# Plug-in Reference


# CUBASE ARTIST 10.5

Music Production System

steinberg

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



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# Table of Contents

# Included Effect Plug-ins

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

## Delay Plug-ins

### MonoDelay

This is a mono delay effect. The delay line uses tempo-based or freely specified delay time settings.


**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**

Sets the delay time in milliseconds.

**Sync**

Activates/Deactivates tempo sync.

**Feedback**

Sets the amount of the signal that is sent back into the delay input. The higher this value, the higher the number of repeats.

**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 level.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the delay from another signal source. 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**.

# MultiTap Delay

**MultiTap Delay** is a versatile tap effect delay with up to 8 tap repeats that allows you to create and edit taps manually, to create taps by clicking a rhythm, or to create random taps. You can set up separate effect chains for the delay loop, for delay taps, and for the overall delay output, with each chain containing up to 6 different effects.

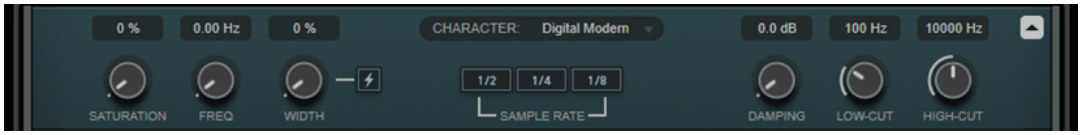
The plug-in offers predefined sound characters that you can freely customize. The delay line uses either tempo-based or freely specified delay time settings. The integrated ducker attenuates the delay output depending on the input signal level, which keeps the delayed signal rather dry during loud or intensely played passages.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.


## Delay Character section

This section allows you to shape the overall sound of the delay. You can choose between 4 predefined sound characters and adjust their parameters to your liking.




**Show/Hide Delay Character Section**

Shows/Hides this section.

**Character pop-up menu**

Allows you to choose the general delay character. **Digital Modern**, **Digital Vintage**, **Tape**, and **Crazy** character are available. Modifying parameters in this section sets the delay character to **Custom**.

**Saturation**

Sets the amount of saturation. This effect is inserted into the delay loop, which means that the saturation of the signal is increased with each repeat.

**Freq**

Sets the frequency of the delay modulation.

**Width**

Sets the intensity of the delay modulation. If this value is set to 0, the signal is not modulated. Activating **Extreme Modulation** increases the modulation to such an extent that drastic variations in tape speed become audible.

**Sample Rate**

Sets the ratio to which the audio sample is decimated. Lower sample rates reduce the high frequency content and sound quality. If no button is activated, no downsampling occurs.

**Damping**

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

**Low-Cut**

Sets the frequency below which low-frequency damping occurs.

**High-Cut**

Sets the frequency above which high-frequency damping occurs.

## Main section

This section contains the general delay parameters and settings, and allows you to add and edit the delay taps.

**Delay**

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

**Sync**

Activates/Deactivates tempo sync.

**Lock/Unlock Delay Value and Number of Taps**

Locks/Unlocks the values of the **Delay** and the **Taps** parameters when loading presets.

**Erase Delay Line**

Erases the delay line.

**Feedback**

Sets the amount of the signal that is sent back into the delay input. The higher this value, the higher the number of repeats.

**Taps display**

Allows you to move taps by dragging their handles and to delete them by double-clicking.

- The **Level** tab allows you to set the level of all taps individually.
- The **Panorama** tab allows you to pan the taps individually in the stereo panorama.
- The **Tap Parameters** tab allows you to adjust the parameters of the selected effect module in the **Tap Effects** section. The pop-up menus provide direct access to the effect modules and their parameters.
- **Spread Taps Evenly for Editing** allows easier editing of taps if they are located very close to each other.

**Grid**

Sets the quantize grid. Taps are quantized to the grid when added or moved.

NOTE

If you hold **Shift**, you can move taps freely even if a grid is set.

**Activate/Deactivate Tapping Mode**

Activates/Deactivates tapping mode so that you can click the **Tap Rhythm** button to create taps.

**Tap Rhythm**

Allows you to create taps by clicking a rhythm with the left mouse button.

**Quantize**

Quantizes all taps to the grid.

**Randomize**

Sets a random number of taps and tap parameter settings. The **Random Taps Options** pop-up menu allows you to specify the minimum and maximum number of taps, the timing, the panning range, and the level range of the random function.

**Taps**

Sets the number of taps.

**Link/Unlink Taps**

Allows you to move all taps simultaneously in the taps display.

**Reset Taps**

Resets the number of taps and all tap parameters to the default.

**Output meter**

Shows the level of the output signal.

**Output**

Adjusts the overall output level.

**Ducker**

This effect attenuates the delay output depending on the input signal level. If the level of the input signal is high, the effect signal is lowered, or ducked. If the level of the input signal is low, the effect signal is raised.

- **FB** suppresses feedback when the delay signal is ducked.
- **DL** erases the delay line one time as soon as the ducking of the delay signal starts.
- **Amount** sets the amount of level reduction that is applied to the delay output. The meter to the right shows the current amount of gain reduction.
- **Release** sets the time after which the effect signal returns to the original level.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the ducking effect from another signal source. If the side-chain signal exceeds the threshold, the delay repeats are ducked. 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**.

**Play Sample Sound**

Plays back a sound sample which allows you to quickly valuate the result of the current plug-in parameter settings.

**Spatial**

Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced 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 level.

**Lock/Unlock Mix Value**

Locks the value of the **Mix** parameter in the main section when loading presets.

## Effects sections

These sections allow you to add, edit, and remove loop, tap, and post effects. You can create effect chains by adding up to 6 different effect modules from 14 effects available overall. You can modify the order of the effects in the chain by dragging the modules.


- Effects in the **Loop Effects** section feed the output signal back into the delay input, allowing for a continuously increasing effect through the loop effect chain.
- Effects in the **Tap Effects** section process the output signal of each delay tap. You can activate/deactivate each effect and set individual parameters for any single tap.
- Effects in the **Post Effects** section affect the overall output signal of the plug-in.

**Bypass**

Bypasses the effect chain in the corresponding effects section.

**Mix**

Sets the balance between dry signal and wet signal for the corresponding effect chain.

**Loop Effects/Tap Effects/Post Effects**

Shows/Hides the corresponding effects section. The section is highlighted if at least one effect module has been added.

**Show/Hide Functional Diagram**

Shows/Hides a functional diagram of the signal flow for **Loop Effects**, **Tap Effects**, or **Post Effects** in the taps display.

**Add Module**

Allows you to add modules to the effect chain of the corresponding section.

**Tap Effects Options**

Opens the options panel for tap effects. **Suspend Effect When Disabled** stops effects from being processed when they are bypassed or deactivated. This reduces the processing load.

IMPORTANT

If this option is activated, crackles may occur when bypassing or activating/deactivating an effect.

**Parameter Link (only available in the Tap Effects section)**

Links the parameters of the same type in all taps. This allows you to edit parameter values of all taps in a module simultaneously. Two link modes are available:

- If **Absolute Mode** is activated and you edit a parameter value of one tap, the corresponding parameter values of the other taps are set to the same value.

- If **Relative Mode** is activated and you edit a parameter value of one tap, the relation of the corresponding parameter values of the other taps remains the same.

**Tap 1-8 (only available in the Tap Effects section)**

Allows you to select a tap for editing the effect parameters.

**Activate/Deactivate Effect (only available in the Tap Effects section)**

Allows you to activate/deactivate the effect for the tap that is selected in the tap display.

RELATED LINKS
Effect Modules on page 10

# Effect Modules

Modules allow you to create an effect chain. Each effect can be used only once in the module chain. You can drag modules in the module chain to rearrange them and change the processing order.

## General Effect Settings

The following settings are available for each module:

**Bypass**

Bypasses the module. This allows you to compare the sound of the unprocessed signal to that of the processed signal.

**Solo**

Solos the module. Only one module can be soloed at a time.

**Remove**

Removes the module from the module chain.

The following effect modules are available:

## Chorus

This is a single-stage chorus effect that doubles the input signal with a slightly detuned version.


**Rate**

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

**Sync**

Activates/Deactivates tempo sync.

**Width**

Sets the depth of the chorus effect. Higher settings produce a more pronounced effect.

**Tone**

Changes the tonal characteristic of the output signal.

**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 level.

## Flanger

This is a classic flanger effect.


**Rate**

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

**Sync**

Activates/Deactivates tempo sync.

**Feedback**

Determines the character of the flanger effect. Higher settings produce a more metallic-sounding sweep.

**Tone**

Changes the tonal characteristic of the output signal.

**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 level.

## Phaser

This is a classic phasing effect.


**Rate**

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

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

**Sync**

Activates/Deactivates tempo sync.

**Width**

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

**Tone**

Changes the tonal characteristic of the output signal.

**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 level.

## Vibrato

This is a pitch modulation effect.


**Rate**

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

If **Tempo Sync** is deactivated, the modulation speed can be set freely with the **Rate** dial.

**Sync**

Activates/Deactivates tempo sync.

**Depth**

Sets the intensity of the pitch modulation.

**Spatial**

Adds a stereo effect to the modulation.

## Envelope Filter

This is a classic envelope filter that allows for auto-wah effects.


**Range**

Determines the frequency range of the filter. **Sweep Downwards** reverses the filter sweep.

**Q-Factor**

Sets the intensity of the envelope filter effect.

**Sensitivity**

Determines how sensitively the effect reacts to the instrument level.

**Attack**

Determines how quickly an effect reacts to the input signal.

**Release**

Sets the gain of the release phase of the signal.

**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 level.

**Type**

Sets the filter type. Low-pass (**LP**), high-pass (**HP**), band-pass (**BP**), and **Notch** filter are available.

## Filter

This is a filter modulation effect. The frequency of the filter is modulated by an LFO or you can set it manually.


**LFO/Manual buttons**

Allows you to select **LFO** and **Manual** mode. In **LFO** mode, you can define the modulation rate or sync it to the project tempo. In **Manual** mode, you can set the frequency manually.

**Freq**

Sets the filter frequency. This parameter is only available in **Manual** mode.

**LFO-Freq**

Sets the filter frequency of the 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, you can set the frequency freely.

This parameter is only available in **LFO** mode.

**Tempo Sync**

Activates/Deactivates tempo sync for the **LFO-Freq** parameter. This parameter is only available in **LFO** mode.

**Q-Factor**

Sets the resonance of the filter.

**Type**

Sets the filter type. Low-pass (**LP**), high-pass (**HP**), band-pass (**BP**), and **Notch** filter are available.

**Low/High**

Set the range of the filter frequency modulation.

## Bit Crusher

This effect uses bit reduction to decimate and truncate the input audio signal to get a noisy, distorted sound.

**Bits (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.

**Sample Div.**

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.

**Mode**

Allows you to select one of the four operating modes. In each mode, the effect sounds differently. Modes **1** and **3** are nastier and noisier, while modes **2** and **4** are more subtle.

## Overdrive

This effect creates a tube-like overdrive effect.


**Drive**

Adds harmonics to the output signal.

**Tone**

Works as a filter effect on the added harmonics.

**Level**

Adjusts the output level.

## Pitch Shifter

This is a pitch-shifting effect.


**Detune**

Changes the pitch of the input signal in semitones.

**Formant**

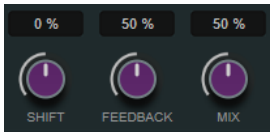
Changes the natural timbre of the input signal.

**Formant Preservation**

Keeps the formants when changing the pitch with the **Detune** control.

## Frequency Shifter

This effect shifts each frequency of the input signal by a fixed amount, which alters the harmonic relations. Adding feedback produces a sound similar to a phaser.



**Shift**

Sets the amount of frequency shift.

**Feedback**

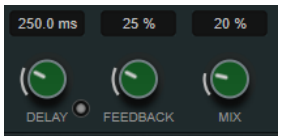
Sets the amount of the signal that is sent from the output of the effect back to its input.

**Mix**

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

## Delay

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




**Delay**

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

**Sync**

Activates/Deactivates tempo sync.

**Feedback**

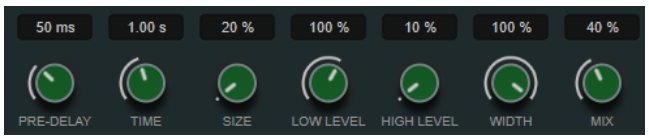
Sets the amount of the signal that is sent back into the delay input. The higher this value, the higher the number of repeats.

**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 level.

## Reverb

This is a versatile reverb for realistic room ambience and reverb effects.




**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**

Allows you to set the reverb time in seconds.

**Size**

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

**Low 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.

**High 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.

**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.

**Mix**

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

## AutoPan

This is an auto-pan effect that allows you to modulate the left/right stereo position.


**Rate**

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

**Sync**

Activates/Deactivates tempo sync.

**Link**

If this button is activated, the 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.

**Waveform Shape**

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.
- **Square** creates an instant jump to full right, then to full left, and then back to center.

## Gate

This effect silences audio signals below a set threshold. As soon as the signal level exceeds the threshold, the gate opens to let the signal through.

**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.

# PingPongDelay

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

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**

Sets the delay time in milliseconds.

**Sync**

Activates/Deactivates tempo sync.

**Feedback**

Sets the amount of the signal that is sent back into the delay input. The higher this value, the higher the number of repeats.

**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 level.

**Spatial**

Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo ping-pong effect.

**Start Left/Start Right**

Determines whether the delay repeat starts on the left or the right channel.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the delay from another signal source. 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.

NOTE

This plug-in works only on stereo tracks.


**Feedback**

Set the number of repeats for each delay.

**Delay**

Sets the delay time 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 level.

**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

This plug-in supports external side-chaining. You can use the side-chain input to control the delay from another signal source. 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.


**Select Amplifier Model**

This pop-up menu allows you to select an amplifier model. You can bypass this section by selecting **No Amp**.

**Drive**

Controls the amount of amp overdrive.

**Bass**

Tone control for the low frequencies.

**Mid**

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.

**Select Cabinet Model**

This pop-up menu allows you to select a speaker cabinet model. You can bypass this section by selecting **No Speaker**.

**Damping Low/High**

These tone controls allow you to shape the sound of the selected speaker cabinet.

# 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.


**Mode**

Allows you to select one of the four operating modes. In each mode, the effect 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.


**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

**Distortion** adds crunch to your tracks.


**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.

**Output**

Sets the output level.

# Distroyer

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


The following parameters influence only the wet signal:

**Lo Filter**

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

**Hi Filter**

Changes the cuttoff 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.


**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.


**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.


## 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 effect 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.

## 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.


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.

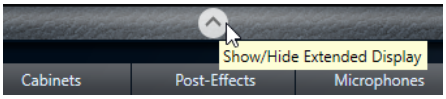
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

#### 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.

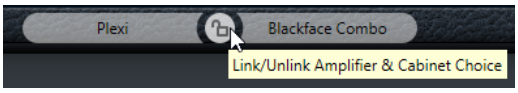



#### Changing the Amplifier and Cabinet Selection in the Compact View

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.

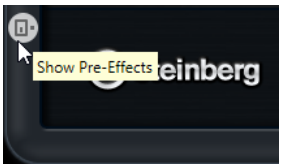
- To lock the amplifier and cabinet combination, activate the **Link/Unlink Amplifier & Cabinet Choice** button. If you now select another amp model, the cabinet selection follows. However, if you select a different cabinet model, the lock is deactivated.




**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.


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 hifi-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.

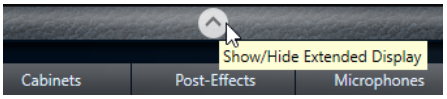
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

**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**

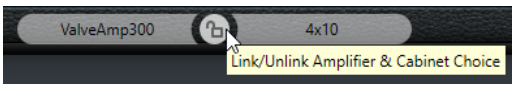
- To toggle between the different views, click the down/up arrow button (**Show/Hide Extended Display**) at the top center of the plug-in frame.




**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.

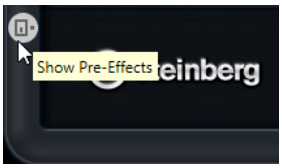
- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the **Link/Unlink Amplifier & Cabinet Choice** button. If you now select another amp model, the cabinet selection follows. However, 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 94

# Dynamics Plug-ins

## Brickwall Limiter

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


Due to its fast attack time, **Brickwall Limiter** can reduce even short audio level peaks without creating audible artifacts. However, this plug-in creates a latency of 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.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

# Compressor

**Compressor** reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both.


**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.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

**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.

**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 **Auto Make-Up Gain** is activated, 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 **Auto Release** 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.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

# DeEsser

**DeEsser** is a special type of compressor that reduces excessive sibilance, primarily for vocal recordings.


You can use it, for example, when close proximity microphone placement and equalizing lead to situations where the overall sound is just right, but where unwanted sibilants occur.

When recording a voice, the position of **DeEsser** in the signal chain is usually after the microphone pre-amp and before a compressor/limiter. This keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics.

## 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

**Side-Chain**

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.

**Freq (25 Hz to 20 kHz)**

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

**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.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

### 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

**EnvelopeShaper** can be used to attenuate or boost the gain of the attack and release phase of audio material.

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**

Sets the gain of the release phase of the signal.

**Output**

Sets the output level.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the effect from another signal source. 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**.

# 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.


**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, the plug-in automatically finds the best release setting for the audio material.

**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.

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

**Q-Factor**

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

**Filter Type (Low-Pass/Band-Pass/High-Pass)**

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

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the gate from another signal source. 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.


**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.


### 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.


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 54

# 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.


**VU Meter**

Shows the amount of gain reduction.

**In/Out Meters**

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

**Input**

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

**Drive (1.0 to 6.0 dB)**

Controls the amount of tube saturation.

**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 **Auto Release** 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.

**Ratio**

Toggles between a low and a high ratio value.

**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 Type (Low-Pass/Band-Pass/High-Pass)**

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

**Center (50 to 20000 Hz)**

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

**Q-Factor**

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

**Monitor**

Allows you to monitor the filtered signal.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

## VintageCompressor

**VintageCompressor** is modeled after vintage type compressors.

This compressor features separate controls for **Input** and **Output** gain, **Attack**, and **Release**. In addition, there is a **Punch** mode which preserves the attack phase of the signal and a program-dependent **Auto Release** function.


**VU Meter**

Shows the amount of gain reduction.

**In/Out Meters**

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

**Input**

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

**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.

**Punch**

If this is activated, the early attack phase of the signal is preserved, retaining the original punch in the audio material, even with short **Attack** settings.

**Release (10 to 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.

**Mix**

Adjusts the mix between dry signal and wet signal, preserving the transients of the input signal.

**Output (-48 to 24 dB)**

Sets the output gain.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

# VSTDynamics

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

The window is divided into three sections containing controls and meters for each effect. Activate the individual effects using the **Gate**, **Compressor**, and **Limiter** buttons. You can select between three different routing options using the **Module Configurator** button.

## Gate

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:

**Input meter**

Shows the level of the input 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.

**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.

**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).

**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, the plug-in automatically finds the best release setting for the audio material.

**Hold (0 to 2000 ms)**

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

**Range**

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

**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 Type (Low-Pass/Band-Pass/High-Pass)**

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

**Center (50 to 20000 Hz)**

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

**Q-Factor**

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

**Monitor**

Allows you to monitor the filtered signal.

## Compressor

**Compressor** reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. It features a separate display that graphically illustrates the compressor curve shaped according to your settings.

**Input meter**

Shows the level of the input signal.

**Graphical display**

Visualizes the settings for **Threshold** and **Ratio** and allows you to adjust them by dragging the handles.

**Gain Reduction meter**

Shows the amount of gain reduction.

**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 **Auto Make-Up Gain** is activated, 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 **Auto Release** is activated, the plug-in automatically finds the best release setting for the audio material.

## Limiter

A limiter ensures that the output level never exceeds a set threshold to avoid clipping in effects following in the chain. Conventional limiters usually require a very accurate setup of the attack and release parameters to prevent the output level from going beyond the set threshold level. **Limiter** adjusts and optimizes these parameters automatically according to the audio material.

**Input meter**

Shows the level of the input signal.

**Gain Reduction meter**

Shows the amount of gain reduction.

**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.

**Output**

Sets the maximum output level.

**Release (10 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 section

**Output meter**

Shows the level of the output signal.

**Module Configurator**

Changes the signal flow through the three effects. Changing the order of the effects can produce different results, and the available routing configurations allow you to quickly compare what works best for a given situation. Click **Module Configurator** to toggle between the following routing configurations:

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

# EQ Plug-ins

## 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.


**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**

Shows the level of the output signal.

# GEQ-10

This is a graphic equalizer that provides 10 frequency bands.


GEQ-10

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**.

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).


## 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.


**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.


**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**

Shows the level of the output signal.

**Morph Factor**

Allows you to mix the output between both filters.

# StepFilter

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


## 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 68
Controlling StepFilter via MIDI on page 68

## 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.


**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 19

# WahWah

**WahWah** is a variable slope band-pass filter that can be auto-controlled by a side-chain signal or via MIDI modeling the well-known analog pedal effect.


You can independently specify the frequency, width, and gain for the **Low** and **High** Pedal positions. The crossover point between the Lo and Hi Pedal positions lies at 50.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the **Pedal** parameter from another signal source. 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**.

## 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 Low/Freq High**

Set the frequency of the filter for the Lo and Hi pedal positions.

**Width Low/Width High**

Set the width (resonance) of the filter for the Lo and Hi pedal positions.

**Gain Low/Gain High**

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.

## 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.


**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.

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 it. 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 preset buttons**

Allow 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.
- **Square** 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, the 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.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the **Width** parameter from another signal source. For a description of how to set up side-chain routing, see the **Operation Manual**.

**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.


**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**

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 level.

# Chorus

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


**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 level.

**Rate**

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

**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

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. 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**.

# Flanger

This is a classic flanger effect with added stereo enhancement.


**Delay**

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

**Feedback**

Determines 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 **Tempo 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 **Tempo Sync** is deactivated, the sweep rate can be set freely with the **Rate** dial.

**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 level.

**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

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. 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.


**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 level.

**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**

Sets the character of the filter effect. The higher the value, the narrower the affected frequency area, which produces a sharper sound and a more pronounced effect.

**Tone**

Sets 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.


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

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

**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 level.

**Lo Filter/Hi Filter**

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

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. 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.


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 LFO affects the oscillator frequency.

**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, for example, square or saw.

**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.


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

**Set 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.

## Mics

**Phase**

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

**Angle**

Sets the simulated microphone angle. A value of 0 ° corresponds to a mono miking setup with a single microphone in front of the speaker cabinet, 180 ° corresponds to a stereo miking setup with a microphone on each side of the cabinet.

**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.

# 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.


**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**

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.


**Rate**

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

If **Tempo Sync** is deactivated, the modulation speed can be set freely with the **Rate** dial.

**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

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. 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.


**Depth**

Sets the intensity of the pitch modulation.

**Rate**

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

If **Tempo Sync** is deactivated, the modulation speed can be set freely with the **Rate** dial.

**Sync**

Activates/Deactivates tempo sync.

**Spatial**

Adds a stereo effect to the modulation.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the **Operation Manual**.

# Other Plug-ins

## LoopMash FX

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


**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.

# 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.


**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.


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

## Correction

**Speed**

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

**Tolerance**

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

**Transpose**

With this parameter you can adjust, or retune, the pitch of the incoming audio in semitone steps. 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**

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.

# Reverb Plug-ins

## REVelation

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


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**

Controls the width of the stereo image. 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 level.

**Lock Mix Value**

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 Rate**

Specifies the frequency of the pitch modulation.

**Modulation Depth**

Sets the intensity of the pitch modulation.

**Modulation Activate**

Activates/Deactivates the chorusing effect.

## 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.


### Input Filters

**Low Frequency**

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.

**High Frequency**

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.

**Low Gain**

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

**High 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.

**Size**

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

**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.

**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

**Low Frequency**

Determines the frequency below which low-frequency damping occurs.

**High Frequency**

Determines the frequency above which high-frequency damping occurs.

**Low 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.

**High 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.

## Envelope

**Amount**

Determines how much the envelope attack and release controls affect the reverb itself. Lower values have a more subtle effect while higher values lead to a more drastic sound.

**Attack**

The envelope settings in **RoomWorks** control how the reverb follows the dynamics of the input signal in a fashion similar to a noise gate or downward expander. Attack determines how long it takes for the reverb to reach full volume after a signal peak (in milliseconds). This is similar to a pre-delay, but the reverb is ramping up instead of starting all at once.

**Release**

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

### Output

**Mix**

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**

Shows 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.

**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.

**Low 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.

**High 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

## 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.

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 creating an artificial stereo image.

**Color**

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

## StereoEnhancer

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

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.

# Tools Plug-ins

## Tuner

This is a guitar tuner.

To tune your instrument, connect it to an audio input, select **Tuner** as an insert effect, and activate **Monitor** for the corresponding track. Click **Mute** if you want to mute the output while tuning your instrument.

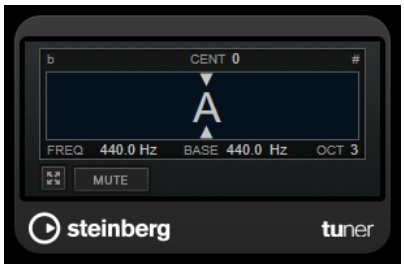
NOTE

Make sure that you deactivate any other effect that alters the pitch, such as a chorus or vibrato.

**Tuner** offers two different viewing modes, analog view and digital view.

- To toggle between analog view and digital view, click the **Toggle between Analog View and Digital View** button.

## Analog View




The graphical display indicates the currently played pitch as a note. The two arrows indicate any pitch deviation. The deviation is also shown in the upper area of the display. If the played note is flat of the pitch, the pitch indicator is located to the left. If the played note is sharp, the pitch indicator is located to the right.

**Cent**

Shows the deviation in pitch. A negative value indicates that the pitch is flat. A positive value indicates that the pitch is sharp.

**Frequency**

Shows the frequency of the played note.

**Base**

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

**Octave**

Shows the octave of the played note.

**Mute**

Mutes/Unmutes the output signal.

## Digital View

This view provides two tuner modes: **Strobe** and **Classic**.

In **Strobe** mode, a colored moving strobe indicates any pitch deviation. If the played note is flat, the strobe moves from right to left. If the played note is sharp, the strobe moves from left to right. The higher the deviation in pitch, the faster the strobe moves. If you play the correct pitch, the strobe stops moving and turns gray.


In **Classic** mode, an indicator shows any pitch deviation. If the played note is flat, the indicator is located left of the middle. If the played note is sharp, the indicator is located right of the middle. If you play the correct pitch, the indicator is located in the middle and turns gray.


**Note**

Shows the currently played pitch.

**Cent**

Shows the deviation in pitch. A negative value indicates that the pitch is flat. A positive value indicates that the pitch is sharp.

**Base**

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

**Octave**

Shows the octave of the played note.

**Frequency**

Shows the frequency of the played note.

**Mute**

Mutes/Unmutes the output signal.

**Strobe/Classic**

Allows you to toggle the display between **Strobe** and **Classic** mode.

# MIDI Effects

This chapter describes the included MIDI realtime effects and their parameters. How to apply and handle MIDI effects is described in the **Operation Manual**.

## Arpache 5

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


**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.

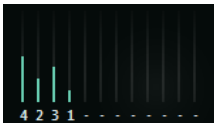
- To create staccato arpeggios, set a smaller value for **Length** than for **Step Size**.
- To create arpeggio notes that overlap each other, set a greater value for **Length** than for **Step Size**.

**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.


## 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**

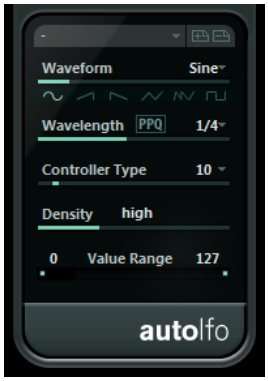
In this mode, the chords played are not separated into notes. Instead, they are used as is, and only the rhythm of the dropped MIDI part is used for playback.

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.




**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**

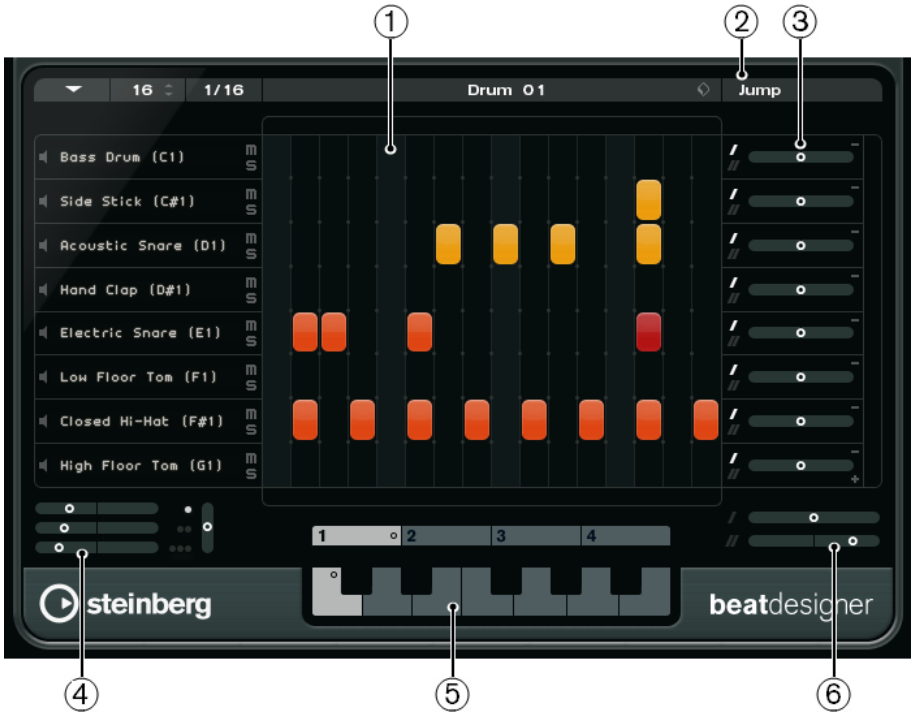
These sliders determine the range of controller values that are sent out, in other words, the bottom and top of the controller curves.

# 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.

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 108
Swing Setting on page 105

## 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.

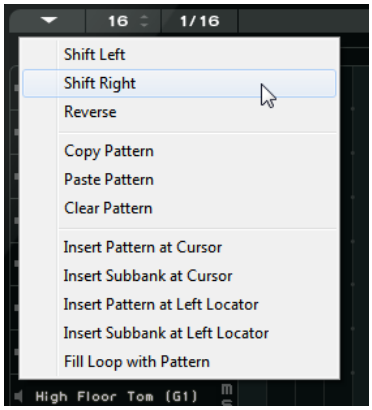
## Lane Handling

- 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 107

## 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**.

   - 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 104

## 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

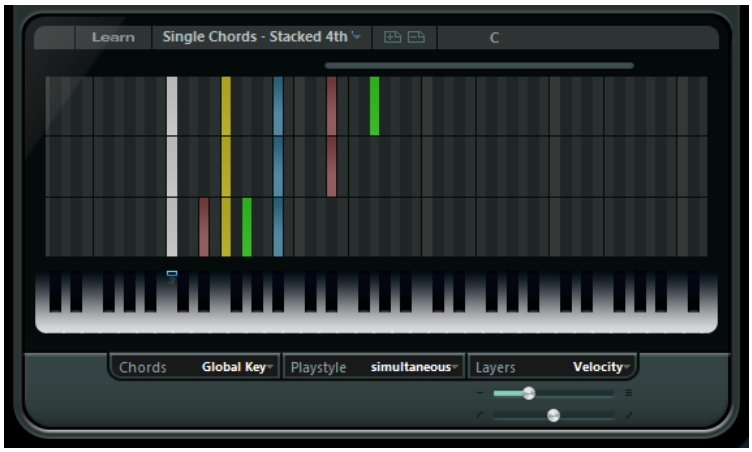
When triggering a pattern that contains sound before the first step (due to flams or lane offsets), these are taken into account as well.

# 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.

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.

## Chord Indicator Lane

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

NOTE

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

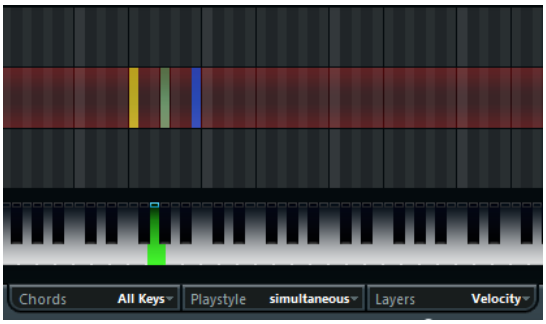
RELATED LINKS
Using Layers on page 111

## 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




   The second layer in **Learn** mode

2. Select the key to which you want to assign a chord by clicking on it on the keyboard display or by pressing the key on a connected MIDI keyboard.
   The red bar now moves to the first layer, indicating that you are ready to record the first chord.

   NOTE

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

3. Play a chord on the MIDI keyboard and/or use the mouse to enter or change the chord in the layer display.
   - Any notes you enter are immediately shown in the **Chorder** display. The notes are shown in different colors, depending on the pitch.
   - If you are entering chords via a MIDI keyboard, **Chorder** learns the chord as soon as you release all keys of your MIDI keyboard.
     As long as a key is pressed, you can continue looking for the right chord.
   - If more than one layer is shown, **Chorder** jumps automatically to the next layer where you can record another chord.
     If all layers for a key are filled, the red bar jumps back to the keyboard display so that you can choose a different trigger key (in **Global Key** mode, the **Learn** mode is deactivated).
   - If you are entering chords with the mouse, **Chorder** does not jump to the next layer automatically.
     You can select/deselect as many notes as you want and then click on another layer or deactivate the Learn mode to continue.
4. Repeat the above with any other keys you want to use.

# Using Layers

The **Layers** pop-up menu at the bottom right of the window allows you to set up chord variations in the layer display above the keyboard. This works with all three modes and provides up to 8 variations for each assignable key, that is, a maximum of 8 different chords in **Global Key** mode, 12 x 8 chords in **One Octave** mode and 128 x 8 chords in **All Keys** mode.

The different layers can be triggered by velocity or interval.

PROCEDURE

1. 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 111

## 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**

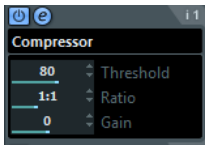
In this mode, the notes are played back in a rapidly changing random order.

**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.




**Threshold**

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

**Ratio**

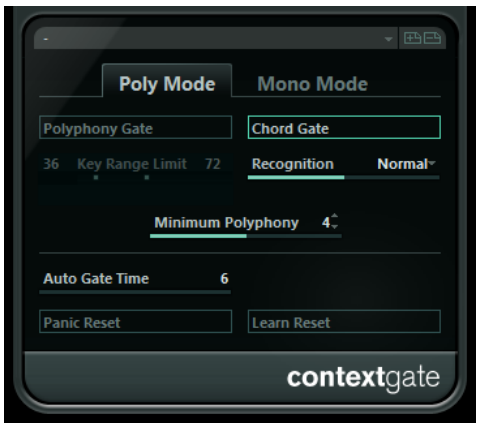
Sets the amount of compression applied to the velocity values above the set threshold. Ratios greater than 1:1 result in compression. Ratios lower than 1:1 result in expansion.

**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.




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 114

# Application Examples

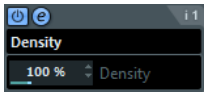
## Poly Mode

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 you guitar without having to use an additional MIDI instrument.

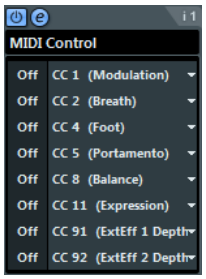
# Density




This generic control panel affects the density of the notes being played from or 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.




- 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.


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.

| Note Value | Ticks |
|---|---|
| 1/32 note | 60 |

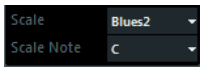
| Note Value | Ticks |
|---|---|
| 1/16 note triplet | 90 |
| 1/16 note | 120 |
| 1/8 note triplet | 160 |
| 1/8 note | 240 |
| Quarter note triplet | 320 |
| Quarter note | 480 |
| Half note | 960 |

# MIDI Modifiers

This plug-in is essentially a duplicate of the **MIDI Modifiers** section in the **Inspector**. This can be useful, for example, if you need extra **Random** or **Range** settings.

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

## Scale Transpose

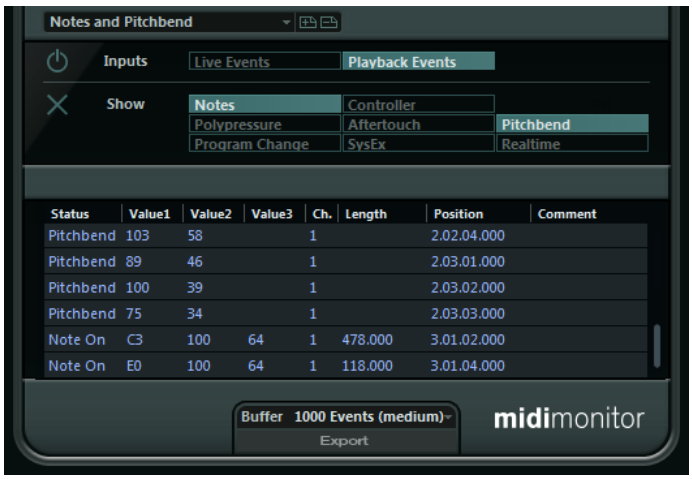



Allows you to transpose each incoming MIDI note, so that it fits within a selected musical scale. The scale is specified by selecting a key (C, C#, D, etc.) and a scale type (major, melodic or harmonic minor, blues, etc.).

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

# MIDI Monitor

This effect monitors incoming MIDI events.




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.

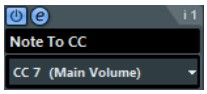
# Micro Tuner


**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




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.

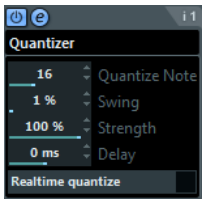
The purpose of this plug-in is to generate a gate effect. This means that the notes that are played control something else. For example, if **Main Volume** (CC 7) is selected, notes with low velocity lower the volume in the MIDI instrument, while notes with a high velocity raise the volume.

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.




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**

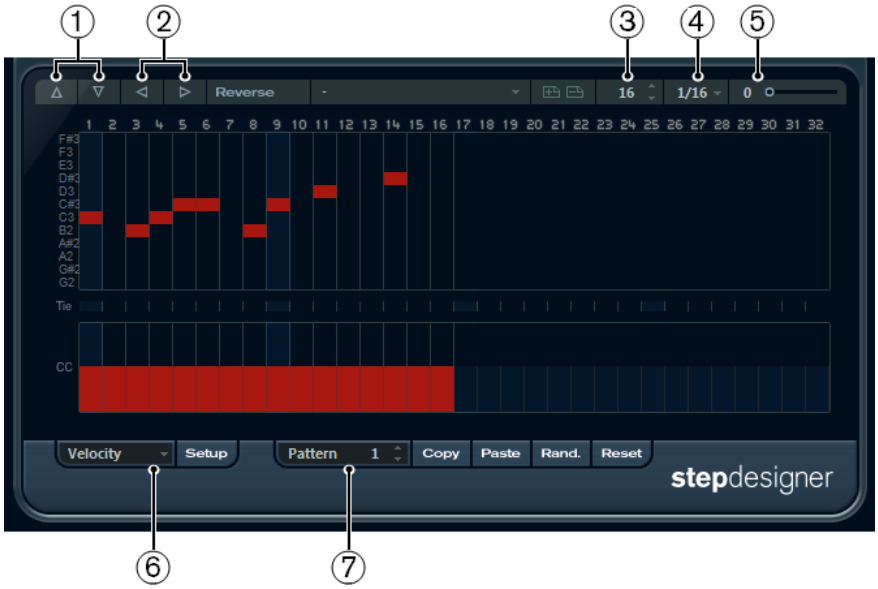
Sets the delay time in milliseconds. This delay can be automated.

**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).




**1** Shift octave up/down

**2** Shift steps left/right

**3** Number of steps

**4** Step size

**5** Swing

**6** Controller pop-up menu

**7** Pattern selector

# Creating a Basic Pattern

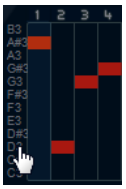
PROCEDURE

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.

NOTE

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

RESULT

If you now start playback in your host application, the pattern plays as well, sending out MIDI notes on the track's MIDI output and channel (or, if you activated **StepDesigner** as a send effect, on the MIDI output and channel selected for the send in the **Inspector**).

# Adding Controller Curves

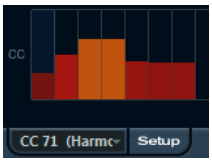
PROCEDURE

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.

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**

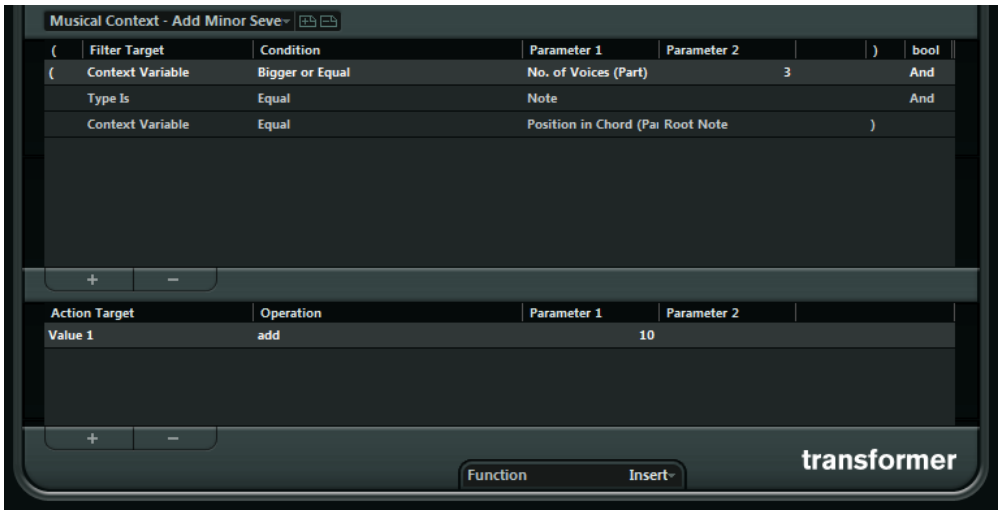
Sends an XG reset message.

**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.




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.

# Included VST Instruments

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

## Groove Agent SE

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

## HALion Sonic SE

This VST instrument is described in detail in the separate document **HALion Sonic SE**.

## LoopMash


**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 130
Applying Slice Selection Modifiers and Slice Effects on page 134

## 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

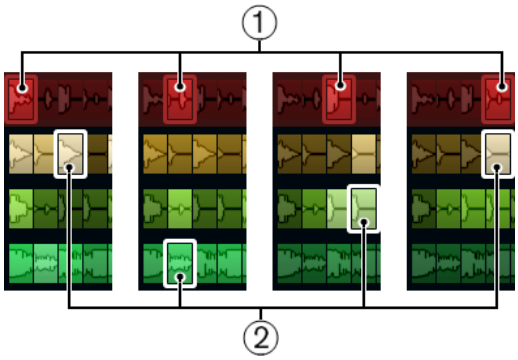
LoopMash does not categorize the sounds, but looks for overall similarity in the sound. For example, LoopMash might replace a low snare drum sound with a kick drum sound, even though a high snare sound is also available. LoopMash always tries to create a loop acoustically similar to the master loop,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

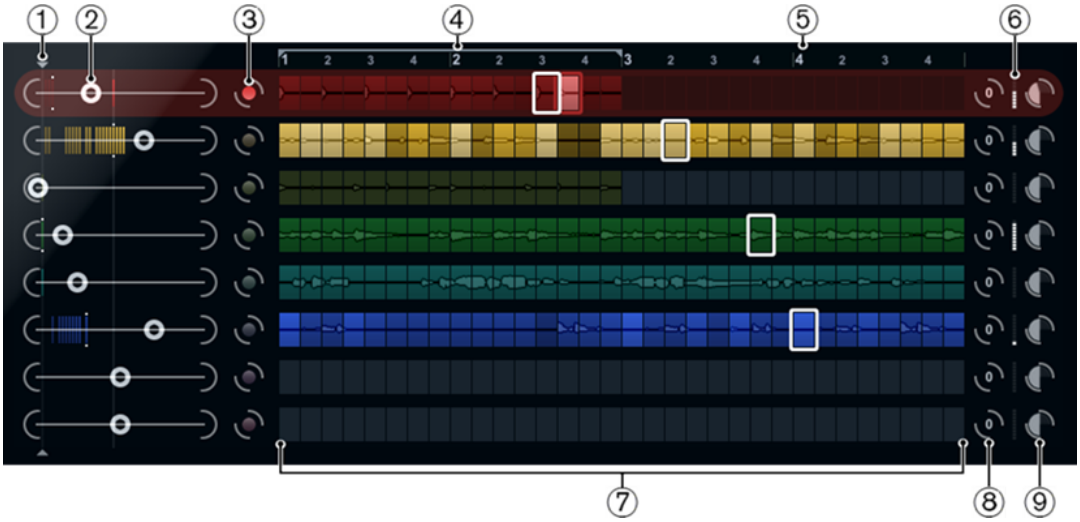
# LoopMash Parameters

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.

NOTE

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

## Defining the Master Loop

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

PROCEDURE

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

## Auditioning Slices

PROCEDURE

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

RELATED LINKS

Transport Controls on page 135

## 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 136

## 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 138

## 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 137

## Creating Composite Tracks

LoopMash allows you to build composite tracks.

PROCEDURE

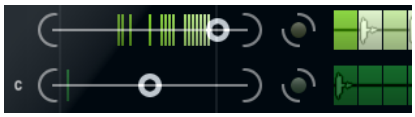
1. Import the loop that you want to extract sounds from.
2. Audition the slices and drag the slices that you want to use onto an empty track.

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


3. Click **OK**.

RESULT

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




You can use this feature in a very versatile way:

- You can assemble a combination of sounds that you like most on one track.
- You can define a certain rhythmic pattern by combining slices from different loops on a composite track and making this track the master loop.
- You can use a composite track as a clipboard, allowing you to include sounds from more than eight loops into your mash.

  You can use one track for importing and removing the loops that you want to search for sounds, and use the remaining seven tracks as composite tracks. This allows for including up to 32 sounds from up to 32 different loop files on each of the seven composite tracks.

NOTE

Composite tracks are quantized according to the set tempo.

RELATED LINKS
Transport Controls on page 135

## 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 137
Performance Controls on page 138

# Transport Controls


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

**Play**

Click the **Play** button to start or stop playback.

**Locate**

Click the **Locate** button to return to the beginning of the loop (bar 1/beat 1). Playback always starts automatically when clicking this button.

**Step left/right**

Clicking the **Step left/right** button steps backwards/forwards through the timeline, playing one slice at a time.

## Setting the LoopMash Tempo

During playback, **LoopMash** can be synchronized to the tempo set in your host application, or can follow its own tempo setting.

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

  If **sync** is activated, you can start playback using the transport controls of your host application. With **sync** deactivated, **LoopMash** starts playing when you click the **Play** button in **LoopMash**.
- If **sync** is deactivated, the current **LoopMash** tempo (in BPM) is displayed in the tempo field to the left of the master button. To change the local tempo, click in the tempo field, enter a new value, and press **Enter**.
- If **sync** is deactivated, you can click the master button (to the right of the tempo field) to copy the tempo of the current master loop into the tempo field.

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

## Controlling Transport Functions with Your MIDI Keyboard

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

**C2**

Start

**D2**

Stop

**E2**

Sync on

**F2**

Sync off

NOTE

If you do not have a MIDI keyboard connected to your computer, you can use the virtual keyboard (see the **Operation Manual**).

# Storing Your Configuration as Scenes

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


**1** Save scene

**2** Remove scene

**3** Jump interval

**4** Selected scene

**5** Pad with associated scene

**6** Empty scene pad

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

IMPORTANT

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

## Setting a Jump Interval

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

PROCEDURE

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

RESULT

NOTE

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

## Triggering Scene Pads with Your MIDI Keyboard

The scene pads are arranged according to the keys on a MIDI keyboard. You can trigger the 24 scene pads with a connected MIDI keyboard starting from C0 and ending with B1.

# 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 136

# 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 132

# Performance Controls


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

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

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

Most of the available effects correspond to the effects that you can apply to single slices, with the green buttons corresponding to the stutter and slur effects and the red buttons to the Mute, Reverse, Staccato effects, etc.

NOTE

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

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

**Cycle 4, 2, 1**

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

**Continue**

Plays back the tracks of the selected slices continuously until you release the button.

NOTE

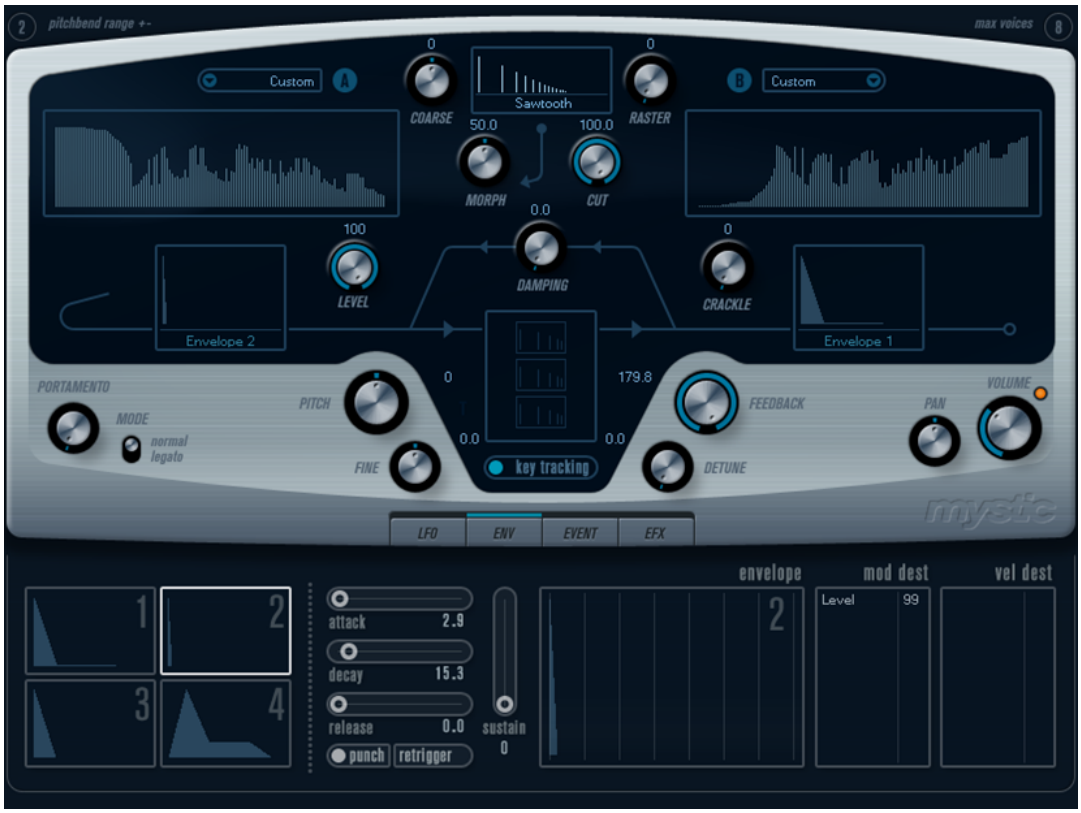
You cannot save global effects in scenes. To apply effects and save them in scenes, use slice effects.

### Triggering the Performance Controls with Your MIDI Keyboard

You can trigger the performance controls with your MIDI keyboard starting from C3 upwards.

RELATED LINKS
Applying Slice Selection Modifiers and Slice Effects on page 134

# Mystic



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

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

The basic principle is the following:

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

  The spectrum of the impulse sound largely affects the tonal quality of the final sound.
- The impulse sound is fed into the three comb filters, in parallel. Each of these has a feedback loop.

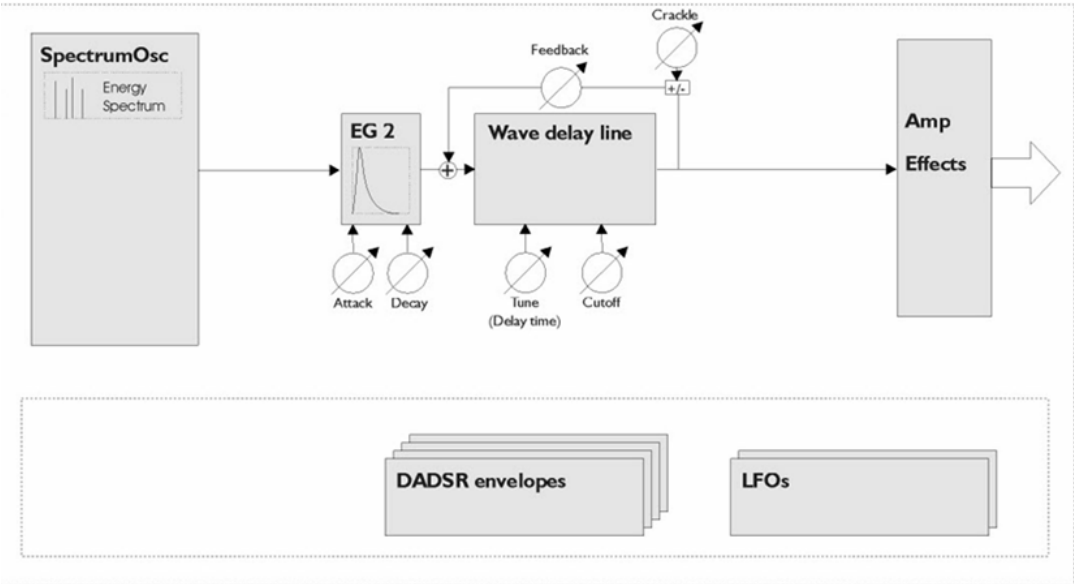
  This means the output of each comb filter is fed back into the filter. This results in a resonating feedback tone.
- When the signal is fed back into the comb filter, it goes via a separate, variable low-pass filter.

  This filter corresponds to the damping of high frequencies in a physical instrument – if this is set to a low cutoff frequency it causes high harmonics to decay faster than the lower harmonics (as when plucking a string on a guitar, for example).
- The level of the feedback signal is governed by a feedback control.

  This determines the decay of the feedback tone. Setting this to a negative value simulates the traveling wave in a tube with one open end and one closed end. The result is a more hollow, square wave-like sound, pitched one octave lower.
- A detune control offsets the fundamental frequencies of the three comb filters, for chorus-like sounds or drastic special effects.

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

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




Functional Diagram

# Sound Parameters

## The Impulse Control Section


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

### Spectrum Displays

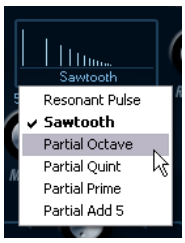

The displays allow you to draw a filter contour with your mouse for spectrum filters A and B.

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

  To set up the contour independently for the two filters, hold down **Shift** and click and drag the mouse in either display.
- Use the Preset pop-up menu to select a preset contour.
- If you want to calculate a random spectrum filter curve, you can choose the **Randomize** function from the Preset pop-up menu.

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

### Waveform Pop-up Menu




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

**Cut**

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

**Morph**

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

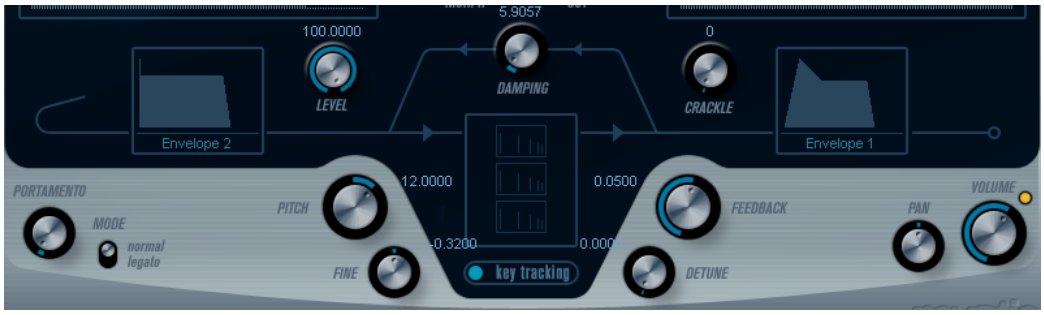
**Coarse**

Offsets the pitch for the impulse sound. In a typical string setup, when the impulse sound is very short, this does not change the pitch of the final sound, but the tonal color.

**Raster**

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

## Comb Filter Sound Parameters




**Damping**

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

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

  If you open the filter completely (turn **Damping** up to max) the harmonic content is static – the sound does not get softer when decaying.

**Level**

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

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

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

**Crackle**

Allows you to send noise directly into the comb filters. Small amounts of noise produce a crackling, erratic effect, higher amounts give a more pronounced noise sound.

**Feedback**

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

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

**Detune**

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

**Pitch and Fine**

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

**Key Tracking**

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

**Portamento**

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

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

## Master Volume and Pan


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

The **Pan** knob controls the position of the instrument in the stereo spectrum. You can use **Pan** as a modulation destination.

# Modulation and Controllers

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

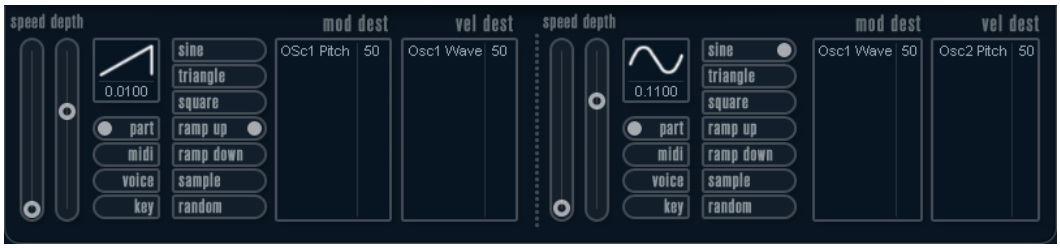

The following pages are available:

- 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 144
Envelope Page on page 146
Event Page on page 148
Effects (EFX) Page on page 149

## 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 145

## 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.

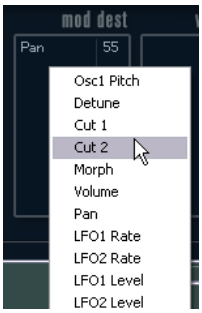
## Assigning LFO Modulation Destinations

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

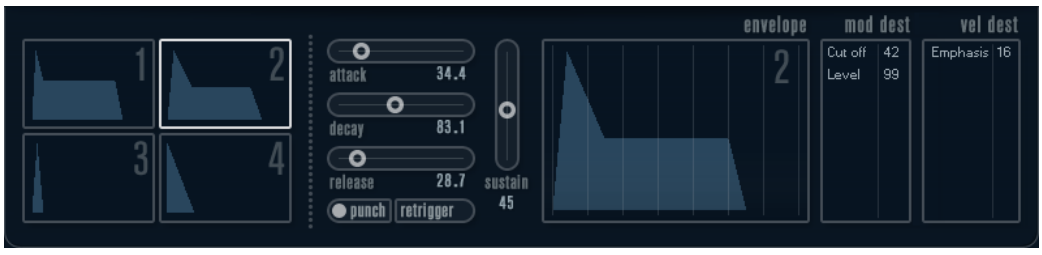
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- 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.

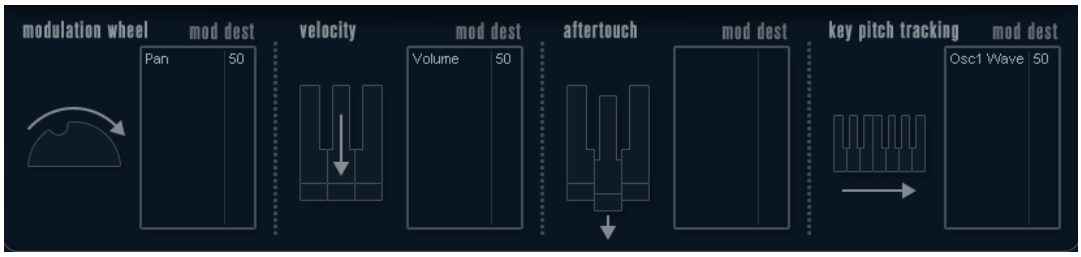
## Envelope modulation velocity control

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.

## Assigning 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.

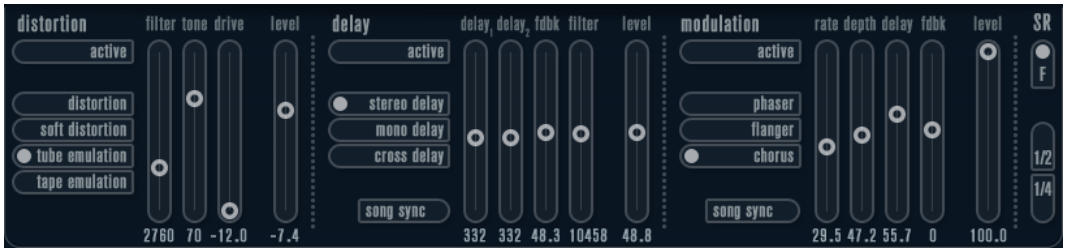
3. Using the same basic method, you can add any number of modulation destinations for the controllers.

   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

This VST instrument is described in detail in the separate document **Padshop**.

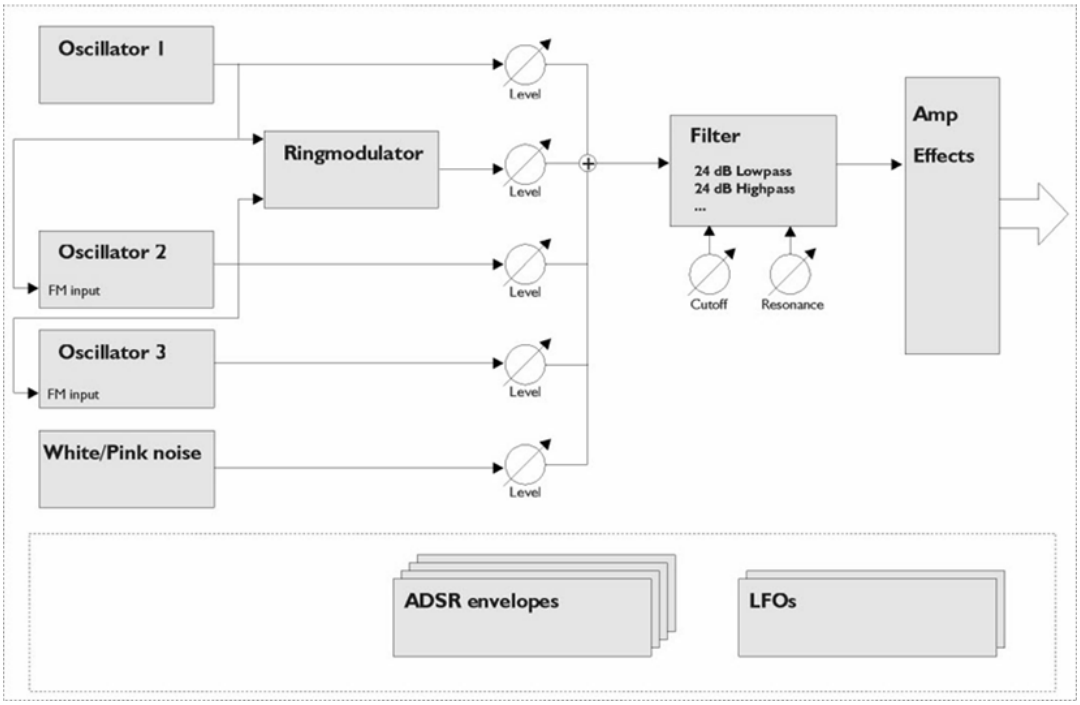
# Prologue


**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**.




Functional Diagram

# 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 emphasizes certain frequency bands. Like the human voice, musical instruments have a fixed set of formants, which give it a unique, recognizable tonal color or timbre, regardless of pitch.

**Vocal 1–7**

These are also formant waveforms, but specifically vocal-oriented. Vowel sounds (A/E/I/O/U) are among the waveforms found in this category.

**Partial 1–7**

Partials, also called harmonics or overtones, are a series of tones which accompany the prime tone (fundamental). These waveforms 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 (±48semitones)**

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 (±48semitones)**

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 146

## 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:

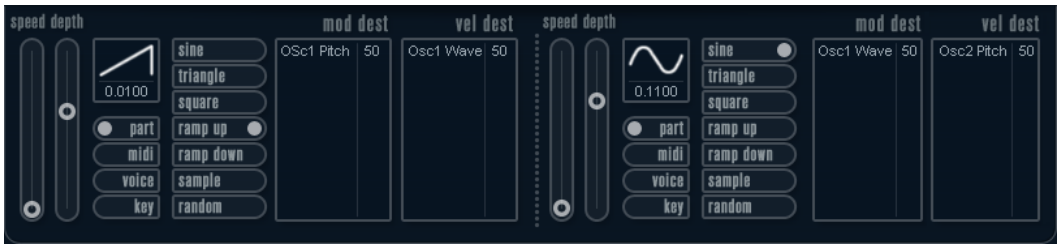
- The **LFO** page has two low frequency oscillators (LFOs) for modulating parameters.
- The **ENV** page contains the four envelope generators that can be assigned to control parameters.
- The **Event** page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments.

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

RELATED LINKS
LFO Page on page 144
Envelope Page on page 146
Event Page on page 148
Effects (EFX) Page on page 149

## 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 145

## 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.

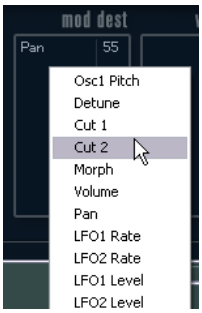
## Assigning LFO Modulation Destinations

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

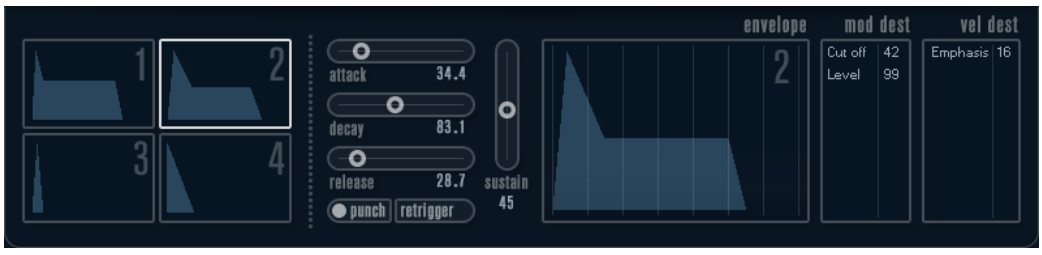
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- 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.

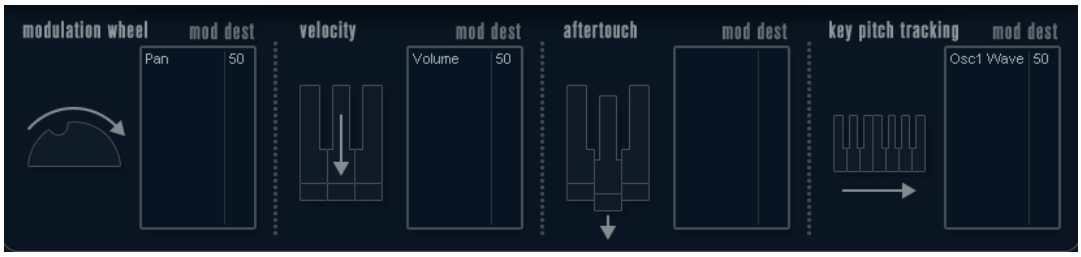
## Envelope modulation velocity control

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.

## Assigning 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.

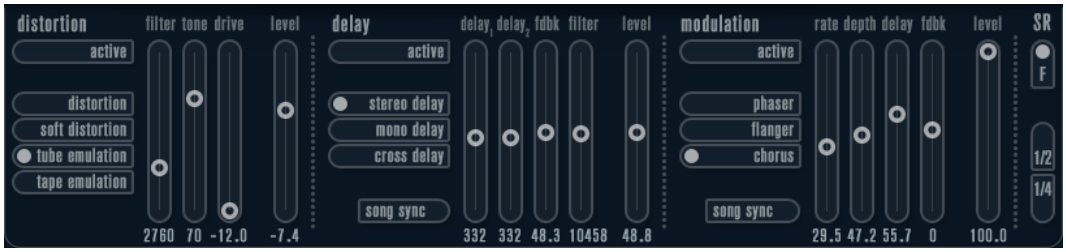
3. Using the same basic method, you can add any number of modulation destinations for the controllers.

   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

This VST instrument is described in detail in the separate document **Retrologue**.

# Spector


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.

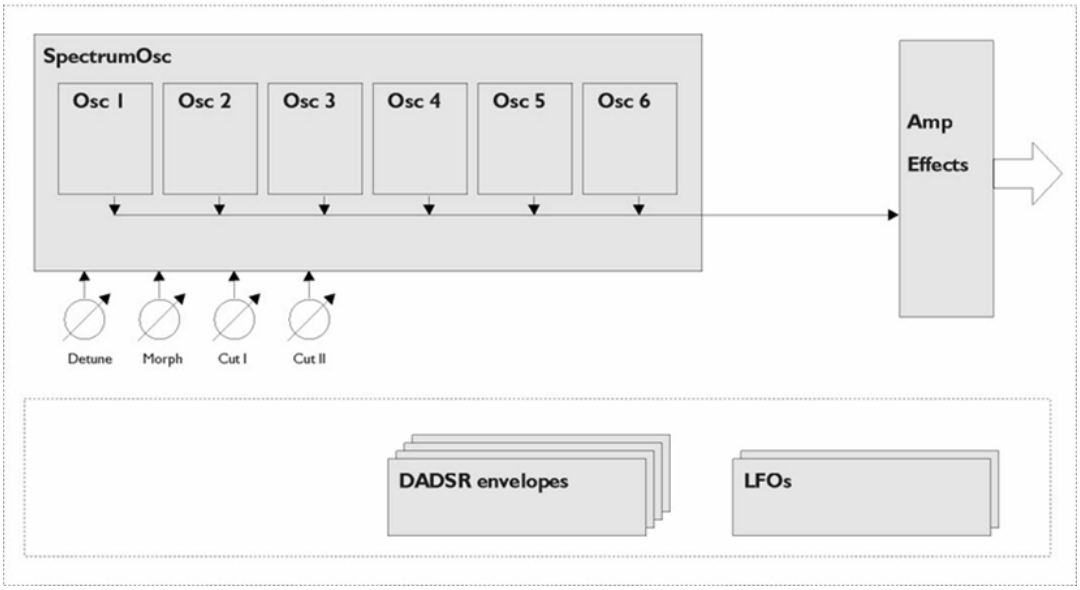
  You can choose between different numbers of oscillators in different configurations (in octaves, in unison, etc.). The oscillators can also be detuned for fat sounds or extreme special effects.
- Each oscillator produces two basic waveforms, labeled A and B.

  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.




Functional Diagram

# Sound Parameters

## Oscillator Section


### A/B Waveform Pop-up Menus

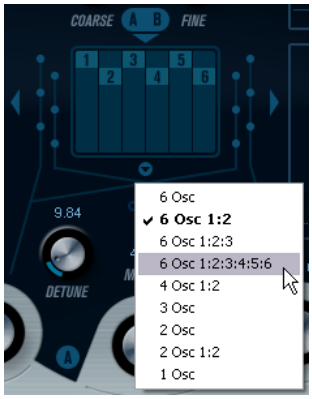
This is where you select basic waveforms for the A and B output of the oscillators. The options are best suited for use with the spectrum filter.

### 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.

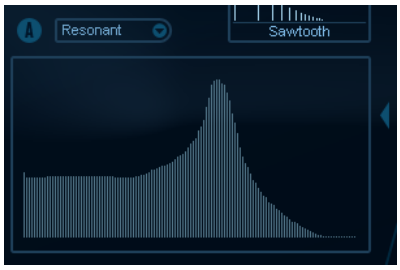
And so on.

## 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.

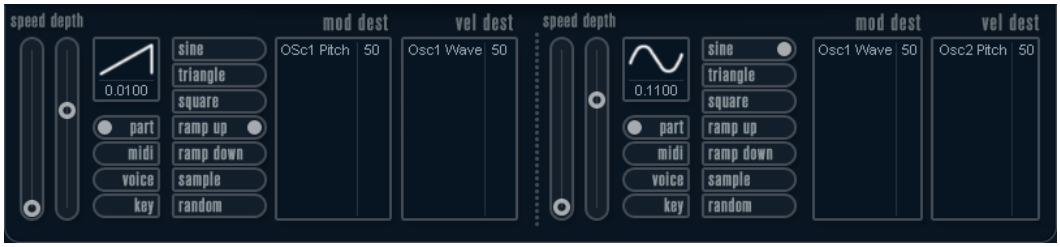

The following pages are available:

- 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.

RELATED LINKS
LFO Page on page 144
Envelope Page on page 146
Event Page on page 148
Effects (EFX) Page on page 149

## 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 145

## 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.

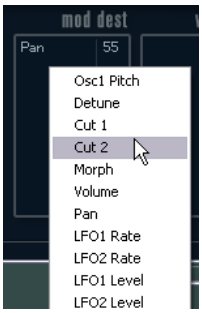
## Assigning LFO Modulation Destinations

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

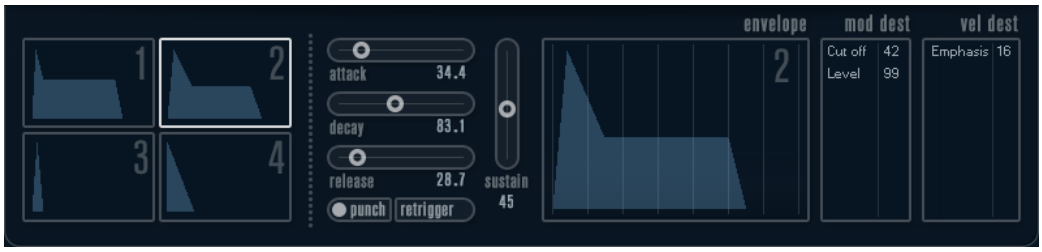
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- 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.

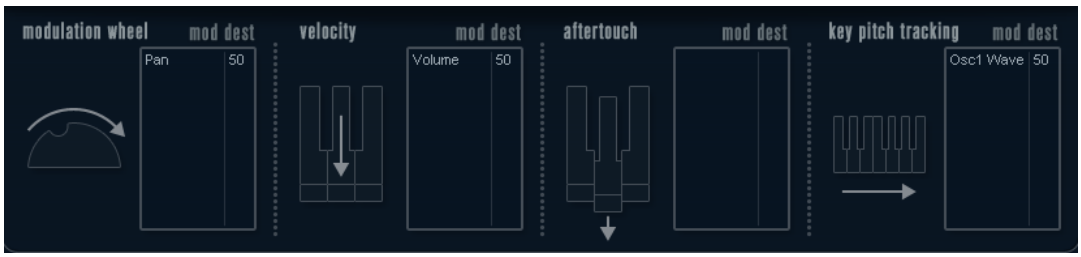
## Envelope modulation velocity control

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.

## Assigning 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.

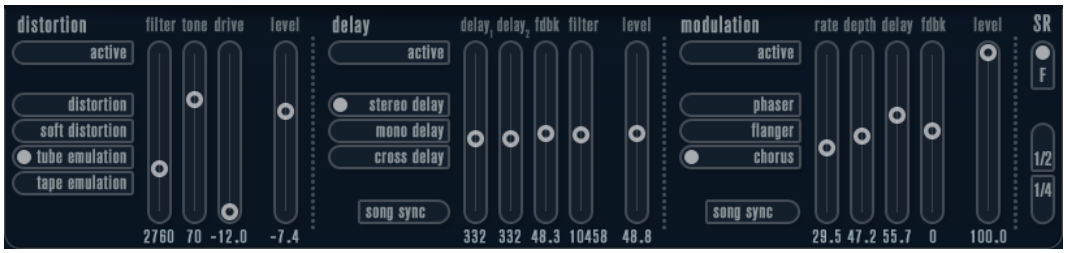
3. Using the same basic method, you can add any number of modulation destinations for the controllers.

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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