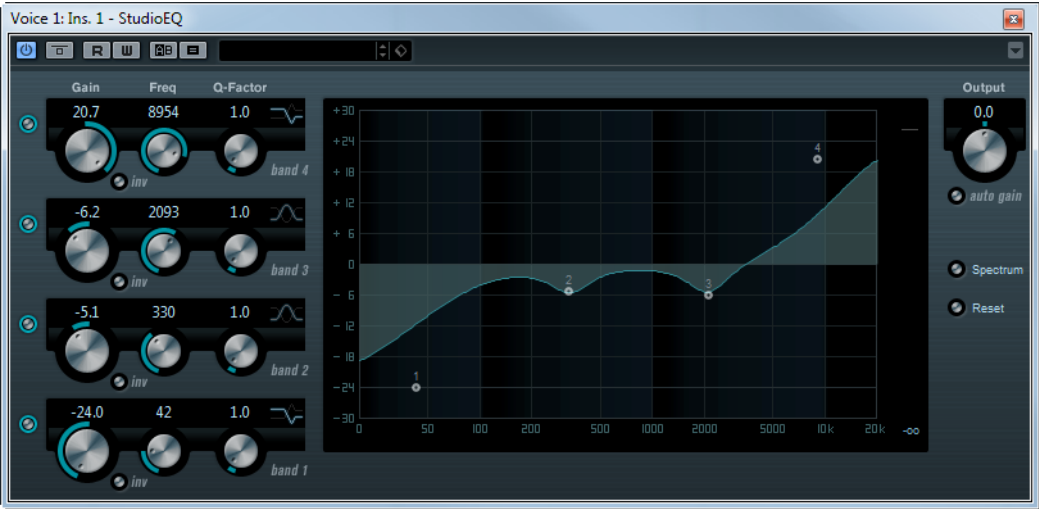
Resonant

Applies serial filters where a gain increase of one band lowers the gain in adjacent bands.

StudioEQ

Studio EQ 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 peak filter (band-pass), or as cut filter (low-pass/high-pass).

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



Band 1 Gain (-20 to +24dB)

Sets the amount of attenuation/boost for the low band.

Band 1 Inv

This 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 in the first place (set the filter to positive gain). After you have found the frequency of the noise, 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 choose between three types of shelving filters, a peak filter (band-pass), and a cut filter (lowpass/high-pass). If **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

This button inverts the gain value of the filter. See also 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

This button inverts the gain value of the filter. See also 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.

Band 4 Inv

This button inverts the gain value of the filter. See also 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 choose between three types of shelving filters, a peak filter, and a cut filter. If **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 +24 dB)

This knob on the top right of the plug-in panel allows you to adjust the overall output level.

Auto Gain

If 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.

Using Modifier Keys

When using the mouse to change the parameter settings, modifier keys can be used. When no modifier key is pressed and you drag an EQ point in the display, the Gain and Frequency parameters are adjusted simultaneously.

[Shift]

If you keep [Shift] pressed and drag the mouse, the Q-factor of the corresponding EQ band is modified.

[Alt]/[Option]

If you keep [Alt]/[Option] pressed and drag the mouse, the frequency of the corresponding EQ band is modified.

[Ctrl]/[Command]

If you keep [Ctrl]/[Command] pressed and drag the mouse, the gain value of the corresponding EQ band is modified.

Making Settings

PROCEDURE

1. Click the corresponding **On/Off** 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).
If 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 dragging points in the EQ curve display.
-

Filter Plug-ins

DualFilter

This plug-in filters out certain frequencies while allowing others to pass through.

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



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.

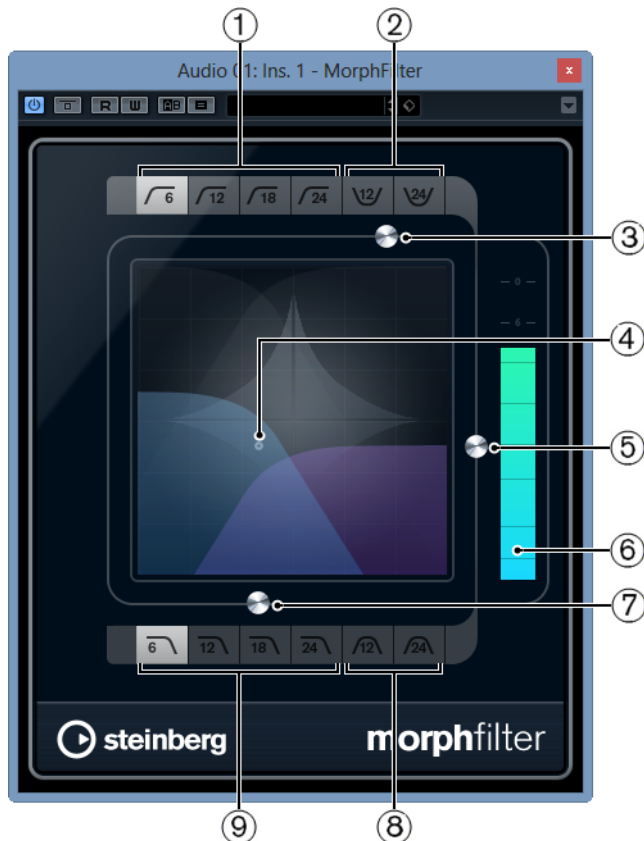
Resonance

Sets the sound characteristic of the filter. With higher values, a ringing sound is heard.

MorphFilter

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

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



1) **High Pass (6, 12, 18, 24dB per decade)**

Eliminates low-frequency signal components. Several filter slopes are available.

2) **Band Rejection (12, 24 dB/per Decade)**

Lets all frequencies pass, except those in the stop band. Several filter slopes are available.

3) **Resonance Factor**

Changes the resonance value of the filters.

4) **x/y control**

Adjusts the **Morph Factor** and the **Frequency** parameters simultaneously.

5) **Morph Factor**

Lets you mix the output between the two selected filters.

- 6) **Level meter**
Shows the output level, giving you an indication of how the filtering affects the overall level of the edited event.
- 7) **Frequency**
Adjusts the cutoff frequency of the filters.
- 8) **Band Pass (12, 24 dB per decade)**
Allows signals falling within a certain frequency range to pass through. Several filter slopes are available.
- 9) **Low Pass (6, 12, 18, 24 dB per decade)**
Eliminates high-frequency signal components. Several filter slopes are available.

PostFilter

This is the filter plug-in to use if you are working on a post-production mix, but 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.

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



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.

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 (20Hz to 1 kHz, or Off)

Allows you to eliminate low-frequency noise. The filter is inactive if the curve point is located all the way to the left.

Low Cut Slope

Allows you to choose a slope value for the low-cut filter.

Low Cut Preview

Use this button between the **Low Cut Freq** button and the graphical display 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.

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

This 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 frequency of the noise, you can use the **Invert** button to cancel it out.

Notch Q-Factor

Sets the width of the notch filter.

Notch Preview

Use the **Preview** button between the notch filter buttons and the graphical display 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 (3Hz to 20kHz, or Off)

This high-cut filter allows you to remove high-frequency noise. The filter is inactive if the curve point is located all the way to the right.

High Cut Slope

Allows you to choose a slope value for the high-cut filter.

High Cut Preview

This button between the **High Cut Freq** button and the graphical display allows you 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

This plug-in is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

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



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 Select** knob below the **Resonance** grid to select a new pattern. New patterns are all set to the same step value by default.

StepFilter Parameters

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

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

If the **Sync** button to the right of the **Sync** pop-up menu is activated (yellow), the pattern playback is synchronized to the project tempo.

Use the pop-up menu to set the pattern beat resolution, that is, the note values the pattern plays in relation to the tempo (1/1 to 1/32, straight, triplet, or dotted).

Output

Sets the overall volume.

Mix

Adjusts the mix between dry and wet signal.

Creating Variations for StepFilter Patterns

You can copy a pattern of the StepFilter to another pattern slot, which is useful for creating variations on a pattern of this plug-in.

PROCEDURE

1. Select the pattern that you want to copy.
2. Click the **Copy** button.

3. Select another pattern slot.
 4. Click the **Paste** button.
-

RESULT

The pattern is copied to the new slot, and can now be edited to create variations.

ToneBooster

ToneBooster is a filter that allows you to raise the gain in a selected frequency range. It is particularly useful if it is inserted before AmpSimulator in the plug-in chain, greatly enhancing the tonal varieties available.

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



Tone

Sets the center filter frequency.

Gain

Adjusts the gain of the selected frequency range by up to 24dB.

Width

Sets the resonance of the filter.

Mode selector

Sets the basic operational mode of the filter: **Peak** or **Band** Mode.

RELATED LINKS

[AmpSimulator on page 11](#)

WahWah

This plug-in is a variable slope band-pass filter that can be auto-controlled by a side-chain signal (if supported) or via MIDI modeling the well-known analog pedal effect.

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



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.

WahWah Parameters

Pedal

Controls the filter frequency sweep.

Pedal Control (MIDI)

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: 6 dB or 12 dB.

NOTE

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 is raised so that the plug-in acts as an auto-wah effect.

MIDI Control

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

If WahWah is used 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

UV22HR

This is an advanced version of Apogee's renowned UV22 dithering algorithm, capable of dithering to 8, 16, 20, or 24 bits.

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



8, 16, 20, 24bit

These buttons allow you to select the intended bit resolution for the final audio. As when using the internal dithering, it is important to set this to the correct resolution.

Hi

Applies a normal dither gain.

Lo

Applies a lower level of dither noise.

Auto black

If this option is activated, the dither noise is gated during silent passages.

IMPORTANT

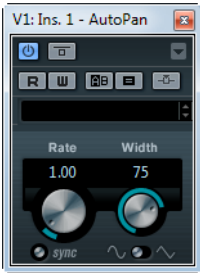
Dithering should always be applied post-fader on an output bus.

Modulation Plug-ins

AutoPan

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

	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	–



Rate

Sets the auto-pan speed.

Sync

Activates/Deactivates tempo sync.

Width

Sets the depth of the auto-pan effect, that is, how far out to the left/right speaker the sound should move.

Waveform Shape selector

Allows you to select the modulation waveform. **Sine** produces a smooth sweep. **Triangle** creates a ramp, that is, a sweep from one speaker to the other and then a quick jump back.

NOTE

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

This effect allows you to create a tremolo with or without an additional panning effect.

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



Waveform

These buttons set the modulation waveform.

Depth

Sets the amount of the effect. This can also be set by clicking and dragging in the graphical display.

Sync

The button above the **Speed** knob activates/deactivates tempo sync.

Speed

If tempo sync is activated, 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 deactivated, the tremolo speed can be set freely with the **Speed** knob.

Mono

If this button is activated, Chopper acts as a tremolo effect only. If this button is deactivated, the modulation waveforms of the left and the right channel are phase-shifted, creating an additional panning effect.

Mix

Sets the level balance between the dry and the wet signal. If this plug-in is used as a send effect, set this parameter to the maximum value.

Chorus

This plug-in is a single-stage chorus effect. It works by doubling the audio that is sent into it with a slightly detuned version.

	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	–



Rate

If tempo sync is deactivated, this sets the sweep rate.

Sync

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.

Spatial

Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.

Mix

Sets the level balance between the dry signal and the wet signal. If the effect 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 chorus sweep. A sine and triangle waveform are available.

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.

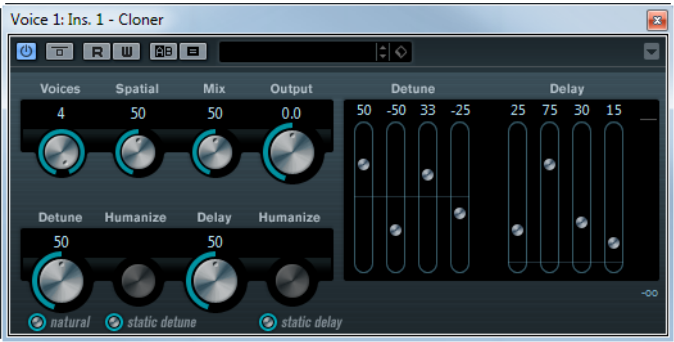
NOTE

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

This plug-in adds up to 4 detuned and delayed voices to the signal, for rich modulation and chorus effects.

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



Voices

Sets the number of voices (up to 4). For each voice, a **Detune** and a **Delay** slider are available in the right half of the panel.

Spatial

Spreads the 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 the effect is used as a send effect, this should be set to the maximum value.

Output

Reduces or increases the output gain by up to 12dB.

Detune slider (1 to 4)

Controls the relative detune amount for each voice. Positive and negative values can be set. 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.

Natural

Activate this button to change the pitch algorithm.

Detune – Humanize

Controls the amount of detune variation if **Static Detune** is deactivated. With **Humanize**, the detune is continuously modulated for a more natural effect.

Static Detune

This button activates/deactivates 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.

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.

Static Delay

This button activates/deactivates the **Static Delay** function. If activated, the set delay amount is static, and the **Humanize** knob is grayed out.

Flanger

This is a classic flanger effect with added stereo enhancement.

	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	–



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

Activates/Deactivates tempo sync.

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.

Manual

The button activates/deactivates the **Manual** function. If activated, the flanger sweep is static, that is, no modulation takes place.

The knob allows you to change the sweep position manually if the **Manual** button is deactivated.

Filter Lo/Hi

Allow you to roll off low and high frequencies of the effect signal.

NOTE

If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope.

Metalizer

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

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



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

Activates/Deactivates filter modulation. If it is deactivated, Metalizer works as a static filter.

Mono

If this is activated, the output of Metalizer is mono.

Speed

If tempo sync is activated, 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 deactivated, the modulation speed can be set freely with the **Speed** knob.

Sync

Activates/Deactivates tempo sync.

Output

Sets the overall volume.

Mix

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

Phaser

This plug-in produces the well-known swooshing phasing effect with additional stereo enhancement.

	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	–



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

Activates/Deactivates tempo sync.

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

If you are 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 this effect is used as a send effect, set this to the maximum level as you can control the dry/effect balance with the send.

Manual

The button activates/deactivates the **Manual** function. If activated, the flanger sweep is static, that is, no modulation takes place.

The knob allows you to change the sweep position manually if the **Manual** button is deactivated.

Filter Lo/Hi

Allow you to roll off low and high frequencies of the effect signal.

NOTE

If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. If 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

RingModulator can produce complex, bell-like enharmonic sounds.

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



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.

Oscillator

LFO Amount

Controls how much the oscillator frequency is affected by the LFO.

Env. Amount

Controls how much the oscillator frequency is affected by the envelope (which is triggered by the input signal). Left of center, a loud input signal decreases the oscillator pitch, whereas right of center the oscillator pitch increases if it is fed a loud input.

Waveform buttons

Allow you to select the oscillator waveform: square, sine, saw, or triangle.

Range slider

Determines the frequency range of the oscillator in Hz.

Frequency

Sets the oscillator frequency ± 2 octaves within the selected range.

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.

Env. Amount

Controls how much the input signal level – via the envelope generator – affects the LFO speed. With negative values, a loud input signal slows down the LFO, whereas positive values speed it up at loud input signals.

Waveform

Allows you to select the LFO waveform: square, sine, saw, or triangle.

Invert Stereo

Inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.

Envelope Generator

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.

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

If this button is activated, the L and R input signals are merged, and produce the same envelope output level for both oscillator channels. If the button is deactivated, each channel has its own envelope that affects the two channels of the oscillator independently.

Final Settings

Output

Sets the overall volume.

Mix

Sets the level balance between the dry signal and the wet signal.

Rotary

This modulation effect simulates the classic effect of a rotating speaker.

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



A rotary speaker cabinet features speakers rotating at variable speeds to produce a swirling chorus effect, commonly used with organs.

Speed

Speed selector (stop/slow/fast)

Allows you to control the speed of the Rotary in three steps.

Speed Change Mode

If this is set to the left, the speed selector settings are taken into account. If this is set to the right, you can modulate the speed with the **Speed Mod** knob and/or with a MIDI controller that you can select on the **Speed Mod Control (MIDI)** pop-up menu.

Speed Mod

If the **Speed Change Mode** setting is set to the right side, this knob allows you to modulate the rotary speed.

Speed Mod Control (MIDI)

This pop-up menu allows you to select the MIDI controller that controls the Rotary speed. If you do not want to use MIDI realtime control, set this to **Automation**. If you select **Pitchbend** as MIDI controller, the speed changes with an up or down flick of the bender. If other MIDI controllers are used, the speed changes at MIDI value 64.

Additional settings

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.

Fast

Allows for a fine adjustment of the high rotor **fast** speed.

Accel.

Allows for a fine adjustment of the high rotor acceleration time.

Amp Mod

Controls the high rotor amplitude modulation.

Freq Mod

Controls the high rotor frequency modulation.

Bass

Slow

Allows for a fine adjustment of the low rotor **slow** speed.

Fast

Allows for a fine adjustment of the low rotor **fast** speed.

Accel.

Allows for a fine adjustment of the low rotor acceleration time.

Amp Mod

Adjusts the modulation depth of the amplitude.

Level

Adjusts the overall bass level.

Microphones

Phase

Adjusts the phasing amount in the sound of the high rotor.

Angle

Sets the simulated microphone angle. 0 = mono, 180 = a mic on each side.

Distance

Sets the simulated microphone distance from the speaker in inches.

Final Settings

Output

Adjusts the overall output level.

Mix

Adjusts the mix between dry and wet signal.

Directing MIDI to the Rotary

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

- If Rotary is used as 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** pop-up menu, MIDI is directed to the plug-in from the selected track.

StudioChorus

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

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



- Rate**
Sets the chorus sweep rate.
- Sync**
Activates/Deactivates tempo sync.
- Width**
Determines the extent of the chorus effect. Higher settings produce a more pronounced effect.
- Spatial**
Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
- Mix**
Sets the level balance between the dry signal and the wet signal. If the effect 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 chorus sweep. A sine and a triangle waveform are available.
- 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.

NOTE

If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. If 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

In this ring modulator effect, 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.

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



Waveform

These buttons allow you to select a pitch modulation waveform.

Tone

Sets the frequency (pitch) of the modulating oscillator.

Depth

Governs the depth of the pitch modulation.

NOTE

Clicking and dragging in the display allows you to adjust **Tone** and **Depth** at the same time!

Speed

If tempo sync is activated, 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 deactivated, the modulation speed can be set freely with the **Speed** knob.

Sync

The button above the **Speed** knob activates/deactivates tempo sync.

On

Activates/Deactivates the modulation of the pitch parameter.

Mono

Governs whether the output is stereo or mono.

Output

Adjusts the output level of the effect.

Mix

Sets the level balance between the dry and the wet signal.

Tremolo

Tremolo produces amplitude modulation.

	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	–



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

Activates/Deactivates tempo sync.

Depth

Governs the depth of the amplitude modulation.

Spatial

Adds a stereo effect to the modulation.

Output

Adjusts the output volume.

NOTE

If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. If 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

This effect creates pitch modulation.

	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	–



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

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.

NOTE

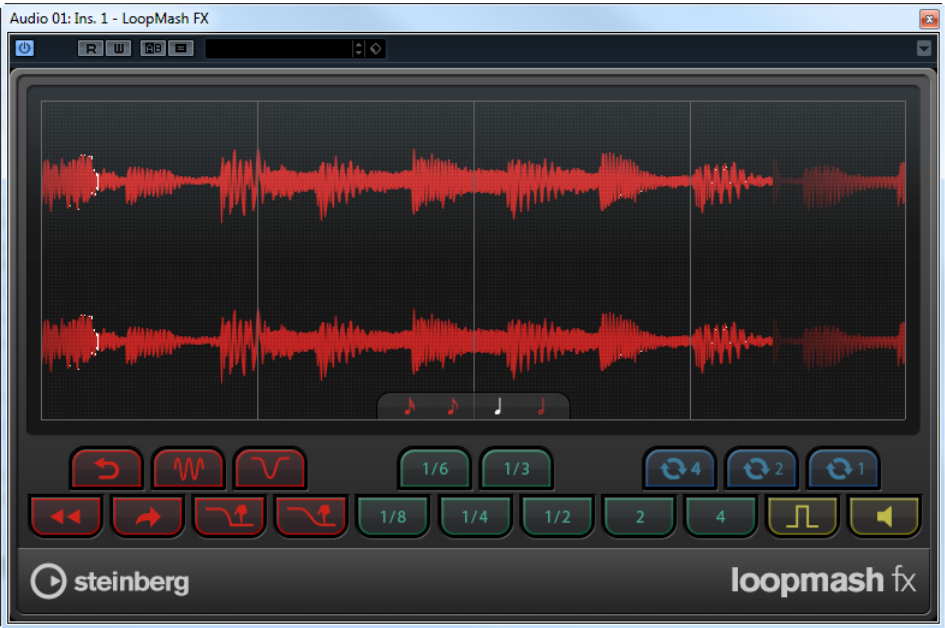
If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the Operation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

Other Plug-ins

LoopMash FX

LoopMash FX is a live performance effect offering DJ effects that can be controlled by a MIDI keyboard.

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



Quantize Note



Sets the note value on which the quantize grid of the effects is based.

Performance Controls

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 pressed.

NOTE

The effects can be automated. The automation of effect parameters is described in the Operation Manual.

Backspin



Simulates a turntable backspin.

Reverse



Plays the slice in reverse.

Tapestart



Simulates a tapestart, that is, speeds the slice up.

Scratch



Plays the slice as if scratched.

Slowdown



Applies a slowdown.

Tapestop 1



Simulates a tapestop, that is, slows the slice down, first lightly then abruptly.

Tapestop 2



Simulates a tapestop, that is, slows the slice down, smoothly.

Stutter



Plays only the initial portion of a slice and repeats it 2, 3, 4, 6, or 8 times during one slice length.

Slur



Stretches the slice over 2 or 4 slice lengths.

Cycle



Sets up a short cycle over 4, 2, or 1 slice. This short cycle is always set up within the loop range that is set in the ruler. Setting up a cycle over 1 slice means that this slice is repeated until you release the button.

Staccato



Shortens the slice.

Mute



Mutes the slice.

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 use the virtual keyboard for triggering the performance controls (for information about the virtual keyboard see the Operation Manual).

Controlling LoopMash FX with a MIDI Keyboard

PROCEDURE

1. Create an audio track and import an audio file, a drum loop, for example.
 2. Insert LoopMash FX as an insert effect.
 3. Play back the drum loop in a cycle.
 4. Create a MIDI track.
 5. On the **Output Routing** pop-up menu for the MIDI track, select **LoopMash FX**.
-

RESULT

You can trigger the different LoopMash FX performance effects with a MIDI keyboard.

VST Connect CUE Mix

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

This plug-in is described in detail in the separate PDF document “VST Connect SE”.

VST Connect SE

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

This plug-in is described in detail in the separate PDF document “VST Connect SE”.

Pitch Shift Plug-ins

Octaver

This plug-in can generate two additional voices that track the pitch of the input signal one octave and two octaves below the original pitch. Octaver is best used with monophonic signals.

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



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 generated signal one octave below the original pitch. A setting of 0 means that the voice is muted.

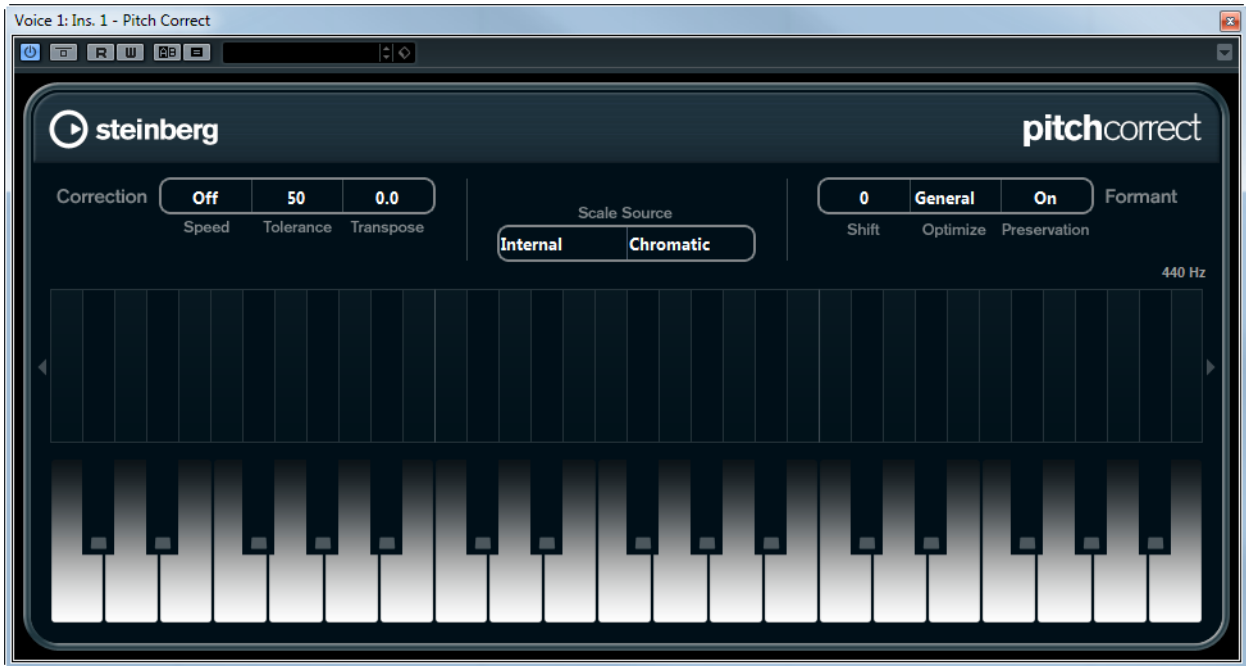
Octave 2

Adjusts the level of the generated signal two octaves below the original pitch. A setting of 0 means that the voice is muted.

Pitch Correct

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

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



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.

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.

Tolerance

Determines the sensitivity of analysis. A low Tolerance value lets Pitch Correct find pitch changes quickly. If the Tolerance value is high, pitch variations in the audio (for example, vibrato) are not immediately interpreted as note changes.

Transpose

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.

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.

- **Chromatic:** The audio is pitched to the closest semitone.
- **Major/Minor:** The audio is pitched to the major/minor scale specified on 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.

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

You have to assign the audio track as the output of your MIDI track and the **Speed** parameter has to be set to a value other than **Off**.

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

You have to assign the audio track as the output of your MIDI track and the **Speed** parameter has to be set to a value other than **Off**.

Chord Track – Chords

Select this option if you want the audio to be shifted to target chords, using the chord information from the Chord track.

NOTE

You have to add a MIDI track in addition to the Chord track and assign Pitch Correct as output of the MIDI track.

Chord Track – Scale

Select this option if you want the audio to be shifted to a scale of target pitches, using the scale information from the Chord track.

NOTE

You have to add a MIDI track in addition to the Chord track and assign Pitch Correct as output of the MIDI track.

Formant

Shift

Changes the natural timbre, that is, the characteristic frequency components of the source audio.

Optimize (General, Male, Female)

Allows you to specify the sound characteristics of the sound sources.

General is the default setting, **Male** is designed for low pitches and **Female** for high pitches.

Preservation (On/Off)

If 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. If this parameter is set to **On**, the formants are kept, maintaining the character of the audio.

Master Tuning

Detunes the output signal.

PitchDriver

This effect allows for sound design in postproduction. It 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.

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



Detune

Lets you detune the pitch of the incoming audio.

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

The created panorama effect can be unstable. For a stable panorama, deactivate the **Spatial** parameter. Then, 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

REVeLation

This effect produces a high-quality algorithmic reverb with early reflections and reverb tail.

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



The early reflections are responsible for the spatial impression in the first milliseconds of the reverb. For emulating different rooms, you can choose between different early reflections patterns and adjust their size. The reverb tail, or late reverberation, offers parameters for controlling the room size and the reverb time. You can adjust the reverb time individually in 3 frequency bands.

Predelay

Determines the amount of time between the dry signal and the onset of the reverb. With higher predelay values, you can simulate larger rooms.

Early Reflections

Here, you select an early reflections pattern. The early reflections pattern contains the most important delays that deliver the key information for the spatial impression of the room.

ER/Tail Mix

Sets the level balance between the early reflections and the reverb tail. At a setting of 50%, early reflections and tail have the same volume. Settings below 50% raise the early reflections and lower the tail, as a result the sound source moves towards the front of the room. Settings above 50% raise the tail and lower the early reflections, as a result the sound source moves towards the back of the room.

Size

Adjusts the length of the early reflections pattern. At a setting of 100%, the pattern is applied with its original length and the room sounds the most natural. At settings below 100%, the early reflections pattern is compressed and the room is perceived smaller.

Low Cut

Attenuates the low frequencies of the early reflections. The higher this value, the less low frequencies are present in the early reflections.

High Cut

Attenuates the high frequencies of the early reflections. The lower this value, the less high frequencies the early reflections will have.

Delay

Delays the onset of the reverb tail.

Room Size

Controls the dimensions of the simulated room. At a setting of 100%, the dimensions correspond to a cathedral or a large concert hall. At a setting of 50%, the dimensions correspond to a medium-sized room or studio. Settings below 50% simulate the dimensions of small rooms or a booth.

Main Time

Controls the overall reverb time of the tail. The higher this value, the longer the reverb tail will decay. At a setting of 100%, the reverb time is infinitely long. The **Main Time** parameter also represents the mid band of the reverb tail.

High Time

Controls the reverb time for the high frequencies of the reverb tail. With positive values, the decay time of the high frequencies is longer. With negative values, it is shorter. Frequencies are affected depending on the **High Freq** parameter.

Low Time

Controls the reverb time for the low frequencies of the reverb tail. For positive values, low frequencies decay longer and vice versa. Frequencies will be affected depending on the **Low Freq** parameter.

High Freq

Sets the cross-over frequency between the mid and the high band of the reverb tail. You can offset the reverb time for frequencies above this value from the main reverb time with the **High Time** parameter.

Low Freq

Sets the cross-over frequency between the low and the mid band of the reverb tail. The reverb time for frequencies below this value can be offset from the main reverb time with the **Low Time** parameter.

Shape

Controls the attack of the reverb tail. At a setting of 0%, the attack is more immediate, which is a good setting for drums. The higher this value, the less immediate the attack.

Density

Adjusts the echo density of the reverb tail. At a setting of 100%, single reflections from walls cannot be heard. The lower this value, the more single reflections can be heard.

High Cut

Attenuates the high frequencies of the reverb tail. The lower this value, the less high frequencies the reverb tail will have.

Width

Adjusts the output of the reverb signal between mono and stereo. At a setting of 0%, the output of the reverb is mono, at 100% it is stereo.

Mix

Sets the ratio between the dry and the wet signal.

Lock

Activate this button (padlock symbol) next to the **Mix** parameter to lock the dry/wet balance while browsing through the available presets.

Modulation

Modulation allows you to enrich the reverb tail through subtle pitch modulations.

Modulation Enable

Activates/Deactivates the chorusing effect.

Modulation Rate

Specifies the frequency of the pitch modulation.

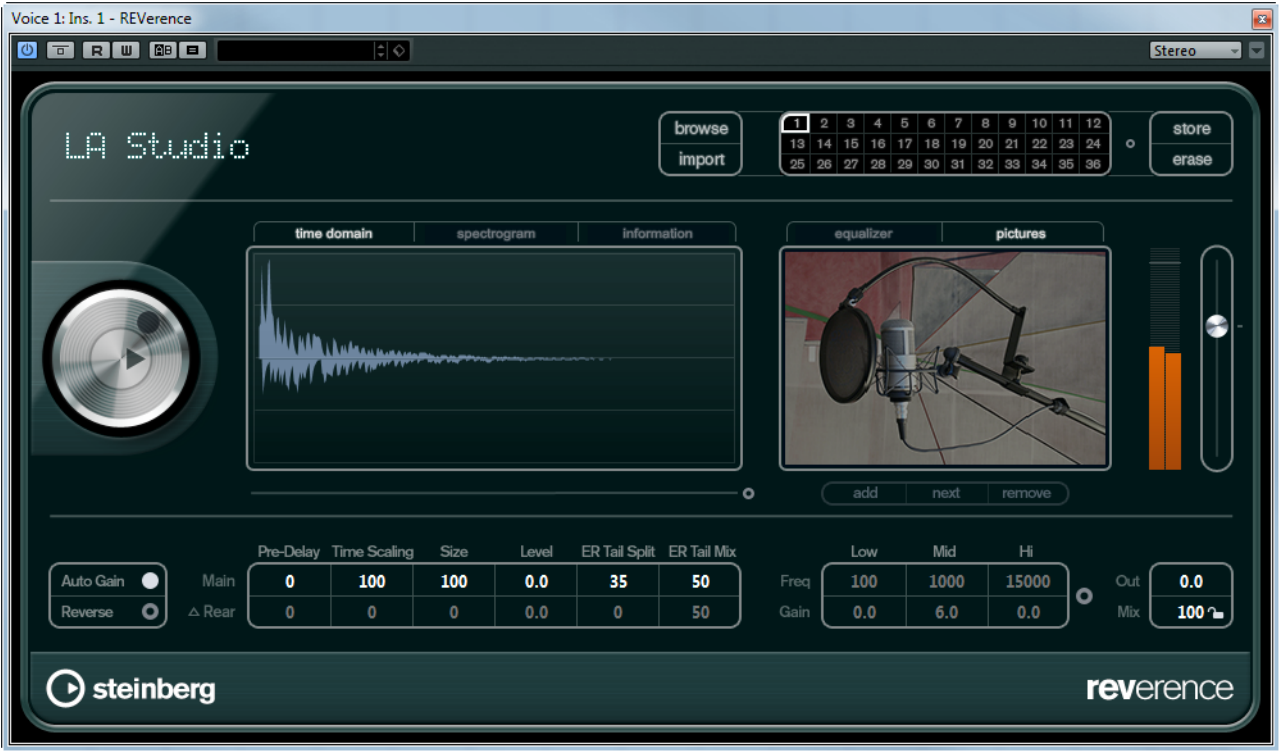
Modulation Depth

Sets the intensity of the pitch modulation.

REVerence

REVerence is a convolution tool that allows you to apply room characteristics (reverb) to the audio.

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



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.

NOTE

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.

Program Matrix

A program is the combination of an impulse response and its settings. These include reverb settings, EQ settings, pictures, and output settings. The program matrix allows you to load programs and to view the name of the impulse response.



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

This button opens a browser window showing the available programs. If you select a program in the browser, it is loaded into the active program slot. To filter the list of impulse responses by room type or the number of channels, for example, activate the **Filters** section (by clicking the **Set Up Window Layout** button at the bottom left of the window).

Import

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.

Program slots

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. If you move the mouse over a used slot, the corresponding program name is displayed below the active program name.

Smooth Parameter Changes

This 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

Stores the active impulse response and its settings as a program.

Erase

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.

Preset



A REVerence preset contains all settings and parameters for the plug-in, that is, all 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.

Program



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:

- 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.

RELATED LINKS

[Reverb Settings on page 100](#)

[EQ Settings on page 102](#)

[The Pictures Section on page 103](#)

[Custom Impulse Responses on page 105](#)

Setting up Programs

PROCEDURE

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.
3. In the browser, 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 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 by following the steps above.

NOTE

If you want to use your set of programs in other projects, save your settings as a plug-in preset.

RELATED LINKS

[Importing Impulse Responses on page 105](#)

Reverb Settings

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

		Pre-Delay	Time Scaling	Size	Level	ER Tail Split	ER Tail Mix
Auto Gain <input checked="" type="radio"/>	Main	0	100	100	0.0	35	50
Reverse <input type="radio"/>	Δ Rear	0	0	0	0.0	0	50

Main

All values shown in the top row apply to all speakers or to the front channels if you are working with surround tracks.

Rear

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.

Auto Gain

If this button is activated, the impulse response is automatically normalized.

Reverse

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

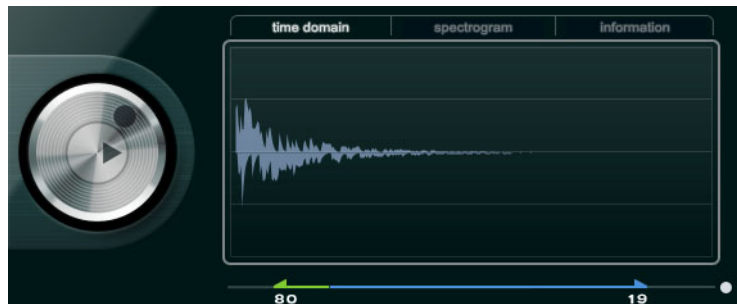
Sets a split point between the early reflections and the tail, allowing you to determine where the reverb tail begins. A value of 60 means that the early reflections are heard for 60ms.

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).



Play

When clicking this 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.

Time Scaling

This wheel lets you adjust the reverb time.

Time Domain

This display shows the waveform of the impulse response.

Spectrogram

This 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

This display shows additional information, such as the name of the program and the loaded impulse response, the number of channels, the length, and Broadcast Wave File information.

Activate Impulse Trimming

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

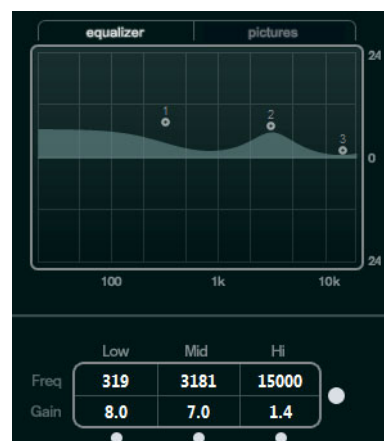
This slider allows you to trim the start and end of the impulse response. Drag the front handle to trim the start of the impulse response, or the end handle to trim the reverb tail.

NOTE

The impulse response is cut without any fading.

EQ Settings

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



EQ curve

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

This button to the right of the EQ parameters activates the EQ for the effect plug-in.

Low Shelf On

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

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

Activates the high shelf filter that boosts or attenuates frequencies above the cutoff frequency by the specified amount.

Hi Freq (5000 to 20000)

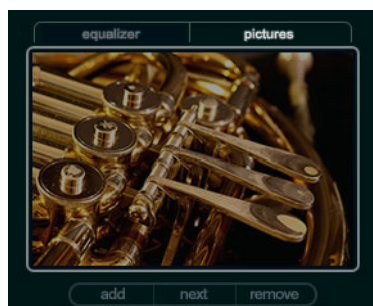
Sets the frequency of the high band.

Hi Gain (-24 to +24)

Sets the amount of attenuation/boost for the high band.

The Pictures Section

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.



NOTE

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

Add

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

If several pictures are loaded, you can click this button to display the next image.

Remove

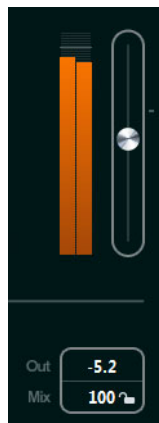
Deletes the active picture.

NOTE

This does not remove the graphics file from your hard disk.

Output Settings

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



Output activity meter

Indicates the overall level of the impulse response and its settings.

Output

This 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.

Lock

Activate this button (padlock symbol) next to the **Mix** parameter to lock the dry/wet balance while browsing through the available presets and programs.

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

PROCEDURE

1. In the program matrix, click **Import**.
 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.
-

RESULT

The program slot turns blue, indicating that a program is loaded.

NOTE

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.

AFTER COMPLETING THIS TASK:

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

Reading Order of Input Channels

REVerence reads input channels in the following order.

Number 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)
4	LL/LR/RL/RR (if inserted on a track with a stereo configuration)
5	L/R/C/LS/RS
6	L/R/C/LFE/LS/RS (LFE is being ignored.)

True Stereo

Impulse responses recorded as true-stereo files allow 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

NOTE

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).

REVerence automatically works in true-stereo mode if 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.

NOTE

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 to 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.

PROCEDURE

1. Transfer your 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.
 5. Click **Open**.
-

RESULT

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

IMPORTANT

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

RoomWorks is a highly adjustable reverb plug-in for creating realistic room ambiance 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.

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



Input Filters

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.

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.

Lo Gain

Controls the amount of boost or attenuation for the low-shelving filter.

Hi Gain

Controls the amount of boost or attenuation for the high-shelving filter.

Reverb Character

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.

Size

Alters the delay times of early reflections to simulate larger or smaller spaces.

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.

Width

Controls the width of the stereo image. 100% gives you full stereo reverb. At 0%, the reverb is all in mono.

Variation

Clicking this button generates a new version of the same reverb program using altered reflection patterns. This is helpful if certain sounds are causing odd ringing or undesirable results. Creating a new variation often solves these issues. There are 1000 possible variations.

Hold

Activating this button freezes the reverb buffer in an infinite loop. You can create some interesting pad sounds using this feature.

Damping

Lo Freq

Determines the frequency below which low-frequency damping occurs.

High Freq

Determines the frequency above which high-frequency damping occurs.

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.

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.

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.

Release

Determines how long after a signal peak the reverb can be heard before being cut off, similar to a release time of a gate.

Output

Mix

Determines the balance of dry signal and wet signal. If RoomWorks is used as an insert effect for an FX channel, you most likely want to set this to 100% or use the **wet only** button.

Wet only

This button defeats the **Mix** parameter, setting the effect to 100% wet or affected signal. This button should normally be activated if RoomWorks is used as a send effect for an FX channel or a group channel.

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%).

Export

Determines if during audio export RoomWorks uses the maximum CPU power for the highest quality reverb. During export, you may want 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 meter

Indicates the level of the output signal.

RoomWorks SE

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

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



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.

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

Sets the level balance between the dry signal and the wet 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

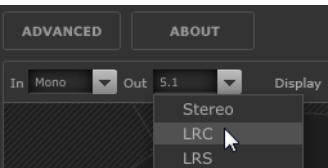
Anymix Pro

	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/downmix 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 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.

NOTE

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

NOTE

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

	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.

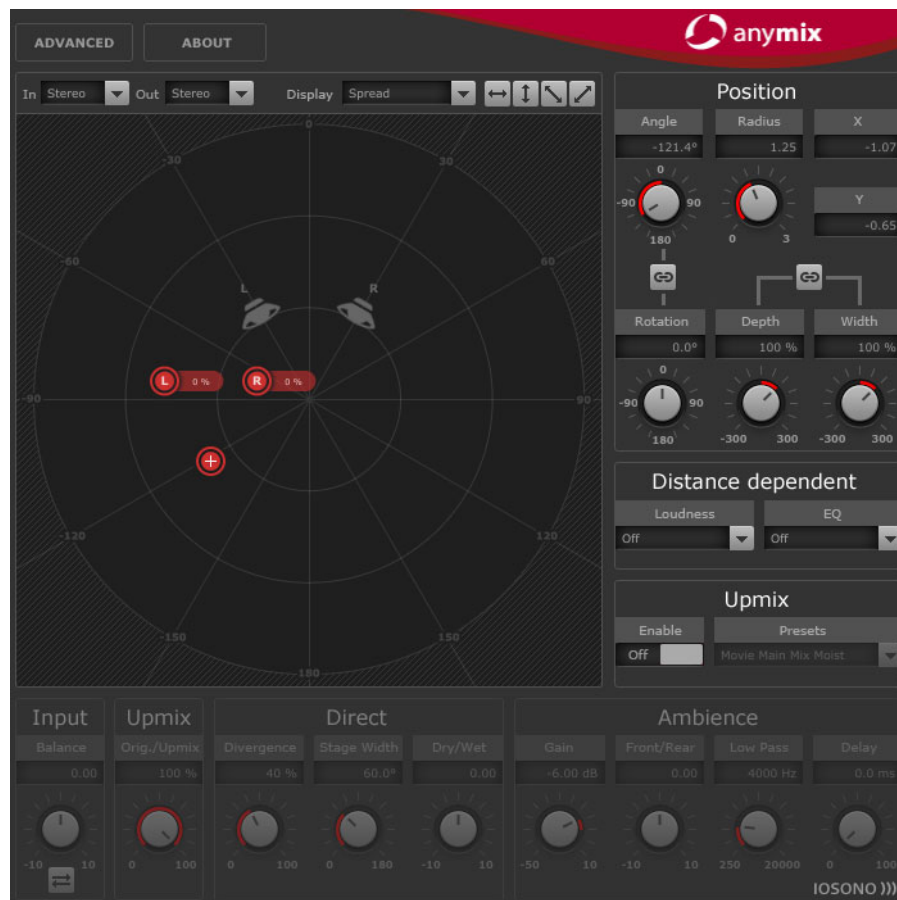
The Plug-in Panel

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.

NOTE

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

Panning Mode



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.

NOTE

The distance between the input channels automatically shrinks if they 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.

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.

NOTE

To fine-adjust the parameters, press [Shift] while using the controls.

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.



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

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.

IMPORTANT

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.

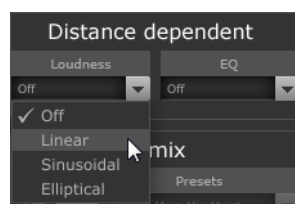


NOTE

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.



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.

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.

NOTE

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.

RELATED LINKS

[Advanced Options on page 120](#)

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.

IMPORTANT

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.

IMPORTANT

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.

NOTE

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.



NOTE

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 are automatically reloaded when the upmix is disabled.

Upmix Presets

Anymix Pro comes with a set of preconfigured upmix presets. If 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**
- **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**
- **Output Gain.**

NOTE

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

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

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

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 if 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 if 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.

NOTE

Matrix decoding is only applied in upmix mode.

LFE Parameters

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.

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 = 12dB/octave
- 3rd order = 18dB/octave
- 4th order = 24dB/octave

NOTE

The amount of LFE can be adjusted for each input channel individually.

NOTE

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

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.

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



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

Switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when creating an artificial stereo image.

StereoEnhancer

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

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



Width

Controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.

Delay

Increases the amount of differences between left and right channels to further increase the stereo effect.

Color

Generates additional differences between the channels to increase the stereo enhancement.

Mono

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	–

In Cubase, the SurroundPanner V5 is only available as a channel panner, but not as an insert effect. For a description of SurroundPanner V5, see the Operation Manual.

Surround Plug-ins

Bass Manager

If your main speakers are small, this plug-in allows you to achieve full-range sound by using the LFE speaker of the studio as a subwoofer. If your LFE is small but the main speakers provide full-range sound, the plug-in allows you to strengthen the LFE sound by routing higher frequencies from the LFE to the main speakers.

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



If your monitoring system is connected directly to your audio interface and lacks a bass management system, you can use Bass Manager in the Control Room inserts to adjust the signal levels and signal routing of the LFE and main speakers. Thus, small speakers can achieve full-range sound.

Input/Output Level Meters

Level In

Shows the level of the input signal.

Level Out

Shows the level of the output signal.

Mains

Subwoofer

If this is activated, the low part of the main speakers is routed to the subwoofer.

Freq

Sets the crossover frequency. The frequencies below this value are routed to the subwoofer.

Gain

Sets the amount of attenuation/boost of the signal that is routed to the subwoofer.

LFE

Center

If this is activated, the LFE signal is routed to the center channel.

L/R

If this is activated, the LFE signal is routed to the left and right channels.

Freq

Sets the crossover frequency. Frequencies above this value are routed to the left and right channels or to the center channel.

Gain

Sets the amount of attenuation/boost of the signal that is routed to the left and right channels or to the center channel.

+10 dB

Boosts the level of the LFE signal by 10dB.

Subwoofer

Solo

Solos the subwoofer signal.

Mute

Mutes the subwoofer signal.

Low Cut

Additional low-cut filter for the subwoofer.

High Cut

Additional high-cut filter for the subwoofer.

+10 dB

Boosts the level of the subwoofer signal by 10dB.

NOTE

You can also use the graphical displays to set the parameters or enter the parameter values manually in the value fields.

MatrixDecoder

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.

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



NOTE

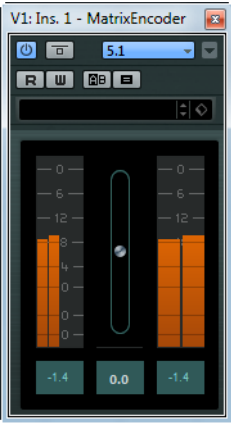
If an encoded mix is played back via the decoder, the Lt/Rt channels are again converted to four outputs (LRCS).

IMPORTANT

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

PROCEDURE

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, use MatrixEncoder with the 5.0 Surround Format.
2. Place the MatrixEncoder in the first post-fader insert slot (#7) for the output bus, followed by the MatrixDecoder (#8).

RELATED LINKS

[Using the MatrixEncoder with the 5.0 Surround Format on page 129](#)

Using the MatrixEncoder/Decoder

PROCEDURE

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 it is 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.

IMPORTANT

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. If 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.

RESULT

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 where 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.

PROCEDURE

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.
-

RESULT

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.

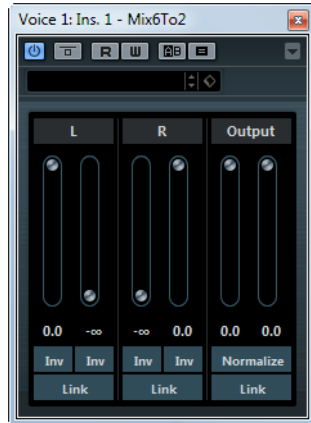
Mix6to2

This plug-in 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.

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

NOTE

This plug-in does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in is only available in the Master Section and if a surround audio montage is active.



Surround Channels

Volume

Govern how much of the signal is included in the left and/or right channel of the output bus.

Link

Links the two volume faders.

Invert

Allow you to invert the phase of the left and right channel of the surround bus.

Output Bus

Volume

Set the volume of the of the mixed output.

Link

Links the two **Output** faders.

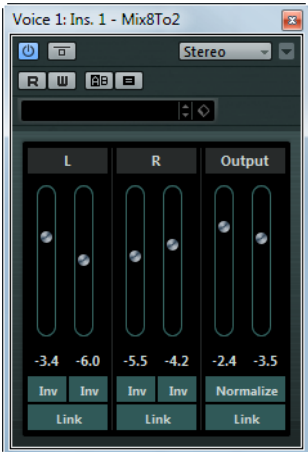
Normalize

If this option is activated, the mixed output is normalized. For example, the output level is automatically adjusted so that the loudest signal is as loud as possible without clipping.

Mix8to2

This plug-in 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.

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



This plug-in does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in is only available in the Master Section and if a 8 channel audio montage is active.

Surround Channels

Volume

Govern how much of the signal is included in the left and/or right channel of the output bus.

Link

Links the two volume faders.

Invert

Allow you to invert the phase of the left and right channel of the surround bus.

Output Bus

Volume

Set the volume of the of the mixed output.

Link

Links the two **Output** faders.

Normalize

If this option is activated, the mixed output is normalized. For example, the output level is automatically adjusted so that the loudest signal is as loud as possible without clipping.

MixConvert V6

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.

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



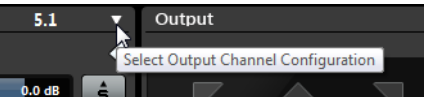
MixConvert V6 can be used as an insert effect like other plug-ins but it also has special functions. The MixConvert V6 plug-in is also used to convert an audio-related channel into a different format if the corresponding input/output configuration is not handled by the SurroundPanner V5. The sequencer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

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.



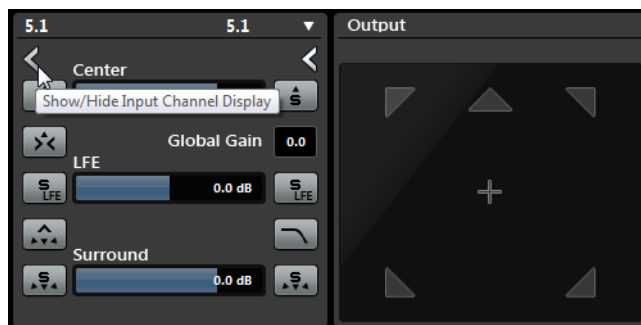
NOTE

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.



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

Solo Channel buttons

Solo 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 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, [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.

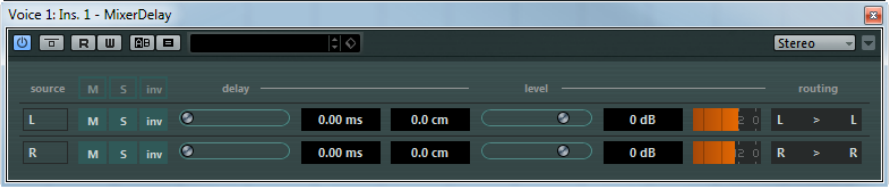
Depending on whether you solo a channel in the input or output configuration, the following applies:

- 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

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

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



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

Mute

Allows you to mute individual channels.

Solo

Allows you to solo individual channels.

Inv

Lets you invert the phase or polarity for individual channels.

Delay

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

Allows you to fine-tune the volume balance between the surround channels.

Volume

Shows the level of the input signal.

Routing

Lets you select/switch the outputs for the channels quickly. You can assign the same output to several channels by holding down [Alt]/[Option] while selecting. Note that there are also several channel routing presets available.

NOTE

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. 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.

IMPORTANT

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

MultiScope

MultiScope can be used for viewing the waveform, phase linearity, or frequency content of a signal.

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

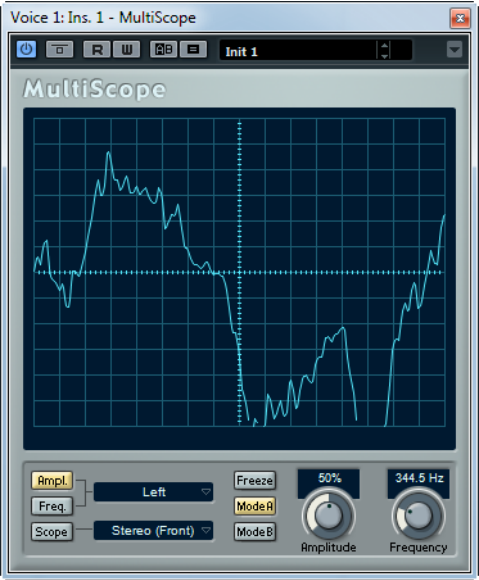
There are three different modes:

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

NOTE

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

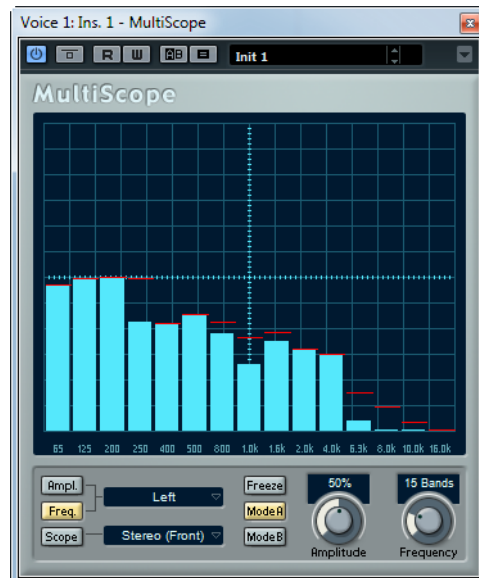
Oscilloscope Mode (Ampl.)



- 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 select the **Left** or **Right** channel for viewing, or **Stereo** for both channels to be shown in the window.
- 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.

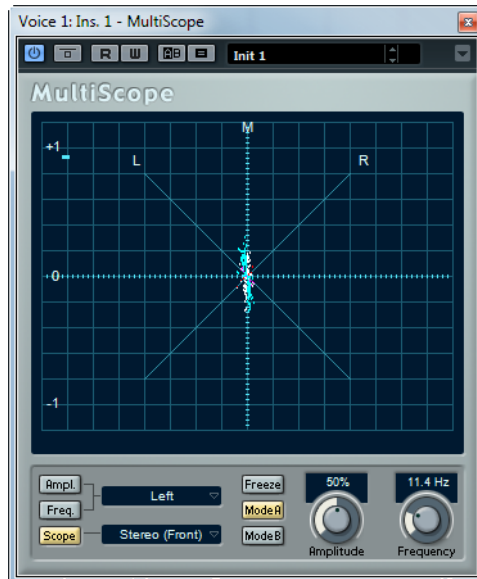
- You can now adjust the **Amplitude** knob to increase/decrease the vertical size of the waveform, and the **Frequency** knob to select the frequency area for viewing.

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 from left to right, starting with the lower frequencies.
- If the source signal is stereo, you can select the **Left** or **Right** channel for viewing, or **Stereo** for both channels to be shown in the window.
- 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.

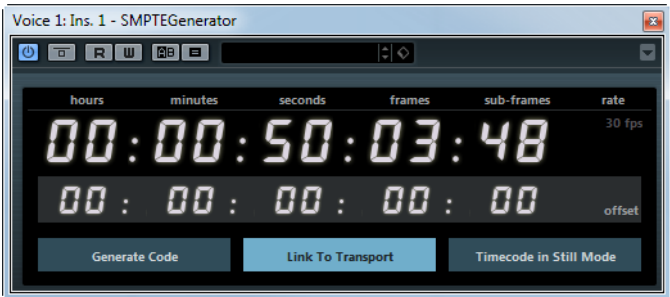
If MultiScope is used with a surround channel in Scope mode, the pop-up menu to the right of the Scope button determines the result:

- 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

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.

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



Main timecode display

This display shows the current timecode.

If **Link to Transport** is deactivated, the generator is in free run mode. You can use the timecode display to set the SMPTE start time.

If **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

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: You 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

If this button is activated, 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.

Link to Transport

If this button is activated, the timecode is synchronized to the Transport panel.

Timecode in Still Mode

If this button is activated, 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.

NOTE

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.

Synchronizing a Device to Your Host

PROCEDURE

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. Deactivate any EQ settings for this track.
2. Connect the corresponding output on the audio hardware to the timecode input on the device that you want to synchronize to your host application.
Make all necessary settings for the external device so that it synchronizes to incoming timecode.
3. Optional: 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**.

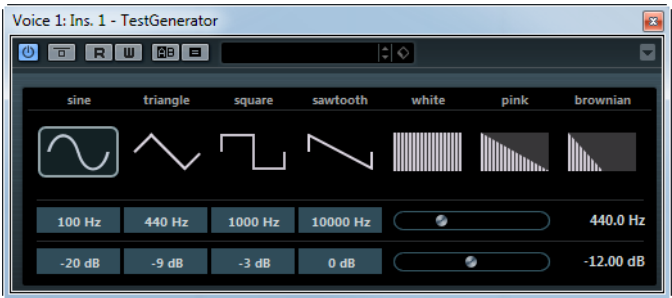
RESULT

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

TestGenerator

This utility plug-in allows you to generate an audio signal, which can be recorded as an audio file.

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



The resulting file can then be used for a number of purposes:

- Testing the specifications of audio equipment
- Measurements of various kinds, such as calibrating tape recorders
- Testing signal processing methods
- 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.

Waveforms and noise section

Allows you to set the basis for the signal generated by the waveform generator. You can choose 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.

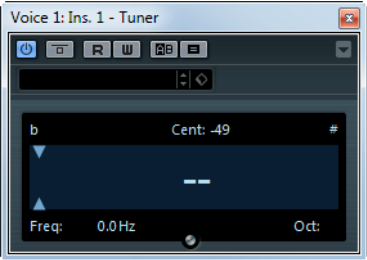
Gain section

Allows you to set the amplitude of the signal. The higher the value, the stronger the signal. You can select one of the preset values, or use the slider to set a value between -81 and 0dB.

Tuner

This is a guitar tuner.

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



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 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

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

A typical arpeggiator accepts a chord as input, and plays back each note in the chord separately, with the playback order and speed set by the user.

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



Play Order buttons

Allow you to select the playback order for the arpeggiated notes. 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.

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 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

You can use the same number in several slots, creating arpeggio patterns that are not possible using the standard play modes. You need to begin with the leftmost slot and then fill the slots to the right.

MIDI Thru

If this button is activated, the notes that you play pass through the plug-in and are sent out together with the arpeggiated notes.

Creating an Arpeggio

PROCEDURE

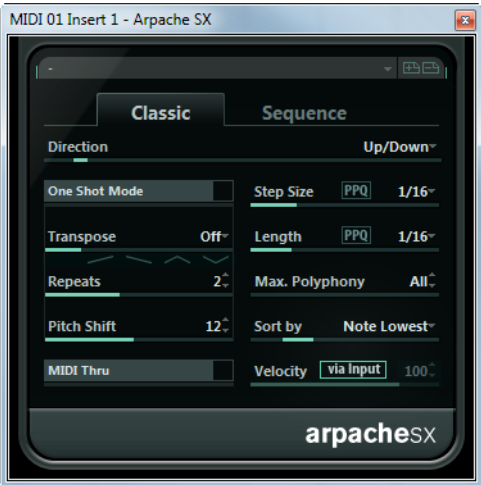
1. Select a MIDI track and activate monitoring (or record enable it) so that you can play through the track.
Make sure that the track is properly set up for playback to a suitable MIDI instrument.
2. Select and activate the arpeggiator.
For now, use it as an insert effect for the selected track.
3. In 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.

Arpache SX

This is a versatile and advanced arpeggiator, capable of creating anything from traditional arpeggios to complex, sequencer-like patterns.

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



Classic vs. Sequence Mode

Arpache SX has two different modes: **Classic** and **Sequence**. The **Classic** mode determines the basic behavior of Arpache SX. **Sequence** mode uses the events of an additional MIDI part as a pattern. This pattern forms the basis for the arpeggio, in conjunction with the MIDI input.

Classic Mode

Direction

Allows you to choose how the notes in the played chord should be arpeggiated.

One Shot Mode

Activate this option if you want the phrase to be played only once. If this option is deactivated, the phrase is looped.

Transpose

With a setting other than **Off**, 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 played notes 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.

Length

Determines the length of the arpeggio notes (in fixed note values or PPQ, if the **PPQ** button is activated).

Max. Polyphony

Determines how many notes should be accepted in the input chord. The **All** setting means there are no limitations.

Sort by

If you play a chord, 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 notes in the chord you play. In Sequence mode you can also activate the **from sequence** option.

Sequence Mode

In Sequence mode, you can import a MIDI part into Arpache SX by dragging it from the Project window onto the **Drop MIDI Sequence** field on the right of the Arpache SX panel.

The notes in the dropped MIDI part are sorted internally, either according to their pitch if the **MIDI Seq. sort by pitch** checkbox is 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 play) 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 – Arpache SX tries to play back the pattern from the dropped MIDI part but using the notes from the MIDI input. The result depends on the **Play Mode** setting.

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 if all keys are released, the phrase continues playing from the last position (where it stopped), if 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 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 random notes.

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.

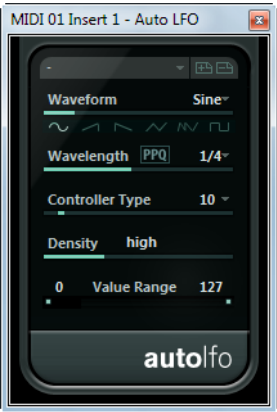
NOTE

You can choose to keep the original note timing, note length, and note velocities from the dropped MIDI part, by selecting **from sequence** for the **Step Size**, **Length** and **Velocity** options.

Auto LFO

This effect 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.

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



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 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.

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.

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

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.

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

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.

Control Panel



Patterns and Subbanks

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 a number (1 to 4) at the top of the display. To select a pattern within this subbank, click 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.

RELATED LINKS

[Triggering Patterns on page 157](#)

[The Swing Setting on page 154](#)

Selecting Drum Sounds

PROCEDURE

1. 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.
 2. To find the right sound, audition the selected drum sound by clicking the **Preview Instrument** button (the speaker icon).
-

Entering Drum Steps

PREREQUISITE

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.

PROCEDURE

- 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.

NOTE

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

Removing Steps

PROCEDURE

- To remove a drum step, simply click on the corresponding field again.

NOTE

To remove a range of drum steps, click and drag over them.

Velocity Settings

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.

NOTE

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.

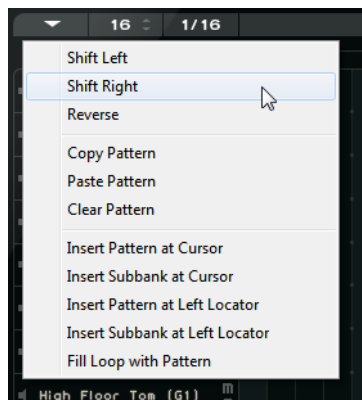
Lane Handling

- 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.

IMPORTANT

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

The Pattern Functions Menu



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 useful if 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.

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.

Insert Pattern at Left Locator

Creates a MIDI part for the current pattern and inserts it in the Project window, at the left locator.

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.

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.

NOTE

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.

RELATED LINKS

[Converting Patterns into MIDI Parts on page 156](#)

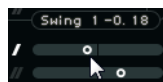
The Swing Setting

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.

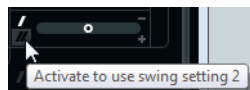
In the lower right section of the Beat Designer panel, you can find two **Swing** sliders. Dragging a slider to the right delays every second or third drum step in the pattern. Dragging to the left makes them play a little earlier.

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.

PROCEDURE

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 the mouse at it. If you click a step, the first square is filled to indicate that you added a flam.



2. Click again to add the second and third flam.
3. In the lower left section of the Beat Designer panel you can make settings for the flams you created.



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.

If 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.

PROCEDURE

1. 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.
2. Experiment with the settings to find out which fit best in your project.

NOTE

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.

Using the Drum Patterns in Your Project

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.

PROCEDURE

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.



- 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.

IMPORTANT

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.

IMPORTANT

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.

NOTE

Once a pattern is converted into a MIDI part, it cannot be opened in the Beat Designer again.

RELATED LINKS

[The Pattern Functions Menu on page 153](#)

Triggering Patterns

If you want to modify your drum patterns in Beat Designer while working on the project, you can trigger the patterns from within the project.

You can trigger the patterns in 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).

PROCEDURE

1. Open Beat Designer for a track.
2. Click the **Jump** field to activate Jump mode.



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

- 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: Activate the **Now** button to switch patterns immediately. If **Now** is deactivated, patterns switch at the beginning of the next bar in the project.
 - If 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.
3. 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.
 4. 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 with **Jump** mode activated to automatically create MIDI parts containing the trigger events.

NOTE

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

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.

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

There are three main operating modes: **All Keys**, **One Octave**, and **Global Key**. You can switch between these modes using the **Chords** pop-up menu.

For every key, you can record up to 8 different chords or variations on so-called layers.



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. If you play any of these keys, you hear the assigned chords instead.

One Octave

This 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. If you play a note in a different octave, you hear a transposed version of the chords set up for this key.

Global Key

In this 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.



NOTE

In **Global Key** mode, the C3 key has a special marking, because this is the only key used in this mode.

RELATED LINKS

[Using Layers on page 161](#)

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. If you choose the trigger note for a chord, for example, the keyboard display is shown in red.

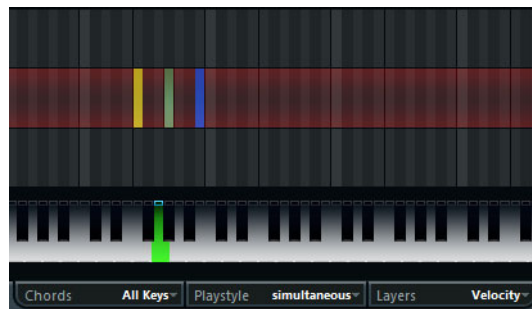
PROCEDURE

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.



The keyboard display in **Learn** mode



The second layer in **Learn** mode

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.

NOTE

In **Global Key** mode, you do not have to choose a trigger key. The first layer is activated automatically.

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, Chorder learns the chord as soon as you release all keys of your MIDI keyboard.

As long as a key is pressed, you can continue looking for the right chord.

- If more than one layer is shown, Chorder jumps automatically to the next layer where you can record another chord.

If all 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).

- If you are entering chords with the mouse, Chorder does not jump to the next layer automatically.

You can select/deselect as many notes as you want and then click on another layer or deactivate the Learn mode to continue.

4. Repeat the above with any other keys you want 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 8 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.

PROCEDURE

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 you want to use.
 3. Enter the chords.
-

RESULT Now you can play the keyboard and trigger the variations according to the selected layer mode.

RELATED LINKS

[Empty Layers on page 162](#)

Layer Modes

You can play the keyboard and trigger the variations according to the selected layer mode.

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 2 variations, 2 velocity zones are used: 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.

With the **Velocity spread** slider at the bottom right of the window, you can change the velocity ranges of the layers.

Interval

In this mode Chorder plays one chord at a time. If the **Interval** mode is selected, you trigger a layer by pressing 2 keys on your keyboard. The lower key determines the base note for the chord. The layer number is determined by the difference between the 2 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 want to use only 1 layer.

Empty Layers

If you enter fewer chords than layers are available for a key, these layers are filled automatically when you deactivate the **Learn** mode.

This works according to the following rules:

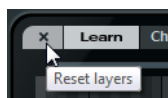
- Empty layers are filled from bottom to top.
- If there are empty layers below the first layer with a chord, these are filled from top to bottom.

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

PROCEDURE

- In **Learn** mode, click **Reset layers** at the top left of the Chorder window.



RESULT

For the selected trigger key, all notes in the different layers are deleted.

Playstyle

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

simultaneous

In this mode, all notes are played back simultaneously.

fast up

In this mode, a small arpeggio is added, starting with the lowest note.

slow up

Similar to **fast up**, but using a slower arpeggio.

fast down

Similar to **fast up**, but starting with the highest note.

slow down

Similar to **slow up**, but starting with the highest note.

fast random

In this mode, the notes are played back in a rapidly changing random order.

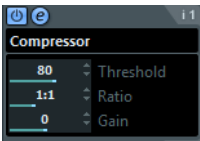
slow random

Similar to **fast random**, but the note changes occur more slowly.

Compressor

This MIDI compressor is used for evening out or expanding differences in velocity. Though the result is similar to what you get with the Velocity Compression track parameter, Compressor presents the controls in a manner more like regular audio compressors.

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



Threshold

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

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.

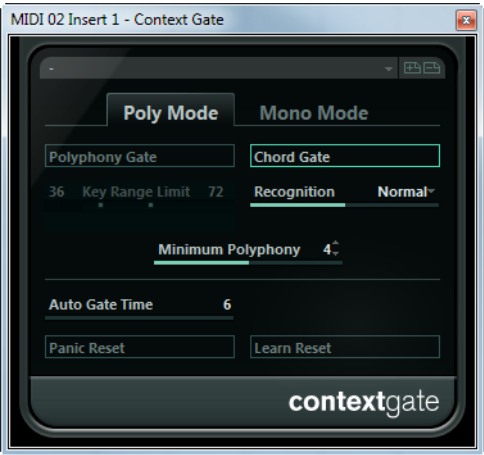
Gain

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

Context Gate allows for selective triggering/filtering of MIDI data.

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



This effect 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.

Poly Mode

Polyphony Gate

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.

Chord Gate

If **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.
- **Normal** mode takes more tensions into account.

Mono Mode

Channel Gate

If 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.

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

Sends an “All Notes Off” message over all channels, in case of hanging notes.

Learn Reset

If 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.

RELATED LINKS

[Application Examples on page 165](#)

Application Examples

Poly Mode

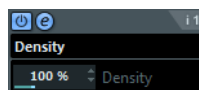
In this mode, you can use Context Gate to accompany yourself during a live guitar performance using a VST instrument. To do this, you might use a guitar-to-MIDI converter: You could then program Context Gate, for example, to allow only those notes to pass the gate that are part of a four-note chord. 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 this 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

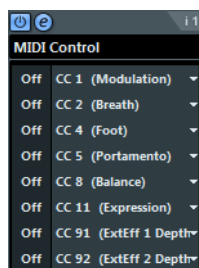
	Cubase LE	Cubase AI	Cubase Elements	Cubase Artist	Cubase	Nuendo	NEK
Included with	–	–	–	X	X	X	–



This generic control panel affects the density of the notes being played from or through the track. If this is set to 100%, the notes are not affected. Density settings below 100% randomly filter out or mute notes. Settings above 100% randomly add notes that were played before.

MIDI Control

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



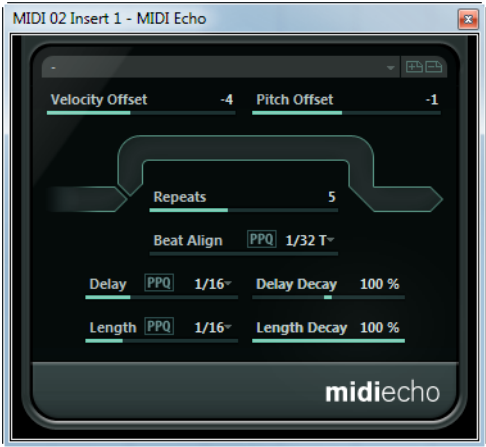
This generic control panel allows you to select up to 8 different MIDI controller types and use the value fields or sliders (which are displayed when you click on a value field while holding down [Alt]/[Option]) to set values for these. A typical use for this would be if you are using a MIDI instrument 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

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.

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



The effect does not echo the actual audio, but the MIDI notes which eventually produce the sound in the synthesizer.

Velocity Offset

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).

Pitch Offset

If you set this to a value other than 0, the 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

The number of echoes (1 to 12) of each incoming note.

Beat Align

During playback, this parameter quantizes the position of the first echo note. You can either set this to rhythmically exact values 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.

NOTE

The echo time can also be affected by the **Delay Decay** parameter.

NOTE

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 or activate the **PPQ** button and choose a **PPQ** value. This makes it easy to find rhythmically relevant delay values, but still allows for experimental settings in between.

Delay Decay

Adjusts how the echo time changes with each successive repeat. The value is set as a percentage.

- If this is set to 100% the echo time is the same for all repeats.
- 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

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.

NOTE

The length can also be affected by the **Length Decay** parameter.

Length Decay

Adjusts how the length of the echoed notes changes with each successive repeat. The higher the setting, the longer the echoed notes.

About Ticks and Note Values

The timing and position-related parameters (**Delay**, **Length**, and **Beat Align**) can all be set in ticks. There are 480 ticks to each quarter note. The parameters allow you to step between the rhythmically relevant values. The following table shows 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

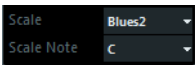
MIDI Modifiers

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.

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

The MIDI Modifiers effect also includes the **Scale Transpose** function that is not available among the track parameters.

Scale Transpose



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 deactivate **Scale Transpose**, select **No Scale** from the **Scale** pop-up menu.

MIDI Monitor

This effect monitors incoming MIDI events.

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



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 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. If you choose **Controller**, 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

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.

NOTE

The larger the buffer, the more processing resources are required.

Export

Allows you to export the monitoring data as a simple text file.

Record Events

This button to the left of the **Inputs** section allows you to start or stop the monitoring of MIDI events.

Clear List

This button to the left of the **Show** section allows you to clear the table of recorded MIDI events.

Micro Tuner

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



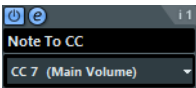
Micro Tuner lets you set up a different microtuning scheme for the instrument, by detuning each key.

- 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).
- You can adjust all keys by the same amount by keeping [Alt]/[Option] pressed.

Micro Tuner comes with a number of presets, including both classical and experimental microtuning scales.

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.

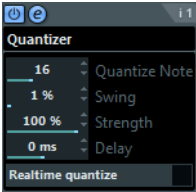
IMPORTANT

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.

Quantizer

This effect allows you to apply quantizing in realtime. This makes it easier to try out different settings when creating grooves and rhythms.

	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.

NOTE

The main **Quantize** function in your Steinberg DAW is described in the Operation Manual.

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. If 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. This delay can be automated.

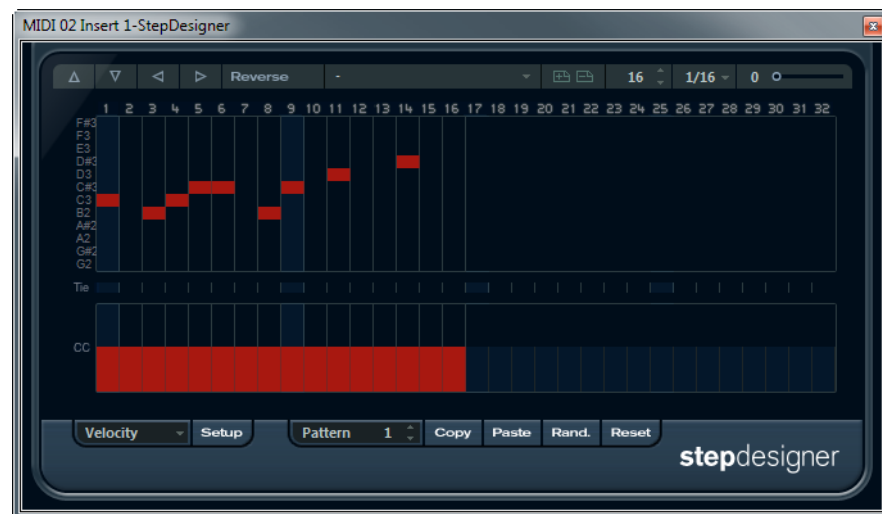
Realtime quantize

During live mode, this option can be used to change the timing of the notes that are played so that they fit the quantize grid.

StepDesigner

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

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



Creating a Basic Pattern

PROCEDURE

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.
This setting determines the step length.
3. Specify the number of steps in the pattern with the **Number of steps** setting.
The maximum number of steps is 32. For example, setting **Step size** to 16 and **Number of steps** to 32 creates a 2bar pattern with sixteenth note steps.

4. Click in the note display to insert notes.

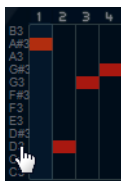
You can insert notes on any of the 32 steps, but StepDesigner only plays back the number of steps set with the **Step size** parameter.

- 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

Each step can contain one note only – StepDesigner is monophonic.



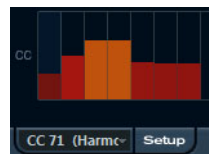
Adding Controller Curves

The Controller pop-up menu has two more items: two controller types.

PROCEDURE

1. 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.
2. 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.



NOTE

If you drag a controller event bar all the way down, no controller value is sent out on that step.

Other Pattern Functions

These functions make it easier to edit, manipulate and manage patterns.

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

Allows you to load/save presets for the effect.

NOTE

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.

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 **Write** automation and switching patterns during playback or by drawing on the automation track for 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.

PROCEDURE

1. Select a MIDI track or create a new one and activate StepDesigner as an insert effect.
 2. Set up several patterns.
 3. Activate 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.
-

RESULT

You now hear the recorded pattern changes.

NOTE

You can only automate the 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 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.

The Available Control Panels

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.

GS 1

Contains effect sends and various sound control parameters for use with instruments compatible with the Roland GS standard.

XG 1

Contains effect sends and various sound control parameters for use with instruments compatible with the Yamaha XG standard.

XG 2

Global settings 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 resets all parameters to their default values, and sends out the corresponding MIDI messages.

GS 1

The following controls are available if the **GS 1 Controls** mode is selected:

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.

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.

XG 1

The following controls are available if the XG 1 mode is selected.

Send 1

Send level for the reverb effect.

Send 2

Send level for the chorus effect.

Send 3

Send level for the variation effect.

Attack

Adjusts the attack time of the sound. Lowering this value shortens the attack, while raising it gives a slower attack. Middle position means no adjustment is made.

Release

Adjusts the release time of the sound. Lowering this value shortens the release, while raising it makes the release time longer. Middle position means no adjustment is made.

Harm.Cont

Adjusts the harmonic content of the sound.

Bright

Adjusts the brightness of the sound.

CutOff

Adjusts the filter cutoff frequency.

Resonance

Adjusts the filter resonance.

XG 2

In this mode, the parameters affect global settings in the instruments. 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, it might be a good idea to create an empty track and use this only for these global settings.

Eff. 1

Allows you to select which type of reverb effect should be used: No effect (reverb deactivated), 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 (chorus deactivated), 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. Select **No Effect** to deactivate the variation effect.

Reset

Sends an XG reset message.

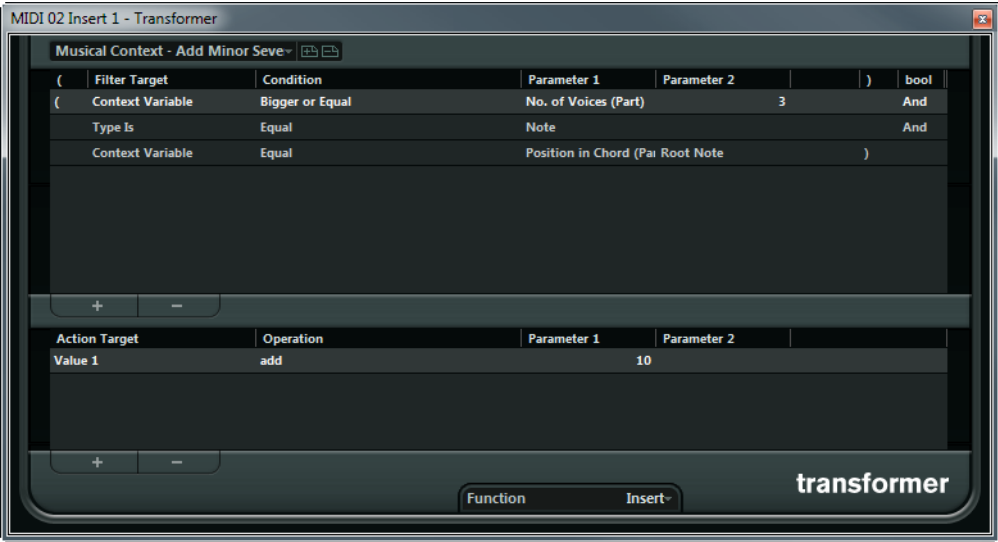
MastVol

Controls the Master Volume of an instrument. Normally, you should leave this at its highest position and set the volumes individually for each channel (with the volume faders in the MixConsole or in the Inspector).

Transformer

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.

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



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.

Included VST Instruments

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

NOTE

Most of the included instruments are compatible with VST3. For further information, see the Operation Manual.

Groove Agent ONE

Groove Agent ONE is a sample-based MPC-style virtual drum machine for creating beats and reconstructing loops.

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



Groups and Pads

Groove Agent ONE provides up to 128 pads, organized in eight groups of 16 pads.

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

- 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.

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.
- 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 samples to the trash icon in the LCD display to the left. The trash icon is found only on the **Voice**, **Filter**, and **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. If 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 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.
- To mute or solo a pad, click the corresponding icon in the upper left corner of a pad. 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. If the pad display 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 panel.

Resetting Pads

- The **Reset** button in the Global section in the top right corner of the panel allows you to clear all pad assignments of the current instance of Groove Agent ONE.
- The **Reset** button is locked by default. Clicking the locked **Reset** button has no effect.
- To unlock the **Reset** button, hold down [Shift] while clicking. The button color changes to red. When you click **Reset** now, all pad assignments are reset.

NOTE

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

Drag and Drop of Audio Material

You can drag one or more samples from your host application onto Groove Agent ONE. Samples can 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

If you select between up to 8 samples and drag them to Groove Agent ONE, dropping them onto a pad (or onto the Layer indicator), the corresponding number of layers for this pad is automatically created.

Drag & Drop to Several Pads

Rather than dropping several samples to the same pad, you can 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 dragged 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 available pads.

The samples are mapped to the available pads, starting with the pad on which you dragged the samples, and then upwards according to the MIDI pitches of the pads.

Replacing Individual Samples

PROCEDURE

1. Drag the new sample to the following location:
 - To the pad if you want to replace a sample mapped to one pad with another sample.
 - To the Layer indicator if you want to replace a sample in a pad layer with another sample.
 2. Press [Alt]/[Option] and drop the sample.
-

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.

PROCEDURE

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 [Shift]. 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.

In 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.
 4. Drag this MIDI file from the MIDI Export pad onto the Project window.

Dropping the file in the Project window creates a new MIDI track. You can also drop the MIDI file on an existing MIDI or instrument track.
 5. Play back the MIDI file.
-

RESULT

The unedited MIDI file plays the same groove as the original audio loop. By editing the MIDI file you can change the original groove.

NOTE

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.

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 if 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.

PROCEDURE

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

PROCEDURE

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.
-

RESULT The Presets browser is closed and the preset is loaded into Groove Agent ONE.

AFTER COMPLETING THIS TASK:

When a sample belonging to a preset cannot be found, Groove Agent ONE prompts you to locate the missing files.

You can click

- **Ignore** to skip this message
- **Locate File** to navigate to a specific folder containing the missing files
- **Search Folder** to browse a specific folder and any subfolders that might contain the missing files

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*.

PROCEDURE

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.
 3. Specify a location and a name for the new archive.
 4. Click **Save**.
-

RESULT

The archive is created and the dialog is closed.

NOTE

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

PROCEDURE

1. In the Exchange section, click the **Import** button.
 2. Navigate to the GAK file.
 3. Click **Open**.
-

RESULT

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 the instance of Groove Agent ONE as a whole. If 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 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, use one of the quick controls below the display, or click on the parameter in the display and adjust it with the mouse.
- You can select multiple pads for 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.

Brightness slider

Use the little slider at the 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

If this 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. If 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. 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

The name of the sample file.

Velocity

Here, you can specify the velocity range for the current layer.

Coarse

Here, you can tune the sample by up to ± 12 semitones.

Fine

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. If you click on a locator and press [Ctrl]/[Command], this zooms in on the waveform and centers the display around the locator.

NOTE

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 row of parameter controls below the LCD display shows the play parameters:

Volume

The volume of the pad selected for editing.

Pan

The panorama setting of the pad 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 the voice parameters.

Mode

Reverses the selected sample so that you hear it backwards.

Coarse

Allows you to tune the pad by up to ± 12 semitones.

Fine

Allows you to fine-tune the pad by up to ± 100 cents.

Mute Gr.

Allows you to 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 selected pad is played 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 the parameters used to edit the Groove Agent ONE filter.

Type

Sets the filter type: low-pass (**LP**), high-pass (**HP**), or band-pass (**BP**). If you set this knob to **OFF**, the settings on this page have no effect.

Cutoff

Sets the filter cutoff frequency.

Q

Sets the filter resonance.

Mod

Determines the influence that velocity has on the cutoff frequency. If this is set to 0%, the setting has no effect. At any other value, the cutoff frequency changes depending on the velocity.

Amplifier Parameters

The row of parameter controls below the LCD display shows the amplifier parameters.

Volume

The volume of the pad selected for editing.

Pan

The panorama setting of the pad 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

Determines the influence that velocity has on the pad volume setting. If this is set to 100%, the pad sounds louder the higher the velocity. If this is set to 0%, velocity has no effect on the pad volume.

Attack Mod

Determines the influence that velocity has on the **Attack** setting. If this is set to 0%, velocity has no effect on the attack. If this is 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 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.

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

NOTE

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

RELATED LINKS

[Slicing a Loop and Triggering Individual Sounds via MIDI on page 185](#)
[Saving a GAK Archive on page 187](#)

Automation of Groove Agent ONE Parameters

You can automate the following Groove Agent ONE parameters in your host application.

- Volume
- Pan
- Mute
- Cutoff
- Resonance

These parameters are available for the pads C1 to B4.

Groove Agent 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 “Groove Agent SE”.

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, open the tutorial preset.

PROCEDURE

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.

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.

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.

RELATED LINKS

[LoopMash Parameters on page 196](#)

[Applying Slice Selection Modifiers and Slice Effects on page 202](#)

How Does LoopMash Work?

Whenever you import a loop into LoopMash, the plug-in analyzes the audio material. It generates 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.

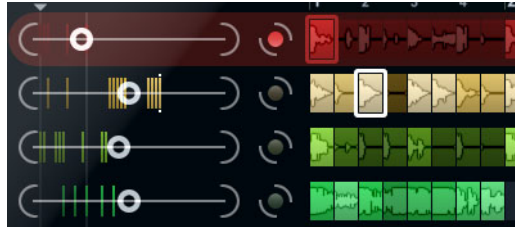
NOTE

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.

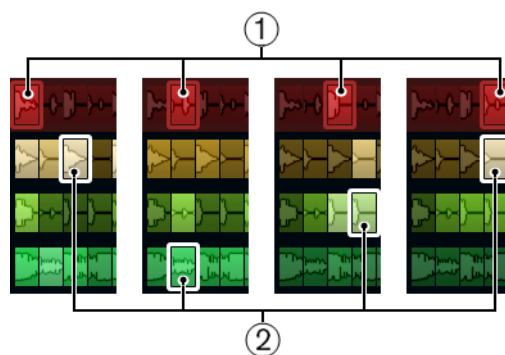
The similarity is shown by the brightness of each slice on each track, and also by the position of each slice on the similarity gain slider to the left of each track (if you click a slice, its position is highlighted on the similarity gain slider). The brighter a slice, the more similar a slice is to 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.



- 1) Master track slices for playback steps 1 to 4.
- 2) 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!

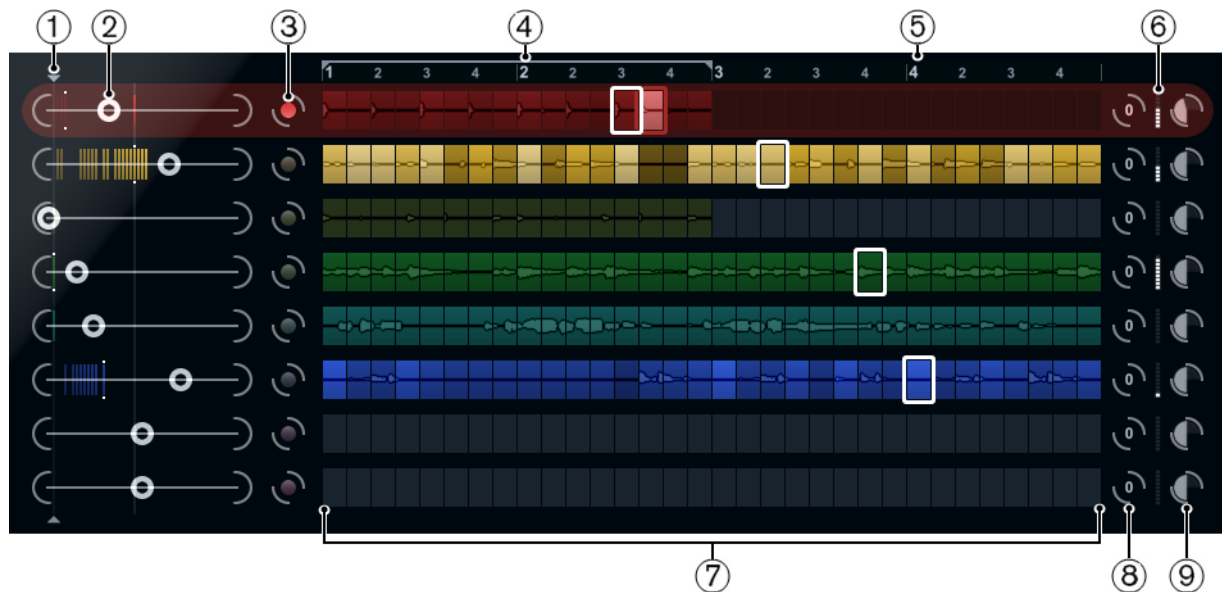
LoopMash Parameters

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

NOTE

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.

- 1) Similarity threshold control
- 2) Similarity gain sliders
- 3) Master track on/off
- 4) Loop range selector
- 5) Ruler showing bars and beats
- 6) VU meter
- 7) Track display
- 8) Track transposition value
- 9) Track volume

Importing and Removing Loops

You can import up to eight audio loops onto the eight tracks in the track display.

PROCEDURE

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.
-

RESULT

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. If a long loop contains more than 32 slices, LoopMash imports only the first 32. Ideally, you would use a loop file cut at bar boundaries. If you import your file from the MediaBay, LoopMash uses the tempo information supplied by the MediaBay for the slicing of the loop.

NOTE

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. The sounds of this loop are replaced by slices selected from the other loops in the current LoopMash configuration.

PROCEDURE

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

Auditioning Slices

PROCEDURE

1. Click on the slice that you want to hear.
 2. Use the **Step** function in the transport controls to step through the slices.
-

RELATED LINKS

[Transport Controls on page 203](#)

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 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.

PROCEDURE

1. 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 account.

NOTE

Short loop ranges (less than 1 bar) may conflict with the jump interval setting.

2. To change the playback range, click the loop range selector and drag it to a different position as a whole.
-

RELATED LINKS

[Storing Your Configuration as Scenes on page 205](#)

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.

PROCEDURE

1. 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.

NOTE

This function is tied to the setting for the **Slice Timestretch** parameter. If **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.

2. 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.

RELATED LINKS

[Audio Parameters on page 207](#)

Setting the Similarity

With the similarity gain slider (to the left of each track), you 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.

PROCEDURE

1. 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.

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.

2. 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 this threshold are not played.

NOTE

On the **Slice Selection** page at the bottom of the LoopMash panel, you can make further settings for influencing which slices are played.

RELATED LINKS

[Slice Selection on page 206](#)

Creating Composite Tracks

LoopMash allows you to build composite tracks.

PROCEDURE

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.



3. Click **OK**.
-

RESULT

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

NOTE

Composite tracks are quantized according to the set tempo.

RELATED LINKS

[Transport Controls on page 203](#)

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.

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** parameter that you set on the Slice Selection page).

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.

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!

NOTE

You can also apply effects to your overall performance.

RELATED LINKS

[Slice Selection on page 206](#)

[Performance Controls on page 208](#)

Transport Controls



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

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.

- Activate the **sync** button (to the right of the **Play** button) to synchronize LoopMash to the project tempo set in your host application.

If **sync** is activated, you can start playback using the transport controls of your host application. With **sync** deactivated, LoopMash starts playing when you click the **Play** button in LoopMash.
- If **sync** 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].
- If **sync** 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** functions with your MIDI keyboard.

C2

Start

D2

Stop

E2

Sync on

F2

Sync off

NOTE

If you do not have a MIDI keyboard connected to your computer, you can use the virtual keyboard (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. For 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.



- 1) Save scene
 - 2) Remove scene
 - 3) Jump interval
 - 4) Selected scene
 - 5) Pad with associated scene
 - 6) Empty scene pad
- 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 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.

IMPORTANT

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

Setting a Jump Interval

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

PROCEDURE

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

RESULT

NOTE

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

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.

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.

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 fewer 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.

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.

Selection Grid

Determines 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 can 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 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.

RELATED LINKS

[Storing Your Configuration as Scenes on page 205](#)

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.

Adapt Mode

With this 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. If the slider is all the way to the left, the slices follow the rhythmic pattern defined by the original master loop.

Slice Timestretch

Allows you 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.

Staccato Amount

If you move this slider to the right, the length of the slices is gradually reduced, giving the output a staccato feel.

Dry/Wet Mix

Sets the balance between the volumes of the master loop and the selected slices from the other tracks.

RELATED LINKS

[Setting Track Transposition Value and Track Volume on page 200](#)

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 the button is 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.

NOTE

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:

Cycle 4, 2, 1

Sets up a short cycle over 4, 2, and 1 slices, respectively. This short cycle is always set up within the loop range that is set in the ruler. 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 selected slices continuously until you release the button.

NOTE

You cannot save global effects in scenes. To apply effects and save them in scenes, use slice effects.

Triggering the Performance Controls with Your MIDI Keyboard

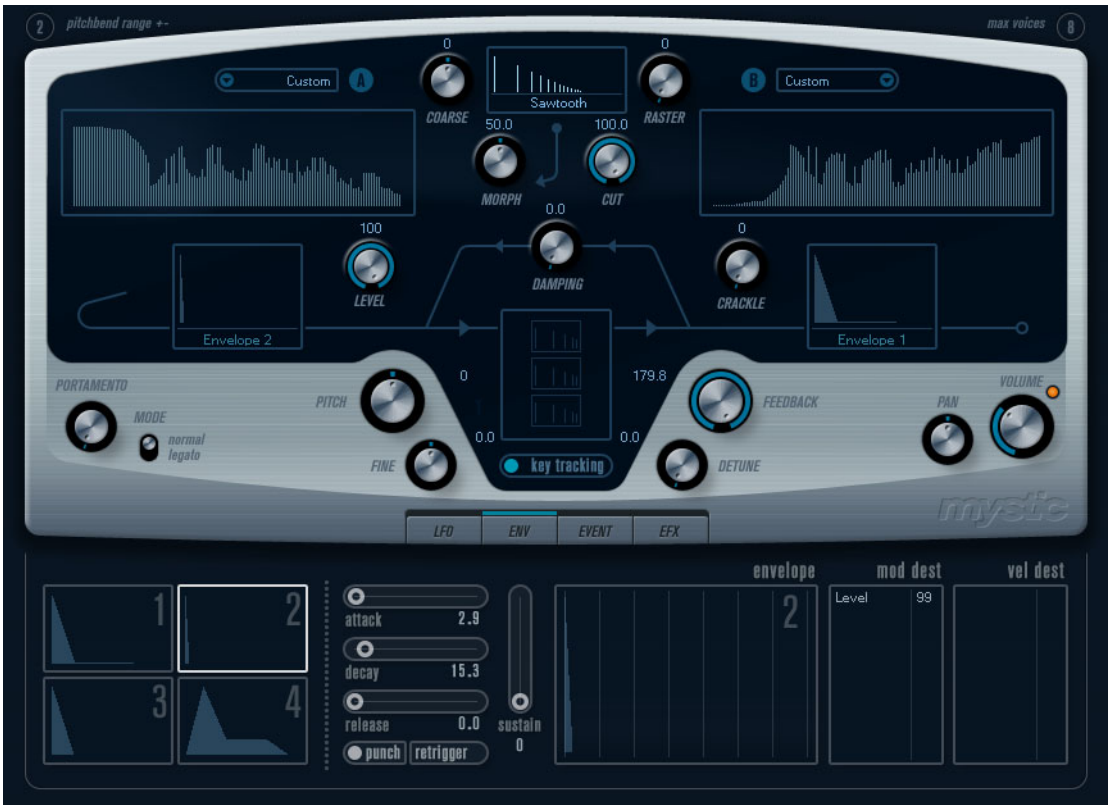
You can trigger the Performance Controls with your MIDI keyboard starting from C3 upwards.

RELATED LINKS

[Applying Slice Selection Modifiers and Slice Effects on page 202](#)

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. 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 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.

- 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 – if this is 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.

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.

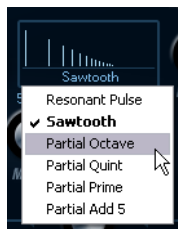
- To set up the contour, click in one of the displays and drag the mouse to draw a curve. This produces the inverse contour in the other display, for maximum sonic versatility.

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 want to calculate a random 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

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.

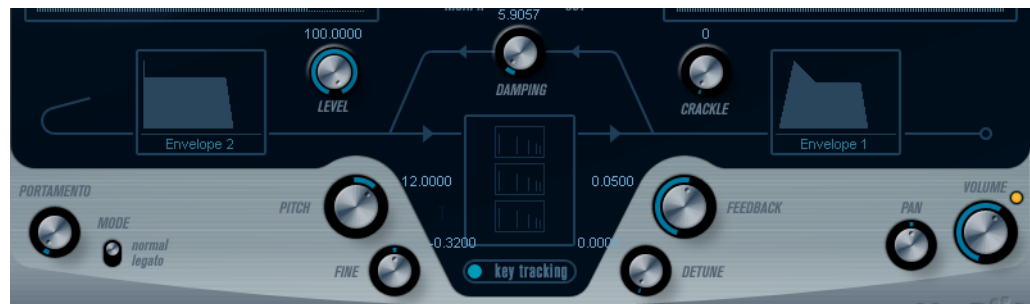
Coarse

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

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

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

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

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

Determines whether the impulse sound should track the keyboard. This affects the sound of the comb filters in a way similar to a key track switch on a regular subtractive synth filter.

Portamento

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 (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** knob 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 of the instrument in the stereo spectrum. 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 that can be assigned to control parameters.
- The **Event** page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments.
- The **Effect** page offers three separate effect types: Distortion, Delay, and Modulation.

RELATED LINKS

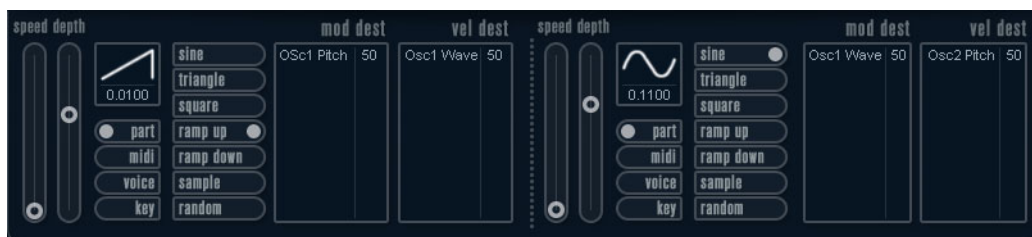
[Envelope Page on page 218](#)

[Event Page on page 221](#)

[Effects \(EFX\) Page on page 222](#)

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 selected preset, there may already be modulation destinations assigned, in which case these are listed in the **Mod Dest** box for each LFO.

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.

Speed

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

Controls the amount of modulation applied by the LFO. If this is set to zero, no modulation is applied.

Waveform

Sets the LFO waveform.

Sync mode (Part/MIDI/Voice/Key)

Sets the sync mode for the LFO.

RELATED LINKS

[Assigning LFO Modulation Destinations on page 217](#)

About the Sync Modes

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

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 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.

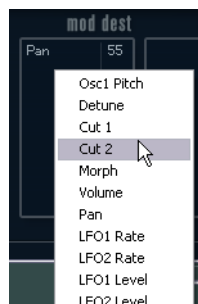
Assigning LFO Modulation Destinations

You can assign a modulation destination for an LFO.

PROCEDURE

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 [Enter].

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.

PROCEDURE

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.

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

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 velocity destination click on its name in the list and select **Off** from the pop-up menu.

LFO modulation velocity control

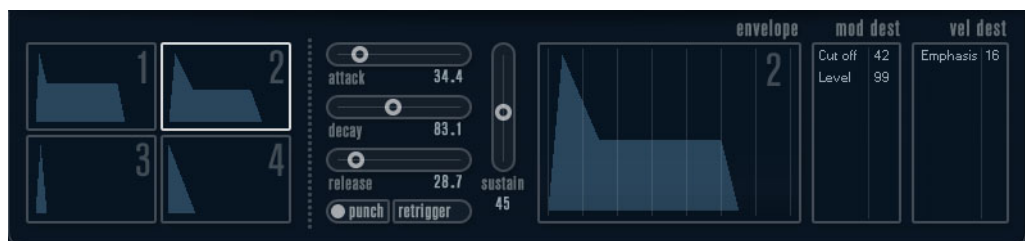
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- 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 2 ways: by using the sliders or by clicking and 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 is assigned, there is no output.
- Envelope 2 is by default assigned to the **Level** parameter.

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** parameter. The **Decay** has no effect if the **Sustain** parameter is set to maximum.

Sustain

Determines the level the envelope rests at after the **Decay** phase. Note that **Sustain** represents a level, whereas the other envelope parameters represent times.

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

If **Punch** is activated, the start of the decay phase is delayed a few milliseconds, that is, 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

If **Retrigger** is activated, the envelope retriggers 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.

Assigning Envelope Modulation Destinations

You can assign a modulation destination for an envelope.

PROCEDURE

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 [Enter].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.

PROCEDURE

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

- 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 velocity destination click on its name in the list and select **Off** from the pop-up menu.

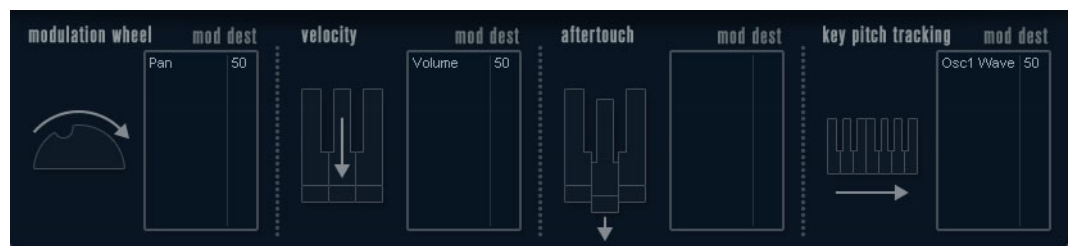
Envelope modulation velocity control

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.



Modulation Wheel

The modulation wheel on your keyboard can be used to modulate parameters.

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.

Key Pitch Tracking

This can change parameter values linearly according to where on the keyboard you play.

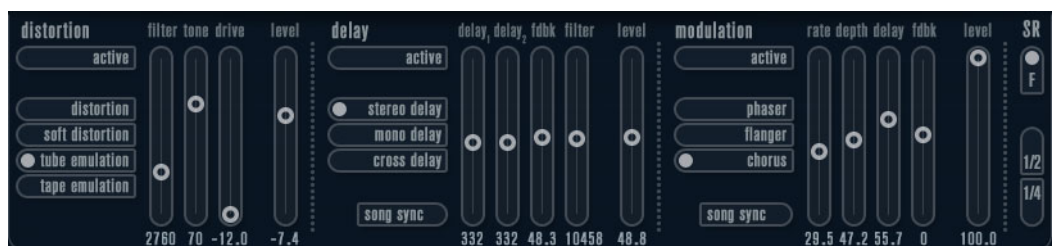
Assign a Controller to a Parameter

PROCEDURE

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 [Enter].
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 choose 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.

Drive

Sets the amount of distortion by amplifying the input signal.

Filter

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

Controls the relative amount of low-pass and high-pass filtered signal.

Level

Controls the output level of the effect.

Delay

You can choose 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.

Song Sync

Activates/Deactivates tempo sync of the delay times.

Delay 1

Sets the delay time ranging from 0 ms to 728 ms. If **MIDI sync** is activated, the range is from 1/32 to 1/1; straight, triplet or dotted.

Delay 2

Same as **Delay 1**.

Feedback

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

Controls the output level of the effect.

Modulation

You can choose between 3 basic modulation characteristics:

- **Phaser** uses an 8-pole all-pass filter to produce the classic phasing effect.

- **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.

Song Sync

Activates/Deactivates tempo sync of the **Rate** parameter.

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

Controls the depth of the delay time modulation.

Delay

Sets the delay time of the four delay lines.

Feedback

Controls the amount of positive or negative feedback for all four delay lines.

Level

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 useful to emulate the lo-fi sounds of older digital synths.

- If the **F** button is active, the selected Part's program plays back with the sample rate set in the host application.
- If the **1/2** button is active, the selected Part's program plays back with half the original sample rate.
- If the **1/4** button 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 for 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.
- Three oscillators, each with 4 standard waveforms plus an assortment of specialized waveforms.
- Frequency modulation.
- Ring Modulation.
- Built-in effects.
- Prologue receives MIDI on all MIDI channels.
You do not have to select a MIDI channel to direct MIDI to the Prologue.

RELATED LINKS

[About the Filter Types on page 233](#)
[Selecting Waveforms on page 227](#)
[About Frequency Modulation on page 231](#)
[Ring Modulation on page 231](#)
[Effects \(EFX\) Page on page 222](#)

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.



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 oscillators, the corresponding Osc controls in the oscillator sections must be set to a suitable value.

OSC 1 Parameters

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

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)

If phase synchronization is activated, all oscillators restart their waveform cycles with every note that is 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. For bass sounds or drum sounds, it is often required that the attack of every note sounds the same, therefore, for these purposes activate phase sync. Phase sync also affects the noise generator.

Tracking button (On/Off)

If **Tracking** is activated, the oscillator pitch tracks the notes played on the keyboard. If **Tracking** is deactivated, the oscillator pitch remains constant, regardless of the note that is played.

Wave Mod button (On/Off)

Activates/Deactivates wave modulation.

Waveform pop-up menu

Sets the basic waveform for the oscillator.

OSC 2 Parameters

Osc 2 (0–100)

Controls the output level of the oscillator.

Coarse (± 48 semitones)

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 activated, this determines the frequency ratio of the oscillator regarding Osc 1.

Wave Mod (± 50)

This parameter is only active if the **Wave Mod** button next to the waveform selector is activated. 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 an LFO, classic PWM (pulse width modulation) is produced. Wave modulation can be applied to any waveform.

Ratio (1–16)

This parameter is only active if **Freq Mod** is activate. It adjusts the amount of frequency modulation applied to oscillator 2. It is normally referred to as FM index.

Sync button (On/Off)

If **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 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)

If **Tracking** is activated, the oscillator pitch tracks the notes played on the keyboard. If **Tracking** is deactivated, the oscillator pitch remains constant, regardless of the note that is played.

Freq Mod button (On/Off)

Activates/Deactivates frequency modulation.

Wave Mod button (On/Off)

Activates/Deactivates wave modulation.

Waveform pop-up menu

Sets the basic waveform for the oscillator.

OSC 3 Parameters

Osc 3 (0–100)

Controls the output level of the oscillator.

Coarse (± 48 semitones)

Determines the coarse pitch for Osc 3. If **FM** is activated, this determines the frequency ratio of the oscillator regarding Osc 1/2.

Fine (± 50 cent)

Fine-tunes the oscillator pitch in cent increments. If **FM** is activated, this determines the frequency ratio of the oscillator regarding Osc 1/2.

Ratio (1–16)

This parameter is only active if the **Freq Mod** button is activated. It adjusts the amount of frequency modulation applied to oscillator 3. It is normally referred to as FM index.

Sync button (On/Off)

If **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 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)

If **Tracking** is activated, the oscillator pitch tracks the notes played on the keyboard. If **Tracking** is deactivated, the oscillator pitch remains constant, regardless of the note that is played.

Freq Mod button (On/Off)

Activates/Deactivates frequency modulation.

Wave Mod button (On/Off)

Activates/Deactivates wave modulation.

Waveform pop-up menu

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 oscillators.

This means that you should turn off the Osc 1 output when using frequency modulation.
- The **Freq Mod** button activates/deactivates frequency modulation.
- 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 if you play a legato note. 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 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 fifth or octave, the ring modulated output sounds harmonic, other intervals produce inharmonic, 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.

RELATED LINKS

[Envelope Page on page 218](#)

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.

Filter type

Sets the filter type to low-pass, high-pass, band-pass, or notch.

Cutoff

Controls the filter frequency or cutoff. If a low-pass filter is used, it can control the opening and closing of the filter, producing the classic sweeping synthesizer sound. How this parameter operates is governed by the filter type.

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 self-oscillate, generating a distinct pitch. For band-pass or notch filters, the Emphasis setting adjusts the width of the band. If you raise the value, the band where frequencies are let through (band-pass), or cut (notch) becomes narrower.

Drive

Adjusts the filter input level. Levels above 0dB gradually introduce a soft distortion of the input signal, and decrease the filter resonance.

Shift

Internally, each filter consists of two or more subfilters connected in series. This parameter shifts the cutoff frequency of the subfilters. The result depends on the 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 12 dB LP or 12 dB HP filter type is selected.

Tracking

If this parameter is set to values over the 12 o'clock position, the filter cutoff frequency 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):

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.

18 dB 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** knob 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 of the instrument in the stereo spectrum. 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 that can be assigned to control parameters.
- The **Event** page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments.
- The **Effect** page offers three separate effect types: Distortion, Delay, and Modulation.

RELATED LINKS

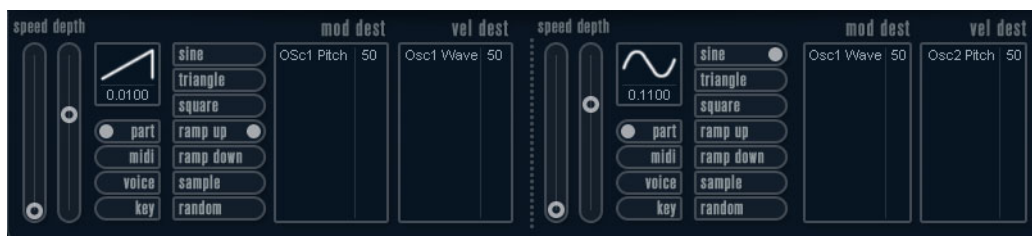
[Envelope Page on page 218](#)

[Event Page on page 221](#)

[Effects \(EFX\) Page on page 222](#)

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 selected preset, there may already be modulation destinations assigned, in which case these are listed in the **Mod Dest** box for each LFO.

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.

Speed

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

Controls the amount of modulation applied by the LFO. If this is set to zero, no modulation is applied.

Waveform

Sets the LFO waveform.

Sync mode (Part/MIDI/Voice/Key)

Sets the sync mode for the LFO.

RELATED LINKS

[Assigning LFO Modulation Destinations on page 217](#)

About the Sync Modes

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

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 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.

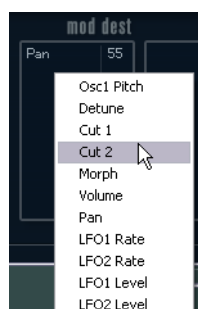
Assigning LFO Modulation Destinations

You can assign a modulation destination for an LFO.

PROCEDURE

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 [Enter].

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.

PROCEDURE

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.
 - You can set positive and negative values by clicking on the value in the list, typing in a new value and pressing [Enter].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 velocity destination click on its name in the list and select **Off** from the pop-up menu.
-

LFO modulation velocity control

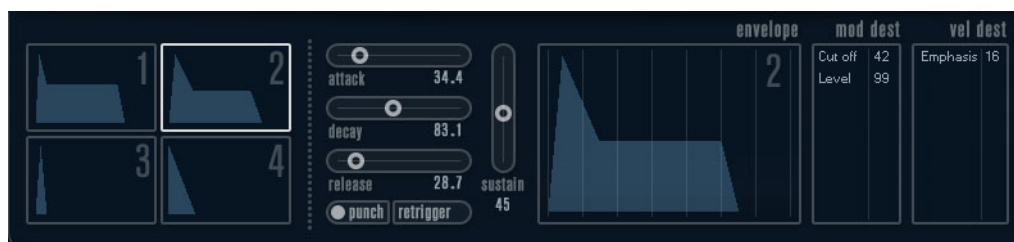
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- 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 2 ways: by using the sliders or by clicking and 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 is assigned, there is no output.
- Envelope 2 is by default assigned to the **Level** parameter.

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** parameter. The **Decay** has no effect if the **Sustain** parameter is set to maximum.

Sustain

Determines the level the envelope rests at after the **Decay** phase. Note that **Sustain** represents a level, whereas the other envelope parameters represent times.

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

If **Punch** is activated, the start of the decay phase is delayed a few milliseconds, that is, 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

If **Retrigger** is activated, the envelope retriggers 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.

Assigning Envelope Modulation Destinations

You can assign a modulation destination for an envelope.

PROCEDURE

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 [Enter].
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.

PROCEDURE

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 [Enter].

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 velocity destination click on its name in the list and select **Off** from the pop-up menu.

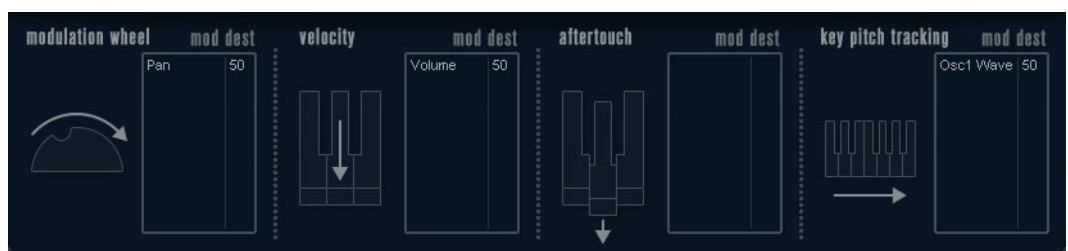
Envelope modulation velocity control

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.



Modulation Wheel

The modulation wheel on your keyboard can be used to modulate parameters.

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.

Key Pitch Tracking

This can change parameter values linearly according to where on the keyboard you play.

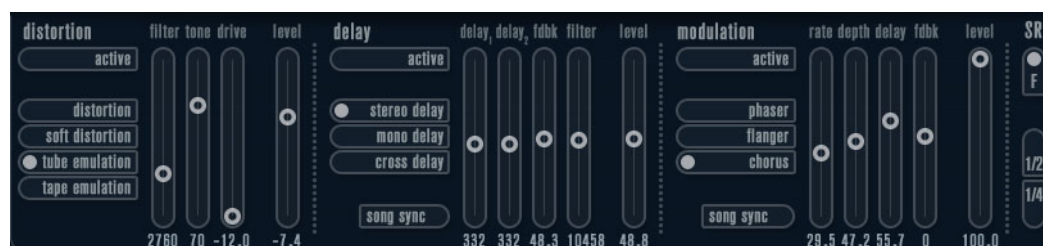
Assign a Controller to a Parameter

PROCEDURE

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 [Enter].
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 choose 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.

Drive

Sets the amount of distortion by amplifying the input signal.

Filter

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

Controls the relative amount of low-pass and high-pass filtered signal.

Level

Controls the output level of the effect.

Delay

You can choose 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.

Song Sync

Activates/Deactivates tempo sync of the delay times.

Delay 1

Sets the delay time ranging from 0 ms to 728 ms. If **MIDI sync** is activated, the range is from 1/32 to 1/1; straight, triplet or dotted.

Delay 2

Same as **Delay 1**.

Feedback

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

Controls the output level of the effect.

Modulation

You can choose between 3 basic modulation characteristics:

- **Phaser** uses an 8-pole all-pass filter to produce the classic phasing effect.
- **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.

Song Sync

Activates/Deactivates tempo sync of the **Rate** parameter.

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

Controls the depth of the delay time modulation.

Delay

Sets the delay time of the four delay lines.

Feedback

Controls the amount of positive or negative feedback for all four delay lines.

Level

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 useful to emulate the lo-fi sounds of older digital synths.

- If the **F** button is active, the selected Part's program plays back with the sample rate set in the host application.

- If the **1/2** button is active, the selected Part's program plays back with half the original sample rate.
- If the **1/4** button 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 for 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).

RELATED LINKS

[Modulation and Controllers on page 215](#)

Sound Parameters

Oscillator Section



A/B Waveform Pop-up Menus

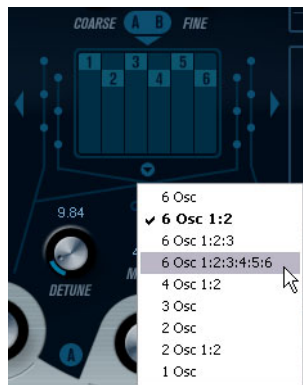
This is where you select basic waveforms for the A and B output of the oscillators. The options are especially suited for use with the spectrum filter.

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 selected oscillator configuration).



6 Osc

6 oscillators with the same pitch.

6 Osc 1:2

3 oscillators with base pitch and 3 pitched one octave down.

6 Osc 1:2:3

Three groups of two oscillators with the pitch ratio 1:2:3 (2 oscillators with base pitch, 2 oscillators at half the frequency of the base pitch, and 2 oscillators at a third of the frequency).

6 Osc 1:2:3:4:5:6

6 oscillators tuned with the pitch ratio 1:2:3:4:5:6 (known as the subharmonic series).

4 Osc 1:2

2 oscillators with base pitch and 2 pitched one octave down.

3 Osc

3 oscillators with the same pitch.

2 Osc

2 oscillators with the same pitch.

2 Osc 1:2

One oscillator with base pitch and one pitched one octave down.

1 Osc

A single oscillator. In this mode, the **Detune** and **Cut II** parameters are not active.

Detune

Detunes the oscillators. Low values give gentle chorus-like detuning. Raising the control detunes the oscillators by several semitones for special effects.

Raster

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

- If **0** is selected, all harmonics are present.
- If **1** is selected, only every second harmonic is present.
- If **2** is selected, only every third harmonic is present.

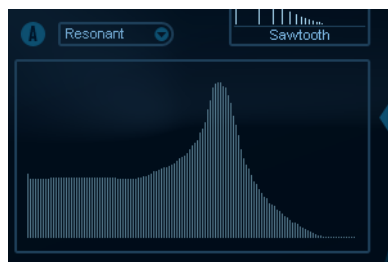
And so on.

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 if you play a legato note. Legato mode only works with monophonic parts.

Spectrum Filter Sections



This is where you create the contours, that is, 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.
- To change the contour, click and draw with the mouse.
- If you want to calculate a random spectrum filter curve, select **Randomize** from the Preset pop-up menu.

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

Cut I and II



These parameters work 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.

NOTE

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

Controls the mix between the sound of spectrum filters A and B. If the **Morph** knob is turned fully left, only the A sound is heard. If it is turned right only the B sound is heard. This allows you to seamlessly morph between two totally different sounds.

Master Volume and Pan

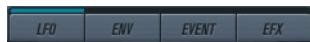


The master **Volume** knob controls the master volume (amplitude) of the instrument. By default, this parameter is controlled by Envelope 1, to generate an amplitude envelope for the oscillators.

The **Pan** knob controls the position of the instrument in the stereo spectrum. 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 that can be assigned to control parameters.
- The **Event** page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments.

- The **Effect** page offers three separate effect types: Distortion, Delay, and Modulation.

RELATED LINKS

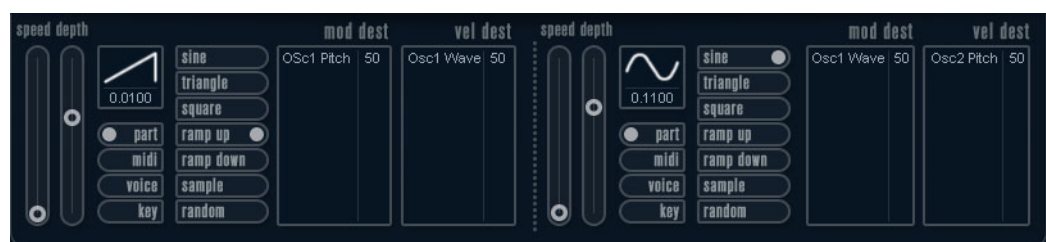
[Envelope Page on page 218](#)

[Event Page on page 221](#)

[Effects \(EFX\) Page on page 222](#)

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 selected preset, there may already be modulation destinations assigned, in which case these are listed in the **Mod Dest** box for each LFO.

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.

Speed

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

Controls the amount of modulation applied by the LFO. If this is set to zero, no modulation is applied.

Waveform

Sets the LFO waveform.

Sync mode (Part/MIDI/Voice/Key)

Sets the sync mode for the LFO.

RELATED LINKS

[Assigning LFO Modulation Destinations on page 217](#)

About the Sync Modes

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

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 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.

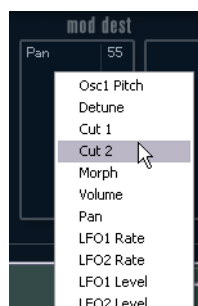
Assigning LFO Modulation Destinations

You can assign a modulation destination for an LFO.

PROCEDURE

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 [Enter].

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.

PROCEDURE

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.

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

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 velocity destination click on its name in the list and select **Off** from the pop-up menu.
-

LFO modulation velocity control

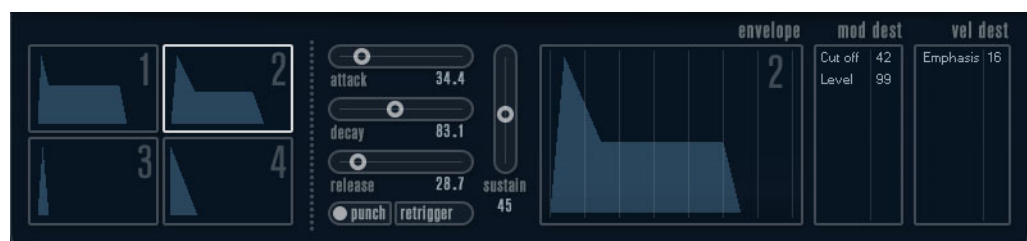
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- 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 2 ways: by using the sliders or by clicking and 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 is assigned, there is no output.
- Envelope 2 is by default assigned to the **Level** parameter.

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** parameter. The **Decay** has no effect if the **Sustain** parameter is set to maximum.

Sustain

Determines the level the envelope rests at after the **Decay** phase. Note that **Sustain** represents a level, whereas the other envelope parameters represent times.

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

If **Punch** is activated, the start of the decay phase is delayed a few milliseconds, that is, 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

If **Retrigger** is activated, the envelope retriggers 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.

Assigning Envelope Modulation Destinations

You can assign a modulation destination for an envelope.

PROCEDURE

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 [Enter].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.

PROCEDURE

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 [Enter].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 velocity destination click on its name in the list and select **Off** from the pop-up menu.

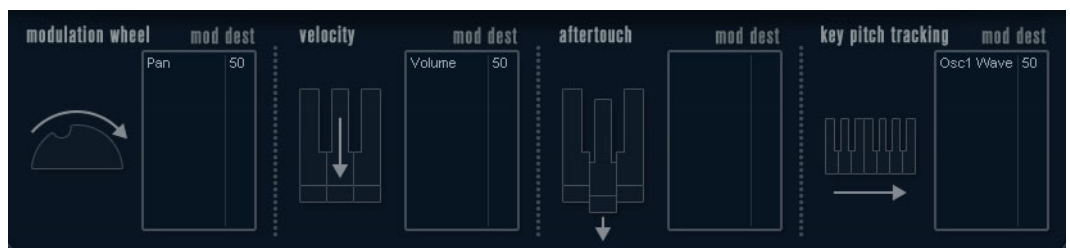
Envelope modulation velocity control

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.



Modulation Wheel

The modulation wheel on your keyboard can be used to modulate parameters.

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.

Key Pitch Tracking

This can change parameter values linearly according to where on the keyboard you play.

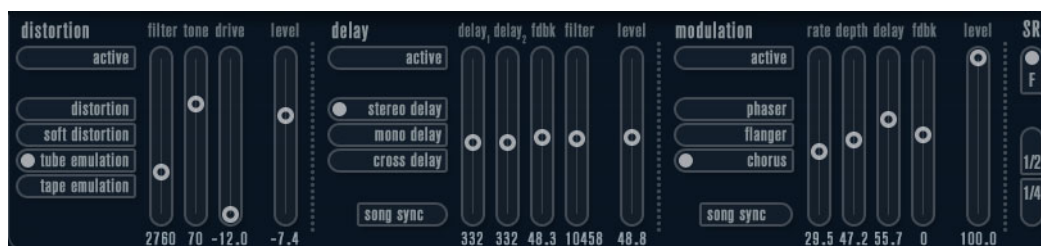
Assign a Controller to a Parameter

PROCEDURE

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 [Enter].
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 choose 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.

Drive

Sets the amount of distortion by amplifying the input signal.

Filter

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

Controls the relative amount of low-pass and high-pass filtered signal.

Level

Controls the output level of the effect.

Delay

You can choose 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.

Song Sync

Activates/Deactivates tempo sync of the delay times.

Delay 1

Sets the delay time ranging from 0 ms to 728 ms. If **MIDI sync** is activated, the range is from 1/32 to 1/1; straight, triplet or dotted.

Delay 2

Same as **Delay 1**.

Feedback

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

Controls the output level of the effect.

Modulation

You can choose between 3 basic modulation characteristics:

- **Phaser** uses an 8-pole all-pass filter to produce the classic phasing effect.

- **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.

Song Sync

Activates/Deactivates tempo sync of the **Rate** parameter.

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

Controls the depth of the delay time modulation.

Delay

Sets the delay time of the four delay lines.

Feedback

Controls the amount of positive or negative feedback for all four delay lines.

Level

Controls the output level of the effect.

SR Parameters

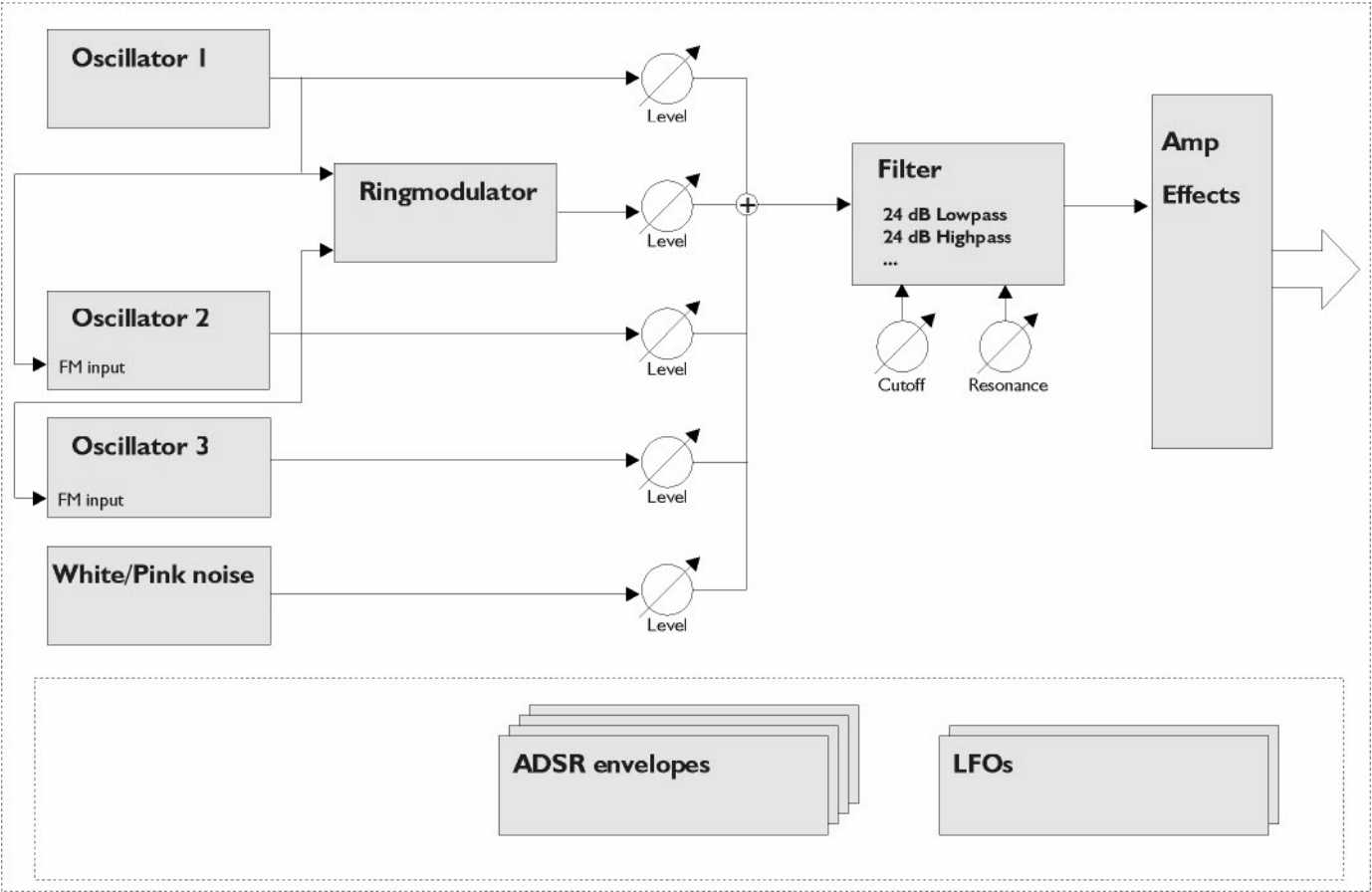
With these buttons, you can change the sample rate. Lower sample rates basically reduce the high frequency content and sound quality, but the pitch is not altered. This is useful to emulate the lo-fi sounds of older digital synths.

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

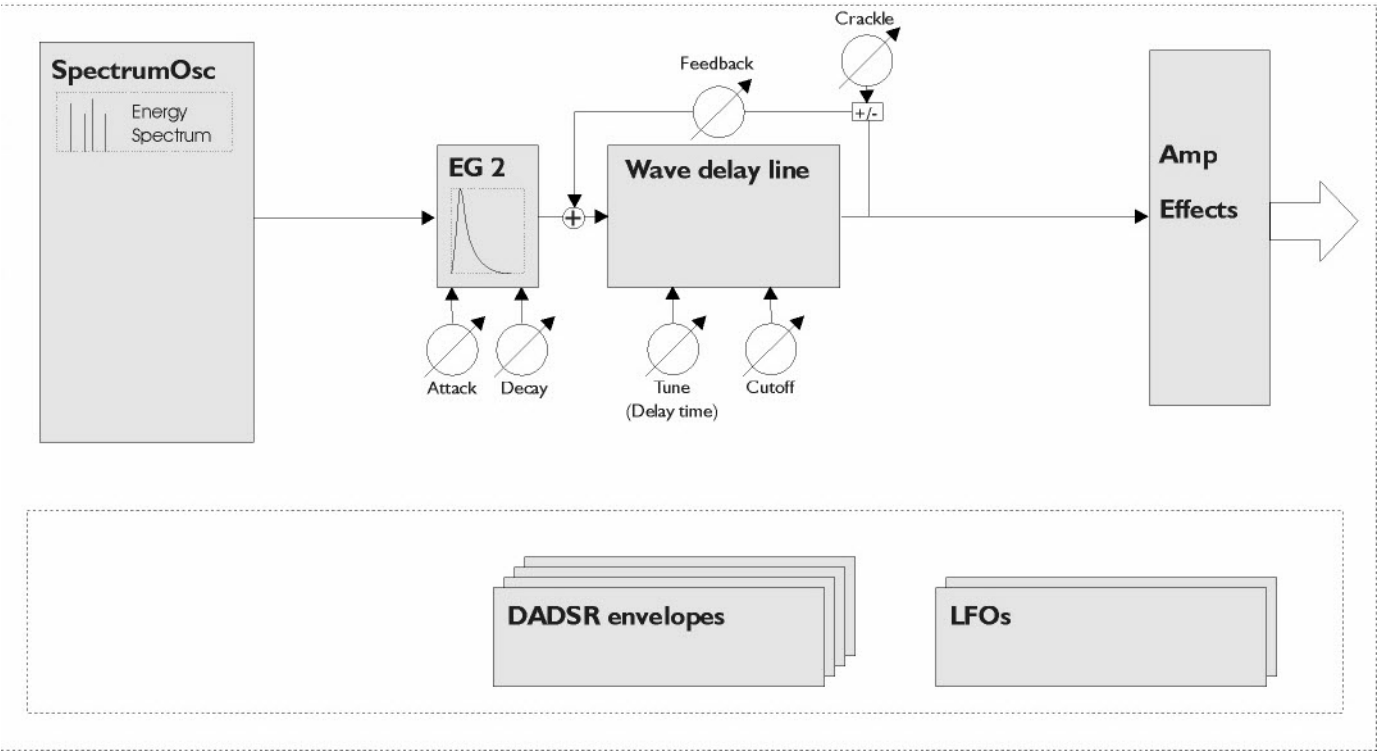
A bonus effect of using lower sample rates is that it reduces the load on the computer CPU, allowing for more simultaneous voices to be played, etc.

Diagrams

Prologue



Mystic



Spector

