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Included Effect Plug-ins

The included plug-in effects are arranged according to their categories.

Ambisonics Plug-ins

VST AmbiConverter

**VST AmbiConverter** allows you to convert Ambisonics audio between Furse-Malham (FuMa) and AmbiX format.

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For a description of **VST AmbiConverter**, 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.

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Delay

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

Delay – Sync
Activates/Deactivates tempo sync for the Delay parameter.

Rate

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 Sync is deactivated, the modulation speed can be set freely with the Rate knob.

Rate – Sync
Activates/Deactivates tempo sync for the Rate parameter.

Width
Sets the amount of delay 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 signal and the wet signal. If the effect is used as a send effect, set this parameter 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 tempo sync is activated, this parameter sets the base note value for synchronizing the modulation to the tempo of the host application.

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

Speed – Sync

Activates/Deactivates tempo sync for the Speed parameter.

Range Lo/Hi

Set 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 with the Speed knob.

Speed – Sync

Activates/Deactivates tempo sync for the Speed parameter.

Range Lo/Hi

Set 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.

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Lo Filter
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.

Hi Filter
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.

Delay
If tempo sync is activated, this sets 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.

Mix
Sets the level balance between the dry signal and the wet signal. If the effect is used as a send effect, set this parameter 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.

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**NOTE**

This plug-in works only on stereo tracks.

**Lo Filter**

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.

**Hi Filter**

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.

**Delay**

If tempo sync is activated, this sets 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.

**Mix**

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

**Spatial**

Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo ping-pong effect.
Start Left/Start Right
Determine whether the delay repeat starts on the left or the right channel.

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.

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NOTE
This plug-in works only on stereo tracks.

Feedback
Set the number of repeats for each delay.

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

Lo Filter
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.

Pan
Sets the stereo position.

Hi Filter
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.

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.

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.

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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.

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Included Effect Plug-ins
Distortion Plug-ins
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.

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

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 (0 to 24 bits)
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.

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

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

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

Output
Sets the output level.
Distortion

**Boost**

Increases the distortion amount.

**Oversampling**

Activates/Deactivates oversampling. Oversampling results in less artifacts for higher distortion.

**NOTE**

If this parameter is activated, the effect requires more processing power.

**Mix**

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

**Tone**

Changes the tonal characteristic of the output signal.

**Feedback**

Feeds part of the output signal back to the effect input. Higher settings increase the distortion effect.

**Spatial**

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

*Distroyer* adds harmonics to the spectrum and allows for distortion effects from smooth overdrive to extreme clipping.

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The following parameters influence only the wet signal:

**Lo Filter**

Changes the cutoff frequency of the low-pass filter that is applied to the wet signal before it gets distorted.

**Hi Filter**

Changes the cutoff frequency of the high-pass filter that is applied to the already distorted wet signal.

**Offset**

Modifies the symmetry of the distortion effect by changing the operation point of the characteristic.

**Drive**

Changes the characteristic of the distortion effect. Lower values lead to a smooth overdrive-like effect. Higher values change the shape of the signal towards a rectangle, leading to extreme distortion.
Oversampling
Activates/Deactivates oversampling. Oversampling results in less artifacts for higher distortion.

NOTE
If this parameter is activated, the effect requires more processing power.

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

DC Filter
Removes DC offset that occurs when using high Offset values.

The following parameters influence both the dry and the wet signal:

Boost
Increases the distortion amount.

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

Shelf Freq
Sets the frequency of the high shelving filter.

Shelf Gain
Sets the gain of the high shelving filter.

Tone
Sets the frequency of the output low-pass filter.

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.

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Included Effect Plug-ins
Distortion Plug-ins
Noise
Sets the amount of added static noise.

Crackle
Adds crackle to create that old vinyl record sound. The speed switch sets the virtual speed of the record in RPM (revolutions per minute).

Distort
Adds distortion.

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

AC
Emulates a constant, low AC hum. The frequency switch sets the virtual frequency of the AC current (50 or 60 Hz), and thus the pitch of the AC hum.

Mix
Sets the amount of overall effect.

Magneto II
Magneto II simulates the saturation and compression of recording on analog tape machines.

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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 machines.

Frequency Range Low/High
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 Low value to 200 Hz or 300 Hz. To avoid the saturation of very high frequencies, set the High parameter to values below 10 kHz.

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.

Quadrafuzz v2
Quadrafuzz v2 is a multi-band distortion and multi-effect plug-in for processing drums and loops but also for treatment of vocals, for example. You can distort up to 4 bands. 5 different distortion modes with several sub-modes are available.

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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 the output level. The vertical value scale to the left shows the gain level of each frequency band. The horizontal scale shows the available frequency range.

- To define the frequency range of the different frequency bands, use the handles at the sides of each frequency band.
- To attenuate or boost the output level of each frequency band by ±15 dB, use the handles on top of each frequency band.

Global Settings

SB

Switches between multi band and single band mode.

Scenes

You can save up to 8 different settings. If the default setting of a scene is active, the selected scene button lights up yellow.

If you change the default settings, the button lights up green, indicating that this scene has customized settings.
To copy the settings of a scene to another scene, select the scene that you want to copy, click **Copy**, and click one of the numbered buttons. You can automate the selection of scenes.

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

**Output (-24 to 24 dB)**
Sets the output level.

**Band Settings**

**Mute**
To mute each frequency band, activate the **M** button in each section.

**Bypassing Frequency Bands**
To bypass each frequency band, activate the **Bypass Band** button in each section.

**Soloing Frequency Bands**
To solo a frequency band, activate the **S** button in each section. Only one band can be soloed at a time.

**In/Out meter**
Display the input and output level.

**Gate**
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.

**Tape**
This band mode simulates the saturation and compression of recording on analog tape machines.

**Drive**
Controls the amount of tape saturation.

**Tape Mode Dual**
Simulates the use of two machines.

**Tube**
This band mode simulates the saturation effects using analog tubes.

**Drive**
Controls the amount of tube saturation.

**Tubes**
Determine the number of tubes that are simulated.

**Dist**
This band mode adds distortion to your tracks.
Drive
Controls the amount of distortion.

FBK
Feeds part of the output signal back to the effect input. Higher settings increase the distortion effect.

Amp
This band mode simulates the sound of various types of guitar amps.

Drive
Controls the amount of amp overdrive.

Amp Types
You can select the following types of guitar amps:
- Amp Clean
- Amp Crunch
- Amp Lead

Dec
This band mode allows you to decimate and truncate the input audio signal to create a noisy, distorted sound.

Decimator
Controls the resulting bit-resolution. The lower the resolution, the higher the distortion effect.

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.

S&H
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.

Delay
To open the Delay section, click the Delay button.

Time
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 delay time can be set freely with the Time knob.

Sync
Activates/Deactivates tempo sync for the corresponding delay.

Duck
Determines how much the delay signal ducks when an audio signal is present.

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

FBK
Determines the number of repeats for each delay.
Mode
If this option is activated, the delay signal is routed back into the distortion unit to create a feedback with distortion.

NOTE
High **FBK** values and low **Duck** values in combination with activated **Mode** can lead to unwanted noise.

Slider

Width
Sets the stereo width for the corresponding band.

Out
Sets the output gain for the corresponding band.

Pan
Sets the stereo position for the corresponding band.

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

SoftClipper

**SoftClipper** adds soft overdrive, with independent control over the second and third harmonic.

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Input (-12 to 24 dB)
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 signal and the wet signal.

Output
Sets the output level.

Second
Controls the second harmonic.
Third
Controls the third harmonic.

**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.

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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 – Sets the amount by which an input signal is being compressed.
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 signal 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 signal 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 signal 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 signal 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 70s and 80s.

Diamond
- The cutting edge hard rock and metal sounds of the 90s.

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 60s.

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.

Selecting and Deactivating 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.

Selecting and Deactivating 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 different 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. You can crossfade between the characteristics of the two microphones.

- To select one of the microphone types or blend between the two types, turn the **Mix** control between the two microphones.

**Placing the Microphone**

- To select a microphone position, click the corresponding ball in the graphic. The selected position is marked in red.

**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.
  - If the pitch is too low, red LEDs are lit on the left. If the pitch is too high, red LEDs are lit on the right. The more LEDs are lit, the lower/higher is the pitch.
- To mute the output signal of the plug-in, click the pedal-like **Master** button. If the output is muted, the LED is not lit. Use this to tune your guitar in silence, for example.
- To change the volume of the output signal, use the **Level** control on the Master page.

**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

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 buttons at the top of the plug-in panel 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 display section 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 you move the mouse pointer over 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.
VST Bass Amp

**VST Bass Amp** is a bass amp simulator. It offers a choice of amplifiers and speaker cabinets that can be combined with stomp box effects.

<table>
<thead>
<tr>
<th>Cubase LE</th>
<th>Cubase AI</th>
<th>Cubase Elements</th>
<th>Cubase Artist</th>
<th>Cubase Pro</th>
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Included with

At the top of the plug-in panel, the following buttons open different pages in the display section of the plug-in panel: **Pre-Effects**, **Amplifiers**, **Cabinets**, **Post-Effects**, **Microphones**, **Configuration**, and **Master**.

These buttons are arranged according to the position of the corresponding elements in the signal chain.

Below the display section, the selected bass amplifier is displayed. The color and texture of the area below the bass amp indicate the selected cabinet.

**Pre/Post-Effects**

On the **Pre-Effects** and **Post-Effects** pages, you can select up to six common bass effects. On both pages, the same effects are available, the only difference being the position in the signal chain (before or after the bass 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.

**Envelope Filter**

**Range** – Determines the frequency range.

**Q-Factor** – Sets the intensity of the envelope filter effect.

**Sensitivity** – Determines how sensitive the effect reacts to the instrument level.

**Attack** – Determines how quickly an effect reacts to the input signal.
Mix – Sets the level balance between the dry signal and the wet signal.
Type – Sets the filter type.
Release – Determines how quickly the effect fades after the input signal stops.

Volume
Pedal – Controls the level of the signal passing through the effect.

Compressor
Intensity – Sets the amount by which an input signal is being compressed.

Compressor MB
Lo Intensity – Sets the compressor effect in the low frequency band. Activate/deactivate Auto Makeup Mode by clicking the LED at the top right of the knob.
Hi Intensity – Sets the compressor effect in the high frequency band. Activate/deactivate Auto Makeup Mode by clicking the LED at the top right of the knob.
Crossover – Determines the crossover frequency between the low frequency band and the high frequency band.
Output – Sets the output level.

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.
Tone – Allows you to attenuate low frequencies.
Mix – Sets the level balance between the dry signal and the wet signal.

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.
Tone – Allows you to attenuate the low frequencies.
Mix – Sets the level balance between the dry signal and the wet signal.

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.
Tone – Allows you to attenuate the low frequencies.
Mix – Sets the level balance between the dry signal and the wet signal.

DI Driver
Level – Sets the output level.
Blend – Blends between normal and tube emulation circuitry. With Blend at 0, Drive and Presence are not active.
**Bass** – Boosts or attenuates low frequencies.
**Treble** – Boosts or attenuates high frequencies.
**Presence** – Boosts or attenuates upper harmonics and attacks.
**Drive** – Sets gain and overdrive.

**Enhancer**

**Enhance** – Simulates the classic enhancer effect.
**Tone** – Allows you to attenuate low frequencies.

**Octaver**

**Direct** – Adjusts the level of the original signal. 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.
**Tone** – Changes the sound character of the generated signal.

**Delay**

**Delay** – Sets the delay time in milliseconds. This parameter can be synchronized to the project tempo.
**Feedback** – The higher this setting, the more delay repeats are created.
**Mix** – Sets the level balance between the dry signal and the wet signal.

**Tape Ducking Delay**

**Delay** – The **Delay** parameter sets the delay time in milliseconds.
**Feedback** – The higher this setting, the more delay repeats are created.
**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.
**Tone** – Allows you to attenuate the low frequencies.
**Mix** – Sets the level balance between the dry signal and the wet signal.

**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.

**Magneto II**

**Drive** – Controls the amount of tape saturation.
**Low/High** – These parameters set the frequency range of the spectrum band to which the tape effect is applied.
**HF-Adjust** – Sets the amount of high frequency content of the saturated signal.

**Gate**

**Threshold** – Determines the level at which the gate is activated. Signal levels above the set threshold open the gate 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.

Graphical EQ

- **Display** – Consists of 8 sliders that set the level of each frequency band. Allows you to draw response curves by clicking and dragging with the mouse.
- **Reset Sliders** – At the lower right of the Display. Flattens all values to 0 dB.
- **Output Slider** – Allows you to control the frequency response.

Reverb

- **Type** – A convolution-based reverb effect. This parameter allows you to switch between the reverb types Studio, Hall, Plate, and Room.
- **Mix** – Sets the level balance between the dry signal and the wet signal.

Sync

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 Bass Amp 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 bass recording, such as gain, equalizers, and master volume. The sound-related parameters bass, low mid, high mid, and treble have a significant impact on the
overall character and sound of the corresponding amp. Shape 1 and Shape 2 offer predefined tone shaping.

**ValveAmp300**
A famous tube amplifier from the 70s, useful for rock playing styles.

**Greyhound**
An amplifier, well known for its typical growl, useful for several playing styles.

**GreenT**
A classic amplifier from the 80s, useful for funk and rock playing styles.

**Paradise**
An amplifier from the 90s, with a hi-fi-like clear tone, that makes it applicable for several styles.

**Tweed**
A classic vintage amplifier from the 50s, with a characterful and bright tone. Originally created for bassists, also used by many guitar players.

**iTech**
A modern amplifier, with a universal sound.

The different amps keep their settings if you switch models, but amp settings are lost when closing **VST Bass AMP**. If you want to use the same settings after reloading the plug-in, you need to set up a preset.

**Selecting and Deactivating 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.

**NOTE**
To scroll through amplifiers, use the mouse wheel when hovering over the amplifier panel.

---

**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.

The following cabinets are available:

**4x10”**
10” speakers provide a punchy clear sound that is suitable for “Slap” bass and regular playing styles.
10” speakers have a cleaner sound and more punch than 15” speakers.

**8x10”**
Compared to 4x10”, double the amount of speakers.

**4x12”**
12” speakers provide a mellow and full sound, making them a good choice between 10” and 15” speakers.

**1x15”**
15” speakers provide more low frequencies compared to the other cabinets. They are suitable for rock and vintage oriented styles.

**Selecting and Deactivating 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 different microphone types.

57  
Dynamic microphone with cardioid pickup pattern.

121  
Ribbon microphone with figure-8 pattern.

409  
Dynamic microphone with supercardioid pickup pattern.

421  
Dynamic microphone with cardioid polar pattern.

545  
Dynamic microphone with cardioid pattern that minimizes feedback.

5  
Dynamic microphone with cardioid pickup pattern.

30  
Reference and measurement microphone with omni directional polar pattern.

87  
Condenser microphone with omni directional pattern.

You can choose between different microphone positions. These positions result from two different angles (on axis and off axis) and three different distances from the cabinet. You can crossfade between the characteristics of the two microphones.

- To select one of the microphone types or blend between the two types, turn the Mix control between the two microphones.
- To select a microphone position, click the corresponding ball in front of the cabinet. The selected position is marked in red.
- To determine the ratio between line and mic, turn the Mix control on the left of the cabinet.

**NOTE**

To scroll through microphones, use the mouse wheel when hovering over a microphone.

**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 all other views, 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 **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.

If the pitch is too low, red LEDs are lit on the left. If the pitch is too high, red LEDs are lit on the right. The more LEDs are lit, the lower/higher is the pitch.

To mute the output signal of the plug-in, click the pedal-like **Master** button. If the output is muted, the LED is not lit. Use this to tune your guitar in silence, for example.

To change the volume of the output signal, use the **Level** control on the **Master** page.

**NOTE**

Master EQ functions only when a cabinet is selected.

---

**Configuration**

On the **Configuration** page, you can specify whether you want to use **VST Bass Amp** 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. Use mono configuration on a stereo track to save processing power.

---

**View Settings**

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 buttons at the top of the plug-in panel 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 display section 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 you move the mouse pointer over 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.

### RELATED LINKS

- **Tuner** on page 145

### Dynamics Plug-ins

#### Brickwall Limiter

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

<table>
<thead>
<tr>
<th>Cubase LE</th>
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<tbody>
<tr>
<td>Included with</td>
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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 1ms. **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)
Determines the level where the limiter kicks in. Only signal levels above the set threshold are processed.

Release (3 to 1000 ms or Auto mode)
Sets the time after which the gain returns to the original level when the signal drops below the threshold. If the Auto button is activated, the plug-in automatically finds the best release setting for 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 uses oversampling to detect and limit 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
Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both.

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<th></th>
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<td>Included with</td>
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<td>Side-chain support</td>
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Compressor features a separate display that graphically illustrates the compressor curve that is 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 0 dB)**
Determine the level where the compressor kicks in. Only signal levels above the set threshold are processed.

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

**High Ratio**
Sets the ratio to a fixed value of 20:1.

**Make-Up (0 to 24 dB 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.

**Dry Mix**
Mixes the dry input signal to the compressed signal.

**Attack (0.1 to 100 ms)**
Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal passes through unprocessed.
Hold (0 to 5000 ms)
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 1000 ms 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 Release button is activated, the plug-in automatically finds the best release setting for the audio material.

Analysis (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 the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If Live mode is activated, there is no latency, which is better for live processing.

DeEsser
DeEsser reduces excessive sibilance, primarily for vocal recordings. It is a special type of compressor that is tuned to be sensitive to the frequencies produced by the s-sound.

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

**Display**

Shows the spectrum of the input signal.

- To adjust the frequency band, drag the border lines or click in the middle of the band and drag.
- To change the width of the frequency band, hold Shift and drag to the left or right.

**Filter**

**Lo/Hi**

Sets the left and right border of the frequency band. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically displayed in Hz accordingly. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

**NOTE**

Make sure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

**Solo**

Solos the frequency band. This helps you to find the appropriate position and width of that band.

**Diff**

Plays back what DeEsser removed from the signal. This helps you to adjust the frequency band, threshold, and reduction parameters, so that only sharp s-sounds are removed, for example.
Dynamics

Reduction
Controls the intensity of the de-essing effect.

Threshold (-50 to 0 dB)
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.

Release (1 to 1000 ms)
Sets the time after which the de-essing effect returns to zero when the signal drops below the threshold.

Auto
Automatically and continually sets an optimum threshold setting independent of the input signal. The Auto option does not work for low-level signals (< -30 dB peak level). To reduce the sibilants in such a file, set the threshold manually.

Side-Chain

Freq (25 Hz to 20 kHz)
If the Side-Chain button is activated, this sets the frequency of the filter. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically displayed in Hz accordingly. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

NOTE
Make sure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

Q-Factor
If the Side-Chain button is activated, this sets the resonance or width of the filter.

Side
Activates the internal side-chain filter. You can now shape the input signal according to the filter parameters. Internal side-chaining can be useful for tailoring how the gate operates.

Monitor
Allows you to monitor the filtered signal.

Live
If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If Live mode is activated, there is no latency, which is better for live processing.

Positioning the DeEsser in the Signal Chain
When recording a voice, the position of DeEsser 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 can be used to attenuate or boost the gain of the attack and release phase of audio material.

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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 20 dB)**
Sets the gain of the attack phase of the signal.

**Length (5 to 200 ms)**
Sets the length of the attack phase.

**Release (-20 to 20 dB)**
Sets the gain of the release phase of the signal.

**Output**
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.

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Threshold (-60 to 0 dB)

Determines the level where the expansion kicks in. Only signal levels below the set threshold are processed.

Ratio

Sets 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 less drastic results.

Fall (0.1 to 100 ms)

Determines how fast the expander responds to signals below the set threshold. If the fall time is long, more of the early part of the signal passes through unprocessed.

Hold (0 to 2000 ms)

Sets the time the applied expansion affects the signal below the threshold.
Rise (10 to 1000 ms or Auto mode)
Sets the time after which the gain returns to its original level when the signal exceeds the threshold. If the Auto Rise button is activated, the plug-in automatically finds the best rise setting for the audio material.

Analysis (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 the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific 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 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 threshold, the gate opens to let the signal through.

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Included Effect Plug-ins
Dynamics Plug-ins
Attack (0.1 to 1000 ms)
Sets the time after which the gate opens when it is 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 2000 ms)
Determines how long the gate remains open after the signal drops below the threshold level.

Release (10 to 1000 ms or Auto mode)
Sets the time after which the gate closes (after the set Hold time). If Auto Release is activated, Gate automatically finds the best release setting for the audio material.

Threshold (-60 to 0 dB)
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 in an intermediate state (LED lights up in yellow).

Analysis (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.

Range
Adjusts the attenuation of the gate when it is shut. If Range is set to minus infinite dB, the gate is completely shut. The higher the value, the higher the level of the signal that passes through the shut gate.
Live
If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If Live mode is activated, there is no latency, which is better for live processing.

Side-Chain Section

Side-Chain
Activates 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 gate operates.

Monitor
Allows you to monitor the filtered signal.

Center (50 to 20000 Hz)
If the Side-Chain button is activated, this sets the center frequency of the filter.

Q-Factor
If the Side-Chain button is activated, this sets the resonance or width of the filter.

Filter buttons (Low-Pass/Band-Pass/High-Pass)
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.

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

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

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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 24 dB)**
Sets the input gain.

**Release (0.1 to 1000 ms or Auto mode)**
Sets the time after which the gain returns to its original level. If Auto Release is activated, the plug-in automatically finds the best release setting for the audio material.

**Output**
Sets the maximum output level.

**Maximizer**
Maximizer raises the loudness of audio material without the risk of clipping. The plug-in provides two modes, Classic and Modern, that offer different algorithms and parameters.

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Classic

Classic mode provides the classic algorithms from previous versions of this plug-in. This mode is suited for all styles of music.

Modern

In Modern mode, the algorithm allows for more loudness than in Classic mode. This mode is particularly suited for contemporary styles of music. Modern mode also provides additional settings to control the release phase:

- Release sets the overall release time.
- Recover allows for a faster signal recovering at the beginning of the release phase.

Optimize

Determines the loudness of the signal.

Mix

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

Output

Sets the maximum output level.

Soft Clip

If this button is activated, Maximizer starts limiting or clipping the signal softly. At the same time, harmonics are generated, adding a warm, tube-like characteristic to the audio material.
MIDI Gate

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

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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. Therefore, it needs both audio and MIDI data to function.

**Attack (0 to 500 ms)**
Sets the time after which the gate opens when it is triggered.

**Hold**
Determines how long the gate remains open after a note-on or note-off message. The Hold Mode settings are taken into account.

**Release (0 to 3000 ms)**
Sets the time after which the gate closes after the set Hold time.

**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 keyboard.

**RESULT**

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

**RELATED LINKS**

- **Setting Up MIDI Gate** on page 49

**MultibandCompressor**

**MultibandCompressor** allows a signal to be split into four frequency bands. You can specify the level, bandwidth, and compressor characteristics for each band.

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NOTE

To compensate for output gain loss that is caused by compression, **MultibandCompressor** uses an automatic make-up gain. If side-chaining is activated for a frequency band in the side-chain section, the automatic make-up gain is deactivated for this 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 gain level of each frequency band. The horizontal scale shows the available frequency range.

- To define the frequency range of the different frequency bands, use the handles at the sides of each frequency band.
- To attenuate or boost the gain of the frequency bands by ±15 dB after compression, use the handles at the top of each frequency band.

**Live**

If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If **Live** mode is activated, there is no latency, which is better for live processing.

**Bypassing Frequency Bands**

To bypass each frequency band, activate the **Bypass Band** button in each section.

**Soloing Frequency Bands**

To solo a frequency band, activate the **S** button in each section. Only one band can be soloed at a time.
Output (-24 to 24 dB)
Sets the output level.

Compressor Section
You can specify the **Threshold** and **Ratio** by moving breakpoints or using the corresponding knobs. The threshold is marked by the first breakpoint where the line deviates from the straight diagonal.

**Threshold (-60 to 0 dB)**
Determines the level where the compressor kicks in. Only signal levels above the set threshold are processed.

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

**Attack (0.1 to 100 ms)**
Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal passes through unprocessed.

**Release (10 to 1000 ms 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 Release** button is activated, the plug-in automatically finds the best release setting for the audio material.

Side-Chain Section
To open the side-chain section, click the **SC** button at the bottom left of the plug-in window.

**IMPORTANT**
To be able to use the side-chain function for the bands, global side-chain must be activated for the plug-in.

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

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

Side-Chain
Activates the internal side-chain filter. The side-chain signal can then be shaped according to the filter parameter.

Monitor
Allows you to monitor the filtered signal.
MultibandEnvelopeShaper allows a signal to be split into four frequency bands. You can attenuate or boost the gain of the attack and release phase of audio material for each band.

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**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. The vertical value scale to the left shows the gain level of each frequency band. The horizontal scale shows the available frequency range.

- To define the frequency range of the different frequency bands, use the handles at the sides of each frequency band.
- To attenuate or boost the gain of the frequency band, use the handles at the top of each frequency band.

**Live**

If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If **Live** mode is activated, there is no latency, which is better for live processing.

**Bypassing Frequency Bands**

To bypass each frequency band, activate the **Bypass Band** button in each section.
Soloing Frequency Bands
To solo a frequency band, activate the $ button in each section. Only one band can be soloed at a time.

Output (-24 to 24 dB)
Sets the output level.

Shaper Section
You can specify the Attack, Length, and Release by moving breakpoints or using the corresponding knobs. Be careful with levels when boosting the gain. You can reduce the output level to avoid clipping.

Attack (-20 to 20 dB)
Sets the gain of the attack phase of the signal.

Length (5 to 200 ms)
Sets the length of the attack phase.

Release (-20 to 20 dB)
Sets the gain of the release phase of the signal.

Sensitivity (-40 to -10 dB)
Sets the sensitivity of the detection.

Output
Sets the output level.

MultibandExpander
MultibandExpander allows a signal to be split into four frequency bands. You can reduce the output level in relation to the input level for signals below the set threshold for each band. This is useful if you want to enhance the dynamic range or reduce the noise in quiet passages.

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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 expansion. The vertical value scale to the left shows the gain level of each frequency band. The horizontal scale shows the available frequency range.

- To define the frequency range of the different frequency bands, use the handles at the sides.
- To attenuate or boost the gain of the frequency band after expansion, use the handles on top of each frequency band.

**Live**

If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If **Live** mode is activated, there is no latency, which is better for live processing.

**Bypassing Frequency Bands**

To bypass each frequency band, activate the **Bypass Band** button in each section.

**Soloing Frequency Bands**

To solo a frequency band, activate the **S** button in each section. Only one band can be soloed at a time.

**Output (-24 to 24 dB)**

Sets the output level.
Expander Section

You can specify the **Threshold** and **Ratio** by moving breakpoints or using the corresponding knobs. The first breakpoint from which the line deviates from the straight diagonal is the threshold point.

**Threshold (-60 to 0 dB)**
Determines the level where the expansion kicks in. Only signal levels below the set threshold are processed.

**Ratio**
Sets the amount of gain boost applied to signals below the threshold.

**Maximum Reduction**
Sets the maximum amount by which the level is reduced when the signal falls below the set threshold.

**Fall (0.1 to 100 ms)**
Determines how fast the expander responds to signals below the set threshold. If the fall time is long, more of the early part of the signal passes through unprocessed.

**Hold (0 to 2000 ms)**
Sets the time the applied expansion affects the signal below the threshold.

**Rise (10 to 1000 ms or Auto mode)**
Sets the time after which the gain returns to its original level when the signal exceeds the threshold. If the **Auto Rise** button is activated, the plug-in automatically finds the best rise setting for the audio material.

**Output**
Sets the output level.

Side-Chain Section

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

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

**Side-Chain**
Activates the internal side-chain filter. The side-chain signal can then be shaped according to the filter parameters. Side-chaining is useful for tailoring how the effect operates.

**Monitor**
Allows you to monitor the filtered signal.
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.

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Drive (1.0 to 6.0 dB)

Controls the amount of tube saturation.

Input

Determines the compression amount. The higher the input gain, the more compression is applied.

Ratio

Toggles between a low and a high ratio value.

Output (-12 to 12 dB)

Sets the output gain.

Character

Keeps the bass tight and preserves its attacks by decreasing the tube saturation for lower frequencies, and adds brilliance by creating harmonics for higher frequencies.

Attack (0.1 to 100 ms)

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 1000 ms or Auto mode)
Sets the time after which the gain returns to its original level. If the Auto Release button is activated, the plug-in automatically finds the best release setting for the audio material.

Mix
Adjusts the mix between dry signal 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 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 gate operates.

Side-chain section
Filter buttons (Low-Pass/Band-Pass/High-Pass)
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.

Center (50 to 20000 Hz)
If the Side-Chain button is activated, this sets the center frequency of the filter.

Q-Factor
If the Side-Chain button is activated, this sets the resonance or width of the filter.

Monitor
Allows you to monitor the filtered signal.

VintageCompressor
VintageCompressor is modeled after vintage type compressors.

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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.
**VSTDynamics**

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

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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 threshold, the gate opens to let the signal through. The gate trigger input can also be filtered using an internal side-chain signal.

The following parameters are available:

**Threshold (-60 to 0 dB)**
- 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 in an intermediate state (LED lights up in yellow).

**Side-Chain**
- Activates 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 gate operates.

**Filter buttons (Low-Pass/Band-Pass/High-Pass)**
- 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.

**Center (50 to 20000 Hz)**
- If the Side-Chain button is activated, this sets the center frequency of the filter.

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

**Monitor**
- Allows you to monitor the filtered signal.

**Attack (0.1 to 100 ms)**
- Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal passes through unprocessed.

**Hold (0 to 2000 ms)**
- Determines how long the gate remains open after the signal drops below the threshold level.
Release (10 to 1000 ms or Auto mode)
Sets the time after which the gate closes after the set Hold time. If the Auto Release button is activated, the plug-in 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 minus infinite dB, 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. The compressor features a separate display that graphically illustrates the compressor curve shaped according to your settings.

Threshold (-60 to 0 dB)
Determines the level where the compressor kicks in. Only signal levels above the set threshold are processed.

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

Make-Up (0 to 24 dB 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 100 ms)
Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.

Release (10 to 1000 ms or Auto mode)
Sets the time after which the gain returns to its original level when the signal drops below the threshold. If the Auto Release button is activated, the plug-in automatically finds the best release setting for 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
Sets the maximum output level.
**Soft Clip**
If this button is activated, the signal is limited when the signal level exceeds -6 dB. At the same time, harmonics are generated, adding a warm, tube-like characteristic to the audio material.

**Release (10 to 1000 ms or Auto mode)**
Sets the time after which the gain returns to its original level. If the Auto Release button is activated, the plug-in automatically finds the best release setting for 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.

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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. The filters of CurveEQ 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.

For detailed information about CurveEQ and its parameters, refer to the documentation provided by Voxengo at http://www.voxengo.com.
DJ-EQ

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

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**Graphical display**

Allows you to set the amount of boost or attenuation for the low, mid, or high band by dragging.

- To set the low, mid, and high frequency gain, click and drag the corresponding band handle.
- To fine-adjust the gain, press **Shift** and drag.
- To set a parameter to zero, press **Ctrl/Cmd** and click it.

**Low Frequency Gain/Mid Frequency Gain/High Frequency Gain**

Set the amount of boost or attenuation for the low, mid, and high band.

**Cut Low Frequencies/Cut Mid Frequencies/Cut High Frequencies**

Cut the low, mid, and high band.

**Output meter**

Indicates the level of the output signal.

**Frequency**

Frequency is a high-quality equalizer with 8 fully parametric bands. The bands can act as either shelving filter, as peak or notch filter (band-pass), or as cut filter (low-pass/high-pass).
Main Layout

Reset

Alt-click this button to reset all parameter values.

Auto Listen for Filters

If this option is activated and you edit a parameter of a band, the corresponding frequency range is isolated. This helps you to focus on a particular frequency range and allows you to locate unwanted frequencies in your audio.

Global Settings

Opens the settings dialog for the spectrum display.

Show/Hide Keyboard

Shows/Hides the keyboard below the graphical editor.
On the keyboard, color indicators reflect the center frequencies of all active equalizer bands. You can adjust the frequency of a band by dragging its color indicator. If you drag the color indicator of a band to a key, the band is set to its exact frequency.

Output

Adjusts the overall output level.
**Band Settings**

Activate/Deactivate Band
Activates/Deactivates the corresponding band.

**NOTE**
- To activate/deactivate a band, you can also double-click the corresponding handle in the graphical editor.
- If a band is deactivated, you can still modify its parameters.

Processing Switches
Allow you to switch between left/right, stereo, and mid/side processing. In **Left/Right** or **Mid/Side** processing mode, you can make different settings for the two channels.

**NOTE**
This setting is only available for stereo tracks.

Linear Phase
Activates/Deactivates linear phase mode for the corresponding band.
Linear phase mode avoids unwanted frequency dependent phase shifts of the audio signal that might occur with standard minimum phase equalizing.

**NOTE**
- Linear phase mode leads to an increase in latency.
- In rare cases, for example, when using low cut filtering with a high slope for bass signals, also an unwanted pre-ringing effect may be audible.

Filter type
You can choose between the filter types **Low Shelf**, **Peak**, **High Shelf**, and **Notch**. For band 1 and 8, you can also select the types **Cut 6**, **Cut 12**, **Cut 24**, **Cut 48**, and **Cut 96**.
- **Low Shelf** boosts or attenuates frequencies below the cutoff frequency by the specified amount.
- **Peak** boosts or attenuates frequencies at the set frequency value with a bell shaped filter.
- **High Shelf** boosts or attenuates frequencies above the cutoff frequency by the specified amount.
- **Notch** boosts or attenuates frequencies at the set frequency value with a very narrow filter.
- **Cut** attenuates frequencies below (band 1) or above (band 8) the set frequency. You can choose between different slopes: 6 dB, 12 dB, 24 dB, 48 dB, or 96 dB per octave.

**Freq**
Sets the frequency of the corresponding band. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

**NOTE**
- You can adjust the Freq parameter of a band in the graphical editor by **Alt**-clicking the corresponding handle and moving the mouse left and right.
- Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

If the band is active, the frequency value is reflected as a highlighted key on the keyboard below the graphical editor.

**Q**
For **Peak** and **Notch** filters, this parameter controls the width of the band. For **Low Shelf** and **High Shelf** filters, it adds a drop or a boost, depending on the gain setting of the band. For **Cut** filters, it adds a resonance.

**NOTE**
- You can adjust the Q parameter of a band in the graphical editor by **Shift**-clicking the corresponding handle and moving the mouse up and down. Alternatively, you can point on the handle and turn the mouse wheel.
- This parameter is not available for **Cut 6** filters.

**Gain**
Sets the amount of attenuation/boost for the corresponding band.

**NOTE**
- You can adjust the Gain parameter of a band in the graphical editor by **Ctrl**/**Cmd**-clicking the corresponding handle and moving the mouse up and down.
- This parameter is not available for **Cut** filters.

**Invert Gain**
Inverts the value of the gain parameter. Positive gain values become negative and vice versa.

**Global Settings**
- To open the **Global Settings**, click **Global Settings** above the spectrum display.

**Spectrum Display**

**Show Spectrum**
Activates/Deactivates the spectrum display.

**Peak Hold**
Holds the peak values of the spectrum display for a short time.
Smooth
Determines the reaction time of the spectrum display. Lower values result in faster reaction times.

Bar Graph
If this option is activated, the frequency spectrum is analyzed into 60 separate bands that are displayed as vertical bars.

Two Channels
If this option is activated, the spectrum of the left and right channels are displayed separately.

Slope
 Tilts the spectrum display around the 1 kHz pivot.

EQ Curve
Show Curve
Shows/Hides the EQ curve in the spectrum display.

Filled
If this option is activated, the EQ curve is filled. Amount allows you to specify the degree of coverage between 10 and 80 %.

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

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![Graphic Equalizer Image](image-url)
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. You have to click one of the sliders before you drag 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**
- Sets 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**
- Allows you to set the filter mode that determines how the various frequency band controls interact to create the response curve.

**EQ Modes**
- The Mode pop-up menu in the lower right corner allows you to select an EQ mode, which add color or character to the equalized output in various ways.

**True Response**
- Applies serial filters with an accurate frequency response.

**Digital 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 asym**
- Applies parallel filters where the resonance is raised when boosting the gain and vice versa.
**ConstQ sym**
Applies parallel filters where the resonance of the first and last bands depends on the sample rate.

**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. All four bands can act as fully parametric peak filters. In addition, the low and high bands can act as either shelving filters (three types) or as cut filters (low-pass/high-pass).

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**Main Layout**

**Reset**
Alt-click this button to reset all parameter values.

**Show Input/Output Spectrum**
Shows the spectrum before and after filtering.

**Output**
Adjusts the overall output level.

**Auto Gain**
If this button is activated, the gain is automatically adjusted, keeping the output level nearly constant regardless of the EQ settings.
Band Settings

Activate/Deactivate Band
Activates/Deactivates the corresponding band.

**NOTE**
- If a band is deactivated, you can still modify its parameters.

Freq
Sets the frequency of the corresponding band. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

**NOTE**
- You can adjust the Freq parameter of a band in the graphical editor by Alt-clicking the corresponding handle and moving the mouse left and right.
- Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

Inv
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.

Q
For Peak filters, this parameter controls the width of the band. For Shelf filters, it adds a drop or a boost, depending on the gain setting of the band. For Cut filters, it adds a resonance.

**NOTE**
- You can adjust the Q parameter of a band in the graphical editor by Shift-clicking the corresponding handle and moving the mouse up and down. Alternatively, you can point on the handle and turn the mouse wheel.

Gain
Sets the amount of attenuation/boost for the corresponding band.

**NOTE**
- You can adjust the Gain parameter of a band in the graphical editor by Ctrl/Cmd-clicking the corresponding handle and moving the mouse up and down.
- This parameter is not available for Cut filters.

Filter type
For the low and high band, you can choose between three types of shelving filters, a peak filter (band-pass), and a cut filter (low-pass/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**.

**Filter Plug-ins**

**DualFilter**

*DualFilter* filters out specific frequencies while allowing others to pass through.

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**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**

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

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Filter A buttons  
Allow you to select the characteristic of the first filter.  
- **Low Pass**  
    Eliminates high-frequency signal components. Filter slopes of 6, 12, 18, and 24 dB per decade are available.  
- **Band Pass**  
    Allows signals falling within a certain frequency range to pass through. Filter slopes of 12 and 24 dB per decade are available.

Filter B buttons  
Allow you to select the characteristic of the second filter.  
- **High Pass**  
    Eliminates low-frequency signal components. Filter slopes of 6, 12, 18, and 24 dB per decade are available.  
- **Band Rejection**  
    Lets all frequencies pass, except those in the stop band. Filter slopes of 12 and 24 dB per decade are available.

**Resonance Factor**  
Sets the resonance value of both filters simultaneously.

**Frequency**  
Sets the cutoff frequency of both filters simultaneously.

**Graphical display**  
Visualizes the settings for all parameters. The handle allows you to adjust the **Morph Factor** and the **Frequency** parameters simultaneously.

**Output meter**  
Indicates the level of the output signal.

**Morph Factor**  
Allows you to mix the output between both filters.
PostFilter

This effect allows quick and easy filtering of unwanted frequencies, creating room for the important sounds in your mix.

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PostFilter 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 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 audio.

Low Cut Freq (20 Hz 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. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

**NOTE**

Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

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 that you want to filter out.
**Notch Freq**

Sets the frequency of the notch filter. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

**NOTE**

Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

**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 (3 Hz to 20 kHz, 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. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

**NOTE**

Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

**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

StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects. You can also trigger pattern steps individually via MIDI.

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**General Operation**

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

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.

If Sync is deactivated, StepFilter allows you to trigger pattern steps individually via MIDI notes.

**Setting Step Values**

- To enter a step, click in the pattern grid windows.
- Individual step entries can be 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.
- Change the value for a step by dragging it up or down.
- Draw in a curve by clicking and dragging in the display.
Selecting New Patterns

- 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 slots to select a new pattern. New patterns are all set to the same step value by default.

StepFilter Parameters

Filter Type
Sets the filter type. A low-pass, a band-pass, and a high-pass filter are available.

Filter Mode
Allows you to choose between two filter modes. Classic mode is compatible with previous versions of this plug-in. Modern mode provides an additional Hard Clip parameter.

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

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.

Rate
If Sync is activated, Rate allows you to specify the base note value for synchronizing the pattern playback to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If Sync is deactivated, you can trigger pattern steps individually via MIDI.

Hard Clip
Adds more high frequency harmonics and distortion to the signal. This parameter is only available in Modern mode.

Glide
Applies glide between the pattern steps, causing values to change more smoothly.

Output
Sets the output level.

RELATED LINKS
Creating Variations for StepFilter Patterns on page 76
Controlling StepFilter via MIDI on page 76
Creating Variations for StepFilter Patterns

You can copy a pattern of the StepFilter to another pattern slot. This 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.

Controlling StepFilter via MIDI

StepFilter allows you to trigger steps individually via MIDI notes.

PREREQUISITE
- Your project contains a MIDI track that is routed to the MIDI input of a track that uses StepFilter as an insert plug-in.
- In StepFilter, **Sync** is deactivated.

PROCEDURE
- Do one of the following:
  - Use the note C0 to increase the step number by one.
  - Use the notes from C1 to D#2 to trigger the steps 1 to 16 directly.

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.

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

Tone
Sets the center filter frequency.

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 10

WahWah

WahWah 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.

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.

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Included Effect Plug-ins
Filter Plug-ins

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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. For a description of how to set up side-chain routing, see the Operation Manual.

MIDI Control

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

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

UV22HR is an advanced version of Apogee's renowned UV22 dithering algorithm, capable of dithering to 8, 16, 20, or 24 bits.

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8, 16, 20, 24 bit

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 auto-pan effect provides several parameters to modulate the left/right stereo position. You can use presets or create individual curves for the modulation waveform. *AutoPan* also allows for chopping effects by linking the modulation of left and right channel.

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**NOTE**

The panning effect of this plug-in works only on stereo tracks.

Waveform display

Shows the shape of the modulation waveform and allows you to manually adjust the shape of the waveform. To draw an individual curve, click a node and move the mouse. To draw a straight line, **Shift**-click a node and move the mouse.

Waveform presets

Allows you to select presets for the modulation waveform.

- **Sine** creates a smooth sweep.
- **Triangle** creates a ramp, that is, a linear movement from full right to full left and back.
• **Pulse** creates an instant jump to full right, then to full left, and then back to center.

• **Random One Shot** creates a random curve. Click this button again to create a new random curve.

• **Random Continuous** automatically creates a new random curve after each period.

**Phase**

Sets the offset for the starting point of the curve. If multiple AutoPan plug-ins are used on different tracks, for example, different offset settings for each track allow for a more organic overall sound.

**Factor**

If Sync is activated, this parameter multiplies the sync rate by the selected factor. This allows you to create very slow movements in panorama.

**Rate**

Sets the auto-pan speed and shows the movement within the panorama. If Sync is deactivated, the speed is set in Hertz. If Sync is activated, you can set the speed in tempo values.

**Sync**

Activates/Deactivates tempo sync.

**Link**

If this button is activated, left and right channel are modulated simultaneously. This results in a chopping effect instead of auto-panning.

In this mode, **Width** sets the intensity of the volume modulation.

**Width**

Sets the amount of deflection to the left and right side of the stereo panorama. If Link is activated, this parameter sets the intensity of the volume modulation.

**Smooth**

Allows you to smooth the transition between individual steps of the panorama curve.

**Chopper**

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

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Waveform buttons
Allow you to select the modulation waveform.

Depth
Sets the intensity 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 option is activated, Chopper acts as a tremolo effect only. If this option 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 signal and the wet signal. If the effect is used as a send effect, set this parameter to the maximum value as you can control the dry/effect balance with the send.

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.

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Delay
Affects the frequency range of the modulation sweep by adjusting the initial delay time.

Width
Sets 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 parameter to the maximum value as you can control the dry/effect balance with the send.

Rate
If Sync is activated, Rate allows you to specify the base note value for synchronizing the modulation sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).
If Sync is deactivated, the sweep rate can be set freely with the Rate knob.

Sync
Activates/Deactivates tempo sync.

Waveform Shape
Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.

Lo Filter/Hi Filter
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.
Cloner

**Cloner** adds up to 4 detuned and delayed voices to the signal, for rich modulation and chorus effects.

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**Graphical display**

Shows the panning position of each voice in the stereo spectrum.

**Voices**

Sets the number of voices. Each voice has a **Detune** and a **Delay** slider.

**Detune slider (1 to 4)**

Sets the relative amount of detune for each voice. Positive and negative values can be set. If set to zero, no detune takes place for that voice.

**Delay slider (1 to 4)**

Sets the relative delay amount for each voice. If set to zero, no delay takes place for that voice.

**Detune**

Sets the overall amount of detune for all voices. If set to zero, no detune takes place regardless of the **Detune** slider settings.

**Natural**

Changes the pitch algorithm that is used for detune.
**Humanize (Detune)**
Sets the amount of detune variation if **Static Detune** is deactivated. With **Humanize**, the detune is continuously modulated for a more natural effect.

**Static (Detune)**
Activate this button to use a static amount of detune.

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

**Spatial**
Spreads the voices across the stereo spectrum. Turn the control clockwise for a deeper stereo effect.

**Output (-12 to 12 dB)**
Sets the output gain.

**Delay**
Govern the overall depth of the delay for all voices. If set to zero, no delay takes place regardless of the **Delay** slider settings.

**Humanize (Delay)**
Controls the amount of delay variation if **Static Delay** is deactivated. With **Humanize**, the delay is continuously modulated for a more natural effect.

**Static (Delay)**
Activate this button to use a static delay amount.

**Flanger**
This is a classic flanger effect with added stereo enhancement.

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**Delay**
Affects the frequency range of the modulation sweep by adjusting the initial delay time.
**Feedback**
Determined the character of the flanger effect. Higher settings produce a more metallic sounding sweep.

**Mode**
Allows you to toggle between **LFO** and **Manual** mode.

*In LFO mode, you can define the sweep rate or sync it to the project tempo. In Manual mode, you can change the sweep manually.*

**Rate**
If **Sync** is activated, **Rate** allows you to specify the base note value for synchronizing the flanger sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

*If **Sync** is deactivated, the sweep rate can be set freely with the **Rate** knob.*

**Sync**
Activates/Deactivates tempo sync.

**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 parameter to the maximum value as you can control the dry/effect balance with the send.

**Lo Range/Hi Range**
Set the frequency boundaries for the flanger sweep.

**Waveform Shape**
Allows you to select the modulation waveform, altering the character of the flanger sweep. A sine and a triangle waveform are available.

**Lo Filter/Hi Filter**
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**.

---

**Metalizer**

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

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<tr>
<th>Included Effect Plug-ins</th>
<th>Modulation Plug-ins</th>
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<td><strong>Metalizer</strong></td>
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Mix
Sets the level balance between the dry signal and the wet signal. If the effect is used as a send effect, set this parameter to the maximum value as you can control the dry/effect balance with the send.

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

Speed
If tempo sync is activated, this is where you specify the base note value for synchronizing the effect to the tempo of the host application (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.

Mono
Switches the output to mono.

Feedback
Determines the character of the metal effect. Higher settings produce a more metallic 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.

Output
Sets the output level.
**Phaser**

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

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**Feedback**

Determines the character of the phaser effect. Higher settings produce a more pronounced effect.

**Width**

Sets the intensity of the modulation effect between higher and lower frequencies.

**Mode**

Allows you to toggle between **LFO** and **Manual** mode.

In **LFO** mode, you can define the sweep rate or sync it to the project tempo. In **Manual** mode, you can change the sweep manually.

**Rate**

If **Sync** is activated, **Rate** allows you specify the base note value for tempo-syncing the phaser sweep (1/1 to 1/32, straight, triplet, or dotted).

If **Sync** is deactivated, the sweep rate can be set freely with the **Rate** knob.

**Sync**

Activates/Deactivates tempo sync.

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

**Lo Filter/Hi Filter**

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.

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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

Waveform buttons

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

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 that 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.

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 with harmonically rich waveforms (square or saw, for example).

Range
Determines the frequency range of the oscillator in Hz.

LFO

Waveform buttons
Allow you to select the LFO waveform: square, sine, saw, or triangle.

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.

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

Envelope Generator
The envelope generator parameters control 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 R>L
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.

Level Settings

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

Output
Sets the output level.

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

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

**Speed settings**

**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.

**Speed selector (stop/slow/fast)**

Allows you to control the speed of the rotary speaker.

**Speed Mod**

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

**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.

**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
Sets the output level.

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

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

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### Delay
Affects the frequency range of the modulation sweep by adjusting the initial delay time.

### Width
Sets 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 parameter to the maximum value as you can control the dry/effect balance with the send.

### Rate
If Sync is activated, Rate allows you to specify the base note value for synchronizing the modulation sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If Sync is deactivated, the sweep rate can be set freely with the Rate knob.

### Sync
Activates/Deactivates tempo sync.
Waveform Shape
Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.

Lo Filter/Hi Filter
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
Tranceformer is a ring modulator effect that modulates incoming audio by an internal, variable frequency oscillator, producing new harmonics. You can use a second oscillator to modulate the frequency of the first oscillator, in sync with the song tempo if needed.

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

Waveform buttons
Allow you to select a pitch modulation waveform.

Waveform display
Allows you to modify the Pitch and Depth parameters simultaneously by dragging.

Pitch
Sets the frequency of the modulating oscillator.

Activate/Deactivate Pitch Modulation
Activates/Deactivates the modulation of the pitch parameter.
Speed
If tempo sync is activated, this is where you specify the base note value for synchronizing the effect to the tempo of the host application (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.

Depth
Sets the intensity of the pitch modulation.

Mono
Switches the output to mono.

Output
Sets the output level.

Tremolo
Tremolo produces amplitude modulation.

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Rate
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 Sync is deactivated, 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
Sets the output level.

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

**Vibrato** creates pitch modulation.

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Depth
Sets the intensity of the pitch modulation.

Rate
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 **Sync** is deactivated, the modulation speed can be set freely with the **Rate** knob.

Sync
Activates/Deactivates tempo sync.

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.
Network Plug-ins

VST Connect CUE Mix

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

VST Connect SE

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Other Plug-ins

LoopMash FX

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

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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.

**Randomizer**

**Randomizer** allows you to create variations of a sound. By setting specific parameter values, you can determine how much these parameters are randomized. This is helpful if you want to use an audio file, for example, the sound of a closing door, several times in your project and let it sound different every time.

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NOTE

This plug-in is recommended for use in offline processing. Using it as a realtime plug-in might increase the latency.

**Randomizer** allows you to define the range within its parameters are allowed to change. **Pitch**, **Impact**, **Color**, and **Timing** act as metaparameters. This means that when you modify one of these metaparameters, a whole set of parameters is modified in the background.

**Pitch**
- Sets the maximum pitch change.
- This parameter allows for very basic and effective sound variations. It is especially suitable for voices and tonal sounds but can also give good results on atonal noises.

**Impact**
- Sets the maximum variations of attack and envelope settings.
- This parameter allows for variations of the onsets and transitions of sounds. Even sounds without hard attacks may get interesting variations.

**Color**
- Sets the maximum amount of coloration.
- This parameter allows you to change the tonal balance of the fundamental frequency and its harmonics. This can give good results on rich sounds, for example, squeaks and effects but also on voices.

**Timing**
- Sets the maximum timing change.
- This parameter allows you to vary the timing of sounds that have a recognizable pattern. At extreme settings, this may change the order of segments within a sound.

NOTE

Setting the above parameters to a value of 5 causes a noticeable yet natural sounding variation. Values higher than 8 alter the sound drastically.

RELATED LINKS

Creating Variations of Audio Events on page 100
Creating Variations of Audio Events

**Randomizer** allows you to create several variations of audio events in one go.

**PREREQUISITE**
In the **Direct Offline Processing** window, **Auto Apply** is activated.

**NOTE**
For detailed information about **Direct Offline Processing**, see the **Operation Manual**.

**PROCEDURE**
1. Create as many copies of the event as you need.
2. Select all event copies.
3. Select **Audio > Plug-ins > Other > Randomizer**.
4. In the dialog, select **New Version**.
   **NOTE**
   You can also make this a permanent setting in the **Preferences** dialog (**Editing-Audio** page).
5. In the **Direct Offline Processing** window, set the values for **Pitch**, **Impact**, **Color**, and **Timing**.
   The selected events are modified on a random basis within the defined parameter values.

**RESULT**
You have created different sounding variations of your audio event.

**AFTER COMPLETING THIS TASK**
Create audio assets by using the **Export Selected Events** dialog and transfer them to a game audio engine, for random playback. In case of Audiokinetic Wwise, use Game Audio Connect to transfer audio assets.

**NOTE**
For detailed information about Game Audio Connect, see the **Operation Manual**.

---

**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.

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Direct

Sets the level balance between the dry signal and the wet signal. A value of 0 means that 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.

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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.
keyboard display, the original audio is displayed in blue while the changes are displayed in orange.

**Correction**

**Speed**
Determine the smoothness of the pitch change. Higher values cause the pitch shift to occur immediately. 100 is a very drastic setting that is designed mainly for special effects.

**Tolerance**
Determine 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. 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
PitchDriver 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.

Detune
Lets you detune the pitch of the incoming audio.

Mix
Sets the level balance between the dry signal 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

Sets the output level.

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 control panel for the card (opened via the **Device Setup** dialog).

Reverb Plug-ins

**REVelation**

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

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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.
Pre-Delay
Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the first reflections to reach the listener.

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

### 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.

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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 or 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 top right of the browser 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 that you want to work with in a session. The selected program slot is indicated by a white frame. Used slots are shown in blue. A red program slot indicates that the impulse response file is missing. 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, but they are represented by different icons.

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

NOTE
Manually imported impulse responses themselves are not part of the program or preset. If you want to move your project to a different computer, you have to move the impulse responses as well.

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 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 109
EQ Settings on page 111
The Pictures Section on page 112
Custom Impulse Responses on page 113
Relocating Content on page 115

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 113

Reverb Settings

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

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**  
Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the first reflections to reach the listener.

**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 split point is set to 60 ms.

**ER Tail Mix**  
Allows you to set up the relation of early reflections and tail. Values above 50 attenuate the early reflections and values below 50 attenuate the tail.

### The Impulse Response Display

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

**Time Scaling**  
This wheel lets you adjust the reverb time.

**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 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, and 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.
Included Effect Plug-ins
Reverb Plug-ins

**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.

![Picture Section](image)

**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.

Out
Adjusts the overall output level.

Mix
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 them 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

REVerence allows you to import your own impulse response files. Before importing these impulse response files, you can preview their effect.

PREREQUISITE
To preview the effect of impulse response files during the import process, do one of the following:

- If you use REVerence as an insert effect, play back the event to which you want to add the effect in a loop.
- If you use REVerence in the Direct Offline Processing window, activate Audition and Audition Loop.
PROCEDURE
1. In the program matrix, click **Import**.
2. In the file dialog that opens, navigate to the location of your impulse response files.
3. Optional: Select an impulse response file to preview it.
4. Select 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 **Audio Connections** window), see below.
5. 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.
6. 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.

<table>
<thead>
<tr>
<th>Number of input channels</th>
<th>Channel order in REVerence</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>L</td>
</tr>
<tr>
<td>2</td>
<td>L/R</td>
</tr>
<tr>
<td>3</td>
<td>L/R/C</td>
</tr>
<tr>
<td>4</td>
<td>L/R/LS/RS (if inserted on a track with a 4.0 channel configuration)</td>
</tr>
<tr>
<td>4</td>
<td>LL/LR/RL/RR (if inserted on a track with a stereo configuration)</td>
</tr>
<tr>
<td>5</td>
<td>L/R/C/LS/RS</td>
</tr>
<tr>
<td>6</td>
<td>L/R/C/LFE/LS/RS (LFE is being ignored.)</td>
</tr>
</tbody>
</table>
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:

<table>
<thead>
<tr>
<th>Channel</th>
<th>The signal from this source...</th>
<th>...was recorded with this microphone</th>
</tr>
</thead>
<tbody>
<tr>
<td>LL</td>
<td>left source</td>
<td>left microphone</td>
</tr>
<tr>
<td>LR</td>
<td>left source</td>
<td>right microphone</td>
</tr>
<tr>
<td>RL</td>
<td>right source</td>
<td>left microphone</td>
</tr>
<tr>
<td>RR</td>
<td>right source</td>
<td>right microphone</td>
</tr>
</tbody>
</table>

**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 because 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 because 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 ambience and reverb effects in stereo and surround formats. The CPU usage is adjustable to fit the needs of any system. From short room reflections to cavern-sized reverb, this plug-in delivers high quality reverberation.

<table>
<thead>
<tr>
<th>Cubase LE</th>
<th>Cubase AI</th>
<th>Cubase Elements</th>
<th>Cubase Artist</th>
<th>Cubase Pro</th>
<th>Nuendo</th>
</tr>
</thead>
<tbody>
<tr>
<td>Included with</td>
<td>–</td>
<td>–</td>
<td>–</td>
<td>–</td>
<td>X</td>
</tr>
</tbody>
</table>

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

Hi Gain
Sets the amount of boost or attenuation for the high-shelving filter.

**Reverb Character**

**Pre-Delay**
Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the first reflections to reach the listener.

**Reverb Time**
Allows you to set the reverb time in seconds.

**Size**
Alters the delay times of the 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. At a setting of 0 %, the output of the reverb is mono; at 100 % it is stereo.

**Variation**
Clicking this button generates a new version of the same reverb program using altered reflection patterns. This is helpful if some 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.

**Hi Freq**
Determines the frequency above which high-frequency damping occurs.

**Lo Level**
Affects the decay time of the 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.

**Hi Level**
Affects the decay time of the 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.
Envelopes

Amount
Determine 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
Sets the level balance between the dry signal and the 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 deactivates 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 smaller version of the RoomWorks plug-in. RoomWorks SE delivers high quality reverberation, but has fewer parameters and is less CPU demanding than the full version.

<table>
<thead>
<tr>
<th>Cubase LE</th>
<th>Cubase AI</th>
<th>Cubase Elements</th>
<th>Cubase Artist</th>
<th>Cubase Pro</th>
<th>Nuendo</th>
</tr>
</thead>
<tbody>
<tr>
<td>Included with</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
</tbody>
</table>
Pre-Delay
Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the 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.

Lo Level
Affects the decay time of the 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.

Hi Level
Affects the decay time of the 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.

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%.

Spatial + Panner Plug-ins

Anymix Pro

<table>
<thead>
<tr>
<th></th>
<th>Cubase LE</th>
<th>Cubase AI</th>
<th>Cubase Elements</th>
<th>Cubase Artist</th>
<th>Cubase Pro</th>
<th>Nuendo</th>
</tr>
</thead>
<tbody>
<tr>
<td>Included with</td>
<td>=</td>
<td>=</td>
<td>=</td>
<td>=</td>
<td>=</td>
<td>X</td>
</tr>
</tbody>
</table>

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:

<table>
<thead>
<tr>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
</tr>
</thead>
<tbody>
<tr>
<td>L</td>
<td>R</td>
<td>C</td>
<td>LFE</td>
<td>LS</td>
<td>RS</td>
<td>RSS/RC</td>
<td>LSS/LC</td>
<td>CS</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
</tr>
</thead>
<tbody>
<tr>
<td>Track configuration: 6.1 Cine</td>
<td>L</td>
<td>R</td>
<td>C</td>
<td>LFE</td>
<td>LS</td>
<td>RS</td>
<td>CS</td>
<td></td>
</tr>
<tr>
<td>Plug-in output configuration: 7.0 Music</td>
<td>L</td>
<td>R</td>
<td>C</td>
<td>LS</td>
<td>RS</td>
<td>LSS</td>
<td>RSS</td>
<td></td>
</tr>
<tr>
<td>Result</td>
<td>OK</td>
<td>Mismatch</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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 lower 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.

![](image)

**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 select one of the following options 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 128

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.

**Proximity**

*Anymix Pro* supports the Proximity surround format. Proximity allows you to take a sound from the speakers and place it very close to the listener.

To make this effect audible, you need an appropriate reproduction device, for example, a Proximity headphone system.
If you use the Proximity format as input or output configuration, you can select one of the following patterns from the **In** and **Out** pop-up menus:

**Variable Input, Proximity Output**
Converts incoming signals to the Proximity format. This allows you to balance the incoming signals between a standard 5.1 setup and two additional Proximity channels.

**Proximity Input, Proximity Output**
Allows you to adjust the FX Level in the Proximity signal.

**Proximity Input, Variable Output**
Decodes the incoming Proximity signal and converts it to any standard output configuration other than 5.1.

**NOTE**
Convert 8.0 and 8.1 tracks to 7.1 before you encode to 7.1 Proximity. This prevents errors in the channel order.

**Proximity FX Level**
Provides the Proximity effect settings.
Maximum

Allows you to set the maximum Proximity effect level that is applied to the input sources.

Distance Control

Allows you to select how the position of the input channels affects the effect level. You can select an attenuation curve from the pop-up menu:

- **Off**
  Provides a constant effect level.

- **Linear**
  The attenuation starts right from the center position and is applied linearly.

- **Sinusoidal**
  The attenuation starts approximately at loudspeaker distance and increases exponentially with distance.

- **Elliptical**
  The attenuation starts at two thirds of the stage with an exponential attenuation curve.

Reference

Allows you to select if the distance-based attenuation of the Proximity level depends on the position of the center icon or on the position of each input channel.

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

**MonoToStereo** turns a mono signal into a pseudo-stereo signal. The plug-in can be used on a mono file or a stereo file with equal channels.

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**NOTE**

This plug-in works only on stereo tracks.

**Width**

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

**Delay**

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

**Color**

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

**Mono**

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

**StereoEnhancer** expands the stereo width of (stereo) audio material. It cannot be used with mono files.

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**NOTE**

This plug-in works only on stereo tracks.
**Delay**
Increases the amount of differences between the left and right channels to further increase the stereo effect.

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

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

**VST AmbiDecoder**

VST AmbiDecoder allows you to convert Ambisonics audio for playback on headphones or multi-channel speaker setups.

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For a description of VST AmbiDecoder, see the Operation Manual.

**VST MultiPanner**

VST MultiPanner is a surround panner that allows you to position a sound source in the surround field or to modify existing premixes.

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VST MultiPanner is only available as a channel panner, but not as an insert effect.

For a description of VST MultiPanner, see the Operation Manual.
Surround Plug-ins

Bass Manager

If your main speakers are small, Bass Manager 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.

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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. 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 10 dB.

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 10 dB.

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

MatrixEncoder

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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. MatrixEncoder takes four separate inputs (LRCS = Left, Right, Center, and Surround) and creates two final outputs: Left-total and Right-total (Lt and Rt).

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

RELATED LINKS
MatrixDecoder on page 136

Setting Up

PROCEDURE
1. In the Audio 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 135

Using the MatrixEncoder/MatrixDecoder

PROCEDURE
1. Set up the mix roughly the way you want it.
   Use the VST MultiPanner 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 indicates the dominant direction of the mix, sometimes referred to as the dominance vector. Part of the processing that is applied 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 MatrixEncoder and MatrixDecoder.

**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, because 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 Audio 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.
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.

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**NOTE**

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

**NOTE**

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

**RELATED LINKS**

MatrixEncoder on page 133

Mix6to2

Mix6to2 lets you quickly mix down your surround mix format to stereo. You can control the levels of up to six surround channels and decide for each channel up to which level it is included in the resulting mix.

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**Included Effect Plug-ins**

Surround Plug-ins
Surround Channels

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

**Link**
Link the volume faders.

**Invert**
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 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

Mix8to2 lets you quickly mix down your surround mix format to stereo. You can control the levels of up to eight surround channels and decide for each channel up to which level it is included in the resulting mix.

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Surround Channels

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

Link
Link the volume faders.

Invert
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 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.

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For a description of MixConvert V6, see the Operation Manual.
MixerDelay

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

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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 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.

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Included with

There are three different modes:
- Oscilloscope (Ampl)
- Phase Correlator (Scope)
- Frequency Spectrum Analyzer (Freq)

Oscilloscope Mode (Ampl)

If Ampl is activated, the display shows signal waveform.

Channel menu

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.

Freeze

Freezes the display.

Mode A/Mode B

This option is not available for this mode.
Amplitude
Allows you to increase/decrease the vertical size of the waveform.

Frequency
Allows you to select the frequency area for viewing.

**Frequency Spectrum Analyzer Mode (Freq)**

If **Freq** is activated, **MultiScope** 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.

**Channel menu**
- 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.

**Freeze**
Freezes the display.

**Mode A/Mode B**
- Allow you 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**.

**Amplitude**
Allows you to increase/decrease the vertical range of the bands.

**Frequency**
Allows you to divide the frequency spectrum into 8, 15, or 31 bands, or you set it to **Spectrum**, which gives you a high-resolution view.
Phase Correlator Mode (Scope)

If Scope is activated, 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, the Channel menu 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.

Freeze

Freezes the display.

Mode A/Mode B

If the channel mode Surround is selected, these buttons allow you to switch between different view modes.

Amplitude

Allows you to increase/decrease the vertical size of the waveform.

Frequency

Allows you to select the frequency area for viewing.

SMPTEGenerator

SMPTEGenerator 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 synchronize directly to SMPTE timecode). This can be very useful if you do not have access to a MIDI-to-timecode converter.

### 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 corresponds 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, **SMPTEGenerator**, 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 **SMPTEGenerator** 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 **SMPTEGenerator**.

5. Activate the **Link to Transport** button.
   The **SMPTEGenerator** 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.

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<th>Included Effect Plug-ins</th>
<th>Cubase LE</th>
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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. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

**NOTE**

Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

**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 0 dB.

**Tuner**

This is a guitar tuner.

**IMPORTANT**

Make sure that you deactivate any other effect that alters pitch, like chorus or vibrato.

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Simply connect a guitar or other instrument to an audio input and select **Tuner** as an insert effect.

**IMPORTANT**

To use **Tuner**, you must activate **Monitor** for the track with **Tuner**.

You can display **Tuner** in an analog view or in a digital view.

- To switch between analog view and digital view, position the mouse pointer in the upper left corner of the plug-in, and click the **Switch between analog view and digital view** button.

**Analog View**

When you play a note, the pitch is shown in the middle of the display. 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.

**Digital View (Cubase Pro and Nuendo only)**

**Note**

Displays the currently played pitch.

**Cent**

Displays the value, deviating from **Note**. A negative value indicates that the pitch is flat. A positive value indicates that the pitch is sharp.

**Strobe/Classic**

You can switch the display between two modes: **Strobe** and **Classic**.

In **Strobe** mode, a blue indicator constantly runs from right to left, that is from positive to negative. The more flat a note is, the faster the indicator moves. As soon as you play a sharp tone, the running direction of the indicator runs from left to
right, that is from negative to positive. When you play the correct pitch, the indicator stops and the display turns gray.

When you play a note in Classic mode, the pitch is shown in the middle of the display. When you play the correct pitch, the middle of the indicator turns gray.

**Base**

Displays the frequency of the base note A. Its default value is 440 Hz. You can adjust Base by ± 15 Hz.

**Octave**

Displays the octave.
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.

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**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. For example, setting this to **16** means the arpeggio is a pattern of 16 notes.

**Length**

Sets the length of the arpeggio notes, as a note value related to the project tempo.

- To create staccato arpeggios, set a smaller value for **Length** than for **Step Size**.
To create arpeggio notes that overlap each other, set a greater value for Length than for Step Size.

**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.

For example, 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 the arpeggiator as an insert effect for the track.
3. Activate the arpeggiator.
4. On the arpeggiator panel, use the **Step Size** setting to set the arpeggio speed.
5. Use the **Length** setting to set the length of the arpeggio notes.
6. Set the **Key Range** parameter to 12.
   This makes the notes arpeggiate within an octave.
7. 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.
8. 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.

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**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.

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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.

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 Beat Designer, you can quickly and easily set up the drums for a project, by experimenting and creating new drum sequences from scratch.

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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

1. Step display
2. Jump mode
3. Swing and Offset controls
4. Flam position settings
5. Pattern display
6. Swing settings
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 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 159
Swing Setting on page 157

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. 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. 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,
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 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, 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.
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, or into another instance of 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 158

Swing Setting

This parameter can be used to create a swing or shuffle rhythm. This adds a more human feel to drum patterns that might otherwise be too static.

Swing offsets 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. You can set up two swing settings with these sliders and then quickly switch between these during playback.

- To delay every second or third drum step in the pattern, drag a slider to the right.
- To make a drum step play earlier in the pattern, drag a slider to the left.
- To switch between the swing settings, click the Swing buttons to the right of the step display.

- To deactivate swing for a lane, click on the selected Swing buttons.

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.

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.

Adding Flams

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, make settings for the flams that you created.
   ● To add the flams before or after the drum step, drag a position slider to the left or right.
     If you add flams before the first drum step in a pattern, this is indicated in the display by a small arrow in the top left corner of this step. Starting playback at the normal pattern start would result in these flams not being played.
   ● To set the velocity for the flams, use the vertical sliders to the right of the flam sliders.
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 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 Beat Designer.

   ![MIDI track example]

   ● 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 **Beat Designer**, to avoid doubling of the drums. **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 **Beat Designer** again.

**RELATED LINKS**

*Pattern Functions Menu* on page 156

**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. Activate **Jump**.

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 instantly (at the moment the event is received) or at the next bar: Activate **Now** 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 instantly, 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.

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<thead>
<tr>
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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.

![Chord Indicator Lane](image)

**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 162

**Entering Chords**

**PROCEDURE**

1. Activate 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](image)

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. On the **Layers** pop-up menu, 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 163

### 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.

The following applies:
- 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.

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Threshold

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

Ratio

Sets the amount of compression applied to the velocity values above the set threshold. Ratios greater than 1:1 result in compression. Ratios lower than 1:1 result in expansion.

Gain

Adds or subtracts a fixed value from the velocities. Since the maximum range for velocity values is 0 to 127, you can use the Gain setting to compensate, keeping the resulting velocities within the range. Typically, negative Gain settings are used for expansion and positive settings for compression.

Context Gate

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

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This effect features two modes: In Poly Mode, Context Gate recognizes certain chords that are played. In Mono Mode, only certain MIDI notes are let through.
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 of the specified MIDI channel are let through. This can be used with MIDI controllers that can send MIDI on 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. Notes are played back until another note within the set range is played.

- 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, you can specify the time, after which note-off messages are sent for the notes that are playing.

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, deactivate the Learn Reset button.

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 **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 your guitar without having to use an additional MIDI instrument.

### Density

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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

This generic control panel allows you to select up to 8 different MIDI controller types and set values for these. You can then use the plug-in as a control panel to adjust the sound of a MIDI instrument from within your host application.

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- To select a controller type, use the pop-up menus on the right.
To change the value of a controller type, enter a value in the value field or click the value field and drag the cursor up or down.

To deactivate a controller, type Off in the value field or click the value field and drag the cursor down until the value field displays Off.

**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.

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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 according to this value. 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.

<table>
<thead>
<tr>
<th>Note Value</th>
<th>Ticks</th>
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<tr>
<td>1/32 note</td>
<td>60</td>
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<tr>
<td>1/16 note triplet</td>
<td>90</td>
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<tr>
<td>1/16 note</td>
<td>120</td>
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<tr>
<td>1/8 note triplet</td>
<td>160</td>
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<tr>
<td>1/8 note</td>
<td>240</td>
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<tr>
<td>Quarter note triplet</td>
<td>320</td>
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<tr>
<td>Quarter note</td>
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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.

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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.

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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

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**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 set the root note that is taken as a reference for the detuning.

• You can adjust all keys by the same amount by keeping Alt pressed.

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

### Note to CC

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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.

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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**

Sets the delay time 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).

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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 2 bar pattern with sixteenth note steps.
4. Click in the note display to insert notes. You can insert notes on any of the 32 steps, but 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.
   - To remove a note from the pattern, click on it again.
Adding Controller Curves

PROCEDURE
1. Open the Controller pop-up menu and select a controller.
   The selection is displayed in the lower controller display.
2. Click in the controller display to draw events.
   The MIDI controller events are sent out during playback along with the notes.

   ![Controller display]

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

Setting Up the Controller Menu

You can specify which two controller types (filter cutoff, resonance, volume, etc.) should be available on the Controller pop-up menu.

PROCEDURE
1. Click Setup.
2. Select the controllers that you want to have available in the Controller pop-up menu and click OK.
   This selection is global, that is, it applies to all patterns.

Adjusting the Step Length

- To make notes shorter, select Gate on the Controller pop-up menu and lower the bars in the controller display.
  If a bar is set to its maximum value, the corresponding note is the full length of the step.
- To make notes longer, you can tie two notes together. This is done by inserting two notes and clicking in the Tie column for the second note.
  If 2 notes are tied, the second note is not triggered – the previous note is lengthened instead. Also, the second note gets the same pitch as the first note. You can add more notes and tie them in the same way, creating longer notes.
Other Pattern Functions

**Shift Octave up/down**
Shifts the entire pattern up or down in octave steps.

**Shift Steps left/right**
Moves the pattern one step to the left or right.

**Reverse**
Reverses the pattern, so that it plays backwards.

**Copy/Paste**
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 resetting controller values.

**Randomize**
Generates a completely random pattern.

**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 StepDesigner.

Automating Pattern Changes

You can create up to 200 different patterns in each StepDesigner instance.

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 the MIDI track.

Note that you can also press a key on your MIDI keyboard to change patterns. For this, you have to set up 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. 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 first 92 patterns.

Track Control

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, Track Control allows you to adjust sounds and effects in your instrument from within your host application.

The Available Control Panels

You select the control panel from the pop-up menu at the top of the effect panel. The following panels are available:

**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

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 makes the attack time longer.

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

**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 makes the attack time longer.

Release
Adjusts the release time of the sound. Lowering this value shortens the release, while raising it makes the release time longer.

Harm.Cont
Adjusts the harmonic content of the sound.

Bright
Adjusts the brightness of the sound.

CutOff
Adjusts the filter cutoff frequency.

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

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

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.

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<th>Cubase AI</th>
<th>Cubase Elements</th>
<th>Cubase Artist</th>
<th>Cubase Pro</th>
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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 Transformer. Where there are differences between the two, this is clearly stated.
This chapter contains descriptions of the included VST instruments and their parameters.

### Groove Agent SE

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This VST instrument is described in detail in the separate document Groove Agent SE.

### HALion Sonic SE

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### LoopMash

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**LoopMash** is a powerful tool for 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.

**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 SE. This allows you to extract certain sounds that you like from **LoopMash** and use them with Groove Agent SE.

The **LoopMash** window has two main areas: the track section in the upper part of the panel, and the parameter section at the bottom.

The selected track is indicated by the background color of the track 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.

On the left of each track, you find the similarity gain sliders. The further to the right you move the similarity gain slider of a track, the more slices are played back from this track.

**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.

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.

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.

RELATED LINKS
LoopMash Parameters on page 183
Applying Slice Selection Modifiers and Slice Effects on page 187

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 the LoopMash 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 Loop Browser), Project window, Pool, Sample Editor (regions), Audio Part Editor, or the File Explorer/macOS 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.
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 188

### 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 189

### 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 191

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 190

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.

![](image)

3. Click **OK**.

RESULT

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

![](image)

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 188

**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.

RELATED LINKS
Slice Selection on page 190
Performance Controls on page 192

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
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.

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 189

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 185

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.
**Mystic**

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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.

**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 187
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.

**RELATED LINKS**

- Mystic Diagram on page 233

## Sound Parameters

### The Impulse Control Section

![Impulse Control Section](image)

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

![Spectrum Displays](image)

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 **EFX** 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 **ENV** 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 **EFX** page offers three separate effect types: Distortion, Delay, and Modulation.

**RELATED LINKS**

- [LFO Page](#) on page 197
- [Envelope Page](#) on page 200
- [Event Page](#) on page 202
- [Effects (EFX) Page](#) on page 203

**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.

![LFO Page Screenshot](#)

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 synchronized to the sequencer tempo.

**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 198

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 selects it and displays the corresponding envelope parameters to the right.
- 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 for the envelope 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.

Included VST Instruments

Mystic
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.

![Event Page](image)

**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 synchronized 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 program of the selected part plays back with the sample rate set in the host application.
- If the **1/2** button is active, the program of the selected part plays back with half the original sample rate.
- If the **1/4** button is active, the program of the selected part 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

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

### Prologue

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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 Prologue.

RELATED LINKS
Prologue Diagram on page 233

Sound Parameters

Oscillator Section

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

Selecting Waveforms

Each oscillator has a number of waveforms that can be selected 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 emphasize 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 produce 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 “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.

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.
However, Osc 2 can be both carrier and modulator as if frequency modulation 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 inharmonious, complex timbres.
- Deactivate oscillator sync when using ring modulation.

**Noise Generator**

A noise generator can be used for simulating drum sounds and breath sounds for wind instruments, for example.

- 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 200

**Filter Section**

The circle in the middle contains the filter parameters. The central control sets the filter cutoff 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 0 dB 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 for the filter types 12 dB LP or 12 dB HP.

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 the filter type using the buttons around the filter cutoff knob. The following filter types are available (listed clockwise starting from the 9 o'clock position):

12 dB LP
Low-pass filters let low frequencies pass and cut out the high frequencies. This low-pass filter has a gentler slope (12 dB/octave above the cutoff frequency), leaving more of the harmonics in the filtered sound.

18 dB LP
This low-pass filter also has a cascade design, attenuating frequencies above the cutoff frequency with a 18 dB/octave slope, as used in the classic TB 303 synth.

24 dB LP
This filter type attenuates frequencies above the cutoff frequency with a 24 dB/octave slope that produces a warm and fat sound.

24 dB LP II
This low-pass filter has a cascade design that attenuates frequencies above the cutoff frequency with a 24 dB/octave slope, which produces a warm and dark sound.
12 dB Band
This band-pass filter cuts both high and low frequencies above and below the cutoff frequency with a 12 dB/octave slope, producing a nasal and thin sound.

12 dB Notch
This notch filter cuts off frequencies near the cutoff frequency by 12 dB/octave, letting the frequencies below and above through. This produces a phaser-like sound.

12 dB HP
A high-pass filter cuts out the lower frequencies and lets the high frequencies pass. This high-pass filter has a 12 dB/octave slope, producing a bright and thin sound.

24 dB HP
This filter has a 24 dB/octave slope, producing 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 EFX 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 ENV 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 EFX page offers three separate effect types: Distortion, Delay, and Modulation.

RELATED LINKS
LFO Page on page 197
Envelope Page on page 200
Event Page on page 202
Effects (EFX) Page on page 203

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 synchronized to the sequencer tempo.

**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 198

**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 selects it and displays the corresponding envelope parameters to the right.
- 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 for the envelope 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 synchronized 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 program of the selected part plays back with the sample rate set in the host application.
- If the 1/2 button is active, the program of the selected part plays back with half the original sample rate.
- If the 1/4 button is active, the program of the selected part 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

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<th>Cubase AI</th>
<th>Cubase Elements</th>
<th>Cubase Artist</th>
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This VST instrument is described in detail in the separate document *Retrologue*.

Spector

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The synthesis used by Spector is based around a spectrum filter. This 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.
- A Morph control lets you mix the output of spectrum filters A and B. Since this can be controlled with envelopes, LFOs, etc. That allows you to create morphing effects.
- Controllers and modulation parameters are also available.

RELATED LINKS
Spector Diagram on page 234
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 best 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 central 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
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 button (link symbol) between the cut controls is activated, the two knobs are linked 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 EFX 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 ENV 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 EFX page offers three separate effect types: Distortion, Delay, and Modulation.
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 synchronized to the sequencer tempo.

**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 198

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**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 selects it and displays the corresponding envelope parameters to the right.
- 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 for the envelope 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 synchronized 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 program of the selected part plays back with the sample rate set in the host application.
- If the **1/2** button is active, the program of the selected part plays back with half the original sample rate.
- If the **1/4** button is active, the program of the selected part 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.
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Prologue Diagram
Spector Diagram

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