Plug-in Reference

CUBASE ELEMENTS 10.5

Personal Music Production System

steinberg

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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.

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.

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.

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.

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.

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.

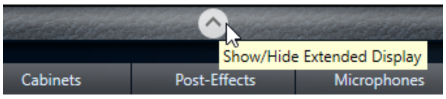
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

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

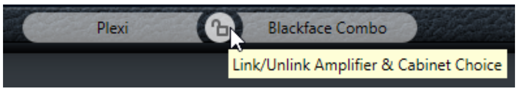
- To toggle between the different views, click the down/up arrow button (Show/Hide Extended Display) at the top center of the plug-in frame.



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.

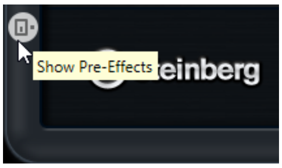
- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the **Link/Unlink Amplifier & Cabinet Choice** button. If you now select another amp model, the cabinet selection follows. However,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



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 1 ms. **Brickwall Limiter** features separate meters for input, output, and the amount of limiting. Position this plug-in at the end of the signal chain, before dithering.

Threshold (-20 to 0 dB)

Determines the level where the limiter kicks in. Only signal levels above the set threshold are processed.

Release (3 to 1000 ms or Auto mode)

Sets the time after which the gain returns to the original level when the signal drops below the threshold. If the **Auto** button is activated, the plug-in automatically finds the best release setting for the audio material.

Link

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

Detect Intersample Clipping

If this option is activated, **Brickwall Limiter** uses oversampling to detect and limit signal levels between two samples to prevent distortion when converting digital signals into analog signals.

NOTE

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

Compressor

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

Compressor features a separate display that graphically illustrates the compressor curve that is shaped according to the **Threshold** and **Ratio** parameter settings. **Compressor** also features a **Gain Reduction** meter that shows the amount of gain reduction in dB, **Soft knee**/**Hard knee** compression modes, and a program-dependent auto feature for the **Release** parameter.

Threshold (-60 to 0 dB)

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.

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 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.

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.

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.

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.

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 27

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.

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.

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 -∞, 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.

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 39
Controlling StepFilter via MIDI on page 40

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 7

WahWah

WahWah is a variable slope band-pass filter that can be auto-controlled 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.

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.

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.

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.

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.

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.

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.

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

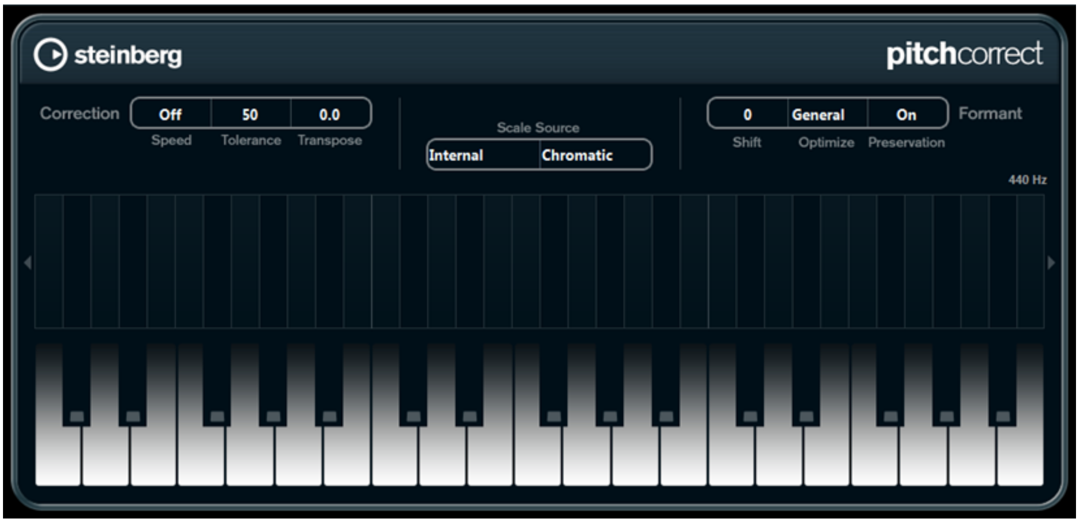
Adjusts the level of the generated signal one octave below the original pitch. A setting of 0 means that the voice is muted.

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

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.

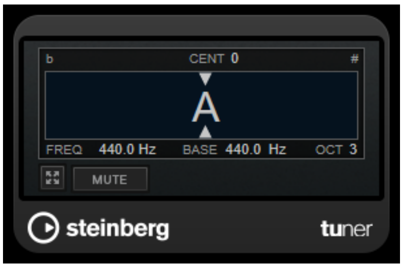
NOTE

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.

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

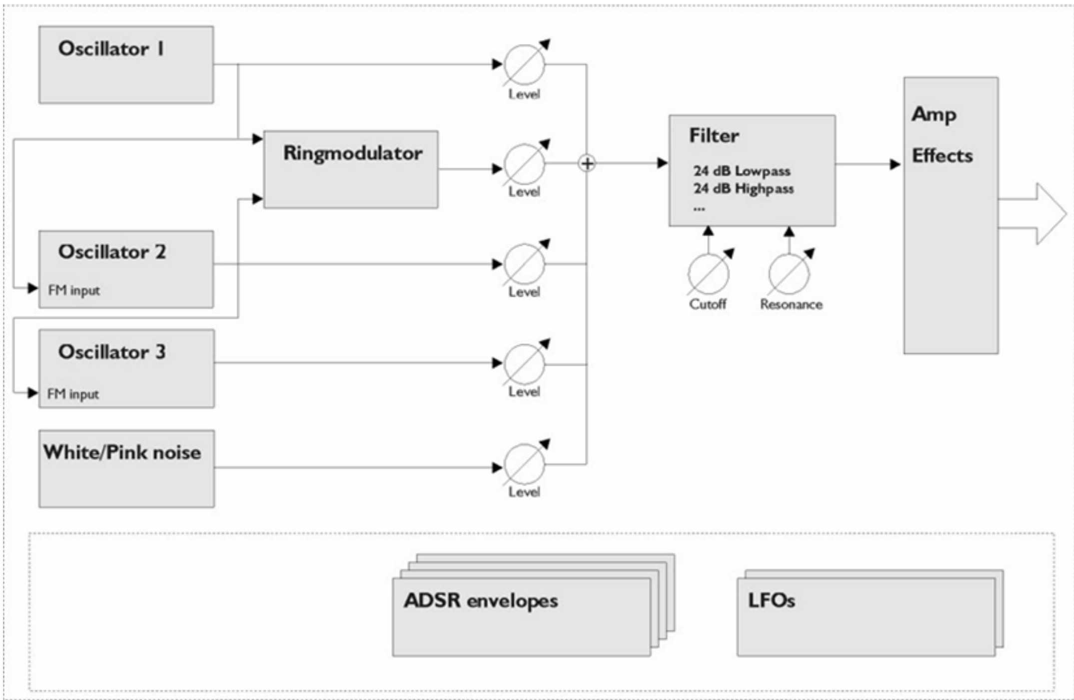
This VST instrument is described in detail in the separate document **HALion Sonic SE**.

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 74

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:

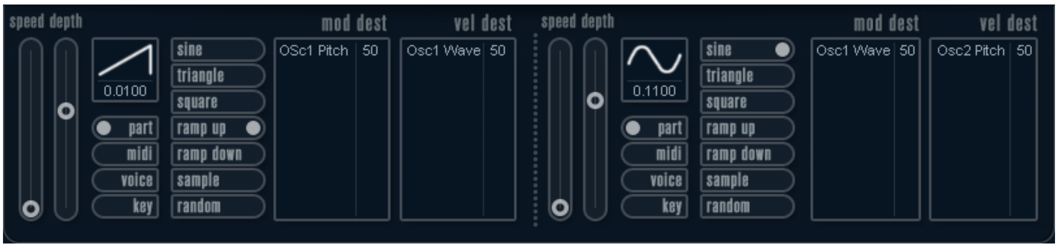
- The **LFO** page has two low frequency oscillators (LFOs) for modulating parameters.

- 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 72
Envelope Page on page 74
Event Page on page 76
Effects (EFX) Page on page 77

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 73

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.

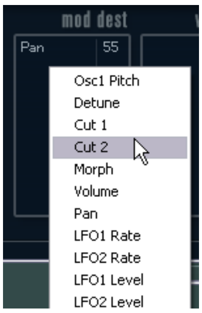
Assigning LFO Modulation Destinations

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

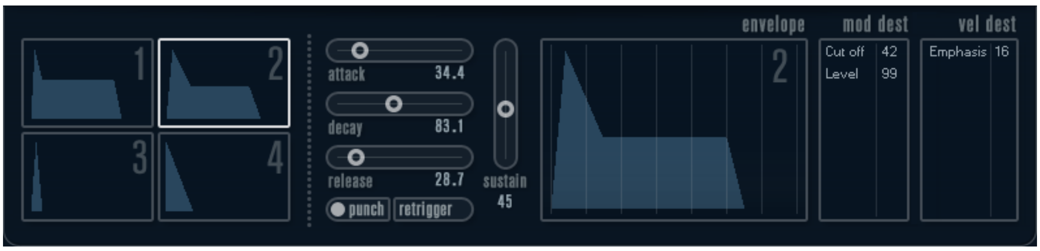
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- 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.

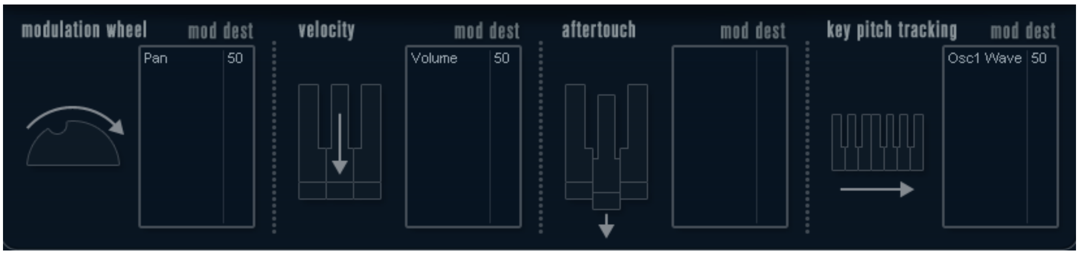
Envelope modulation velocity control

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.

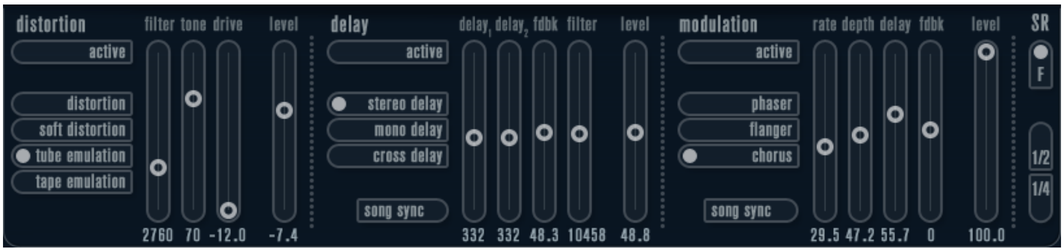
3. Using the same basic method, you can add any number of modulation destinations for the controllers.

   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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