Plug-in Reference

CUBASE ELEMENTS 6
Personal Music Production System

CUBASE AI 6
Integrated Music Production Software

CUBASE LE 6
Music Production Software

steinberg

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

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The included effect plug-ins

Introduction

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

In Cubase, the plug-in effects are arranged in a number of different categories. This chapter is arranged in the same fashion, with the plug-ins listed in separate sections for each effect category.

⇨ Most of the included effects are compatible with VST3, this is indicated by an icon in front of the name of the plug-in as displayed in plug-in selection menus (for further information, see the chapter "Audio effects" in the Operation Manual).

Delay plug-ins

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

MonoDelay

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

The following parameters are available:

Parameter	Description
Delay	If tempo sync is on, this is where you specify the base note value for the delay (1/1–1/32, straight, triplet, or dotted). If tempo sync is off, the delay time can be set freely in milliseconds.
Sync button	The button below the Delay knob is used to switch tempo sync on or off.
Feedback	Sets the number of repeats for the delay.
Filter Lo	This filter affects the feedback loop of the effect signal and allows you to roll off low frequencies from 10 Hz up to 800 Hz. The button below the knob activates/deactivates the filter.
Filter Hi	This filter affects the feedback loop of the effect signal and allows you to roll off high frequencies from 20 kHz down to 1.2 kHz. The button below the knob activates/deactivates the filter.
Mix	Sets the level balance between the dry and the wet signal. If MonoDelay is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

PingPongDelay (Cubase Elements only)

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

The following parameters are available:

Parameter	Description
Delay	If tempo sync is on, this is where you specify the base note value for the delay (1/1–1/32, straight, triplet, or dotted). If tempo sync is off, the delay time can be set freely in milliseconds.
Sync button	The button below the Delay Time knob is used to switch tempo sync on or off.
Feedback	Sets the number of repeats for the delay.
Filter Lo	This filter affects the feedback loop and allows you to roll off low frequencies up to 800 Hz. The button below the knob activates/deactivates the filter.
Filter Hi	This filter affects the feedback loop and allows you to roll off high frequencies from 20 kHz down to 1.2 kHz. The button below the knob activates/deactivates the filter.
Spatial	Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo "ping-pong" effect.
Mix	Sets the level balance between the dry and the wet signal. If PingPongDelay is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

Distortion plug-ins

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

AmpSimulator

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

The following parameters are available:

Parameter	Description
Amplifier pop-up menu	This pop-up menu is opened by clicking on the amplifier name shown at the top of the amp section. It allows you to select an amplifier model. The amp section can be bypassed by selecting "No Amp".
Drive	Controls the amount of amp overdrive.
Bass	Tone control for the low frequencies.
Middle	Tone control for the mid frequencies.
Treble	Tone control for the high frequencies.
Presence	Boosts or dampens the higher frequencies.
Volume	Controls the overall output level.
Cabinet pop-up menu	This pop-up menu is opened by clicking on the cabinet name shown at the top of the cabinet section. It allows you to select a speaker cabinet model. This section can be bypassed by selecting "No Speaker".
Damping Lo/Hi	Further tone controls for shaping the sound of the selected speaker cabinet. Click on the values, enter a new value and press the [Enter] key.

BitCrusher

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

The following parameters are available:

Parameter	Description
Mode	Allows you to select one of the four operating modes of BitCrusher. In each mode the plug-in sounds differently. Modes I and III are nastier and noisier, while modes II and IV are more subtle.
Sample Divider	Sets the amount by which the audio samples are decimated. At the highest setting (65), nearly all of the information describing the original audio signal is eliminated, turning the signal into unrecognizable noise.
Depth	Defines the bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 creates mostly noise.
Output slider	Governs the output level from BitCrusher. Drag the slider upwards to increase the level.
Mix slider	Regulates the balance between the output from BitCrusher and the original audio signal. Drag the slider upwards for a more dominant effect, and downwards if you want the original signal to be more prominent.

DaTube (not in Cubase LE)

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

The following parameters are available:

Parameter	Description
Drive	Regulates the pre-gain of the “amplifier”. Use high values if you want an overdriven sound just on the verge of distortion.
Balance	Controls the balance between the signal processed by the Drive parameter and the dry input signal. For maximum drive effect, set this to its highest value.
Output	Adjusts the post-gain, or output level, of the “amplifier”.

Distortion

Distortion will add crunch to your tracks.

The following parameters are available:

Parameter	Description
Boost	Increases the distortion amount.
Feedback	Feeds part of the output signal back to the effect input, increasing the distortion effect.
Tone	Lets you select a frequency range to which to apply the distortion effect.
Spatial	Changes the distortion characteristics of the left and right channel, thus creating a stereo effect.
Output	Raises or lowers the signal going out of the effect.

Grungelizer

Grungelizer adds noise and static to your recordings – kind of like listening to a radio with bad reception, or a worn and scratched vinyl record. The following parameters are available:

Parameter	Description
Crackle	Adds crackle to create that old vinyl record sound. The farther to the right you turn the knob, the more crackle is added.
RPM switch	When emulating the sound of a vinyl record, this switch lets you set the RPM (revolutions per minute) speed of the record (33/45/78 RPM).
Noise	Regulates the amount of static noise added.
Distort	Adds distortion.
EQ	Turn this knob to the right to cut off the low frequencies, and create a more hollow, lo-fi sound.
AC	Emulates a constant, low hum of AC current.
Frequency switch	Sets the frequency of the AC current (50 or 60 Hz), and thus the pitch of the AC hum.
Timeline	Regulates the amount of overall effect. The farther to the right (1900) you turn the knob, the more noticeable the effect.

Dynamics plug-ins

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

Compressor (Cubase Elements only)

Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. Compressor features separate controls for threshold, ratio, attack, hold, release and make-up gain parameters. Compressor features a separate display that graphically illustrates the compressor curve shaped according to the Threshold and Ratio parameter settings. Compressor also features a Gain Reduction meter that shows the amount of gain reduction in dB, Soft knee/Hard knee compression modes and a program-dependent Auto feature for the Release parameter.

The following parameters are available:

Parameter	Description
Threshold (-60 to 0dB)	Determines the level where Compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1:1 to 8:1)	Sets the amount of gain reduction applied to signals over the set threshold. A ratio of 3:1 means that for every 3dB the input level increases, the output level will increase by only 1dB.
Soft Knee button	If this button is off, signals above the threshold are compressed instantly according to the set ratio (hard knee). When Soft Knee is activated, the onset of compression is more gradual, producing a less drastic result.
Make-up (0 to 24dB or Auto mode)	This parameter is used to compensate for output gain loss, caused by compression. If the Auto button is activated, the knob becomes dark and the output is automatically adjusted for gain loss.
Attack (0.1 to 100ms)	Determines how fast Compressor will respond to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.
Hold (0 to 5000ms)	Sets the time the applied compression will affect the signal after exceeding the threshold. Short hold times are useful for “DJ-style” ducking, while longer hold times are required for music ducking, e.g. when working on a documentary film.
Release (10 to 1000ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, Compressor will automatically find an optimal release setting that varies depending on the audio material.
Analysis (0 to 100) (Pure Peak to Pure RMS)	Determines whether the input signal is analyzed according to peak or RMS values (or a mixture of both). A value of 0 is pure peak and 100 pure RMS. RMS mode operates using the average power of the audio signal as a basis, whereas Peak mode operates more on peak levels. As a general guideline, RMS mode works better on material with few transients such as vocals, and Peak mode better for percussive material, with a lot of transient peaks.
Live button	When this button is activated, the “look ahead” feature of Compressor is disengaged. Look ahead produces more accurate processing, but adds a certain amount of latency as a trade-off. When Live mode is activated, there is no latency, which might be better for “live” processing.

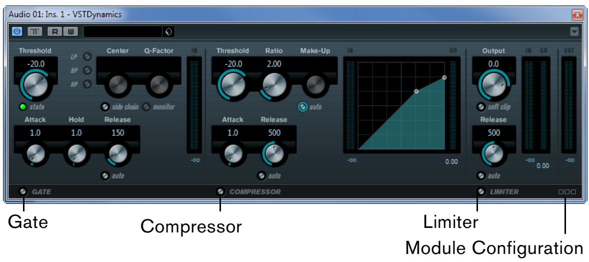
Limiter (not in Cubase LE)

Limiter is designed to ensure that the output level never exceeds a set output level, to avoid clipping in following devices. Limiter can adjust and optimize the Release parameter automatically according to the audio material, or it can be set manually. Limiter also features separate meters for the input, output and the amount of limiting (middle meters).

The following parameters are available:

Parameter	Description
Input (-24 to +24dB)	Allows you to adjust the input gain.
Output (-24 to +6dB)	Determines the maximum output level.
Release (0.1 to 1000ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level. If the Auto button is activated, Limiter will automatically find an optimal release setting that varies depending on the audio material.

VSTDynamics



VSTDynamics is an advanced dynamics processor. It combines three separate processors: Gate, Compressor and Limiter, covering a variety of dynamic processing functions. The window is divided into three sections, containing controls and meters for each processor.

Activating the individual processors

You activate the individual processors using the buttons at the bottom of the plug-in panel.

The Gate section

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

The following parameters are available:

Parameter	Description
Threshold (-60 to 0dB)	Determines the level where Gate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold close the gate.
State LED	Indicates whether the gate is open (LED lights up in green), closed (LED lights up in red) or something in between (LED lights up in yellow).
LP (low-pass), BP (band-pass), HP (high-pass)	These buttons set the basic filter mode.
Center (50 to 22000Hz)	Sets the center frequency of the filter.
Q-Factor (0.001 to 10000)	Sets the resonance or width of the filter.
Monitor (On/Off)	Allows you to monitor the filtered signal.
Attack (0.1 to 100ms)	Sets the time it takes for the gate to open after being triggered.
Hold (0 to 2000ms)	Determines how long the gate stays open after the signal drops below the threshold level.
Release (10 to 1000ms or Auto mode)	Sets the amount of time it takes for the gate to close (after the set hold time). If the Auto button is activated, Gate will find an optimal release setting, depending on the audio material.
Input gain meter	Shows the input gain.

The Compressor section

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

The available parameters work as follows:

Parameter	Description
Threshold (-60 to 0dB)	Determines the level where the compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1:1 to 8:1)	Determines the amount of gain reduction applied to signals above the set threshold. A ratio of 3:1 means that for every 3dB the input level increases, the output level increases by only 1dB.
Make-Up (0 to 24dB)	This parameter is used to compensate for output gain loss, caused by compression. When the Auto button is activated, gain loss is being compensated automatically.
Attack (0.1 to 100ms)	Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.

Parameter	Description
Release (10 to 1000 ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, the compressor will automatically find an optimal release setting that varies depending on the audio material.
Graphical display	Use the graphical display to graphically set the Threshold and Ratio values. To the left and right of the graphical display you will find two meters that show the amount of input gain and gain reduction in dB.

The Limiter section

The limiter is designed to ensure that the output level never exceeds a set threshold, to avoid clipping in following devices. Conventional limiters usually require very accurate setting up of the attack and release parameters to prevent the output level from going beyond the set threshold level. The limiter adjusts and optimizes these parameters automatically according to the audio material. You can also adjust the Release parameter manually.

The following parameters are available:

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

The Module Configuration button

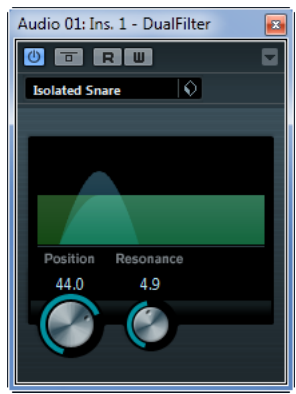
Using the Module Configuration button in the bottom right corner of the plug-in panel, you can set the signal flow order for the three processors. Changing the order of the processors can produce different results, and the available options allow you to quickly compare what works best for a given situation. Simply click the Module Configuration button to change to a different configuration. There are three routing options:

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

Filter plug-ins

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

DualFilter



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

The following parameters are available:

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

StepFilter (Cubase Elements only)

StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

General operation

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

Setting step values

- Setting step values is done by clicking in the pattern grid windows.

- Individual step entries can be freely dragged up or down the vertical axis, or directly set by clicking in an empty grid box. By click-dragging left or right, consecutive step entries are set at the pointer position.

- The horizontal axis shows the pattern steps 1 to 16 from left to right, and the vertical axis determines the (relative) filter cutoff frequency and resonance settings.
The higher up on the vertical axis a step value is entered, the higher the relative filter cutoff frequency or filter resonance setting.

- By starting playback and editing the patterns for the cutoff and resonance parameters, you can hear how your filter patterns affect the sound source connected to StepFilter.

Selecting new patterns

- Created patterns are saved with the project, and up to 8 different cutoff and resonance patterns can be saved internally.
Both the cutoff and resonance settings are saved together in the 8 pattern slots.

- Use the Pattern Selector below the Resonance grid to select a new pattern.
New patterns are all set to the same step value by default.

Using pattern copy and paste to create variations

You can use the Copy and Paste buttons below the Pattern Selector to copy a pattern to another pattern slot, which is useful for creating variations on a pattern.

- Select the pattern you wish to copy, click the Copy button, select another pattern slot, and click Paste.
The pattern is copied to the new slot, and can now be edited to create variations using the original pattern as a starting point.

StepFilter parameters

The following parameters are available:

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

ToneBooster (not in Cubase LE)

ToneBooster is a filter that allows you to raise the gain in a selected frequency range. It is particularly useful when inserted before AmpSimulator in the plug-in chain (see “AmpSimulator” on page 7), greatly enhancing the tonal varieties available.

The following parameters are available:

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

WahWah (not in Cubase LE)

WahWah is a variable slope band-pass filter that can be auto-controlled via MIDI modeling the well-known analog pedal effect (see below). You can independently specify the frequency, width and the gain for the Lo and Hi Pedal positions. The crossover point between the Lo and Hi Pedal positions lies at 50.

The following parameters are available:

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

MIDI control

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

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

Mastering plug-ins

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

UV22HR (Cubase Elements only)

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

The following parameters are available:

Option	Description
Bit Resolution	The UV22HR supports dithering to multiple resolutions: 8, 16, 20 or 24 bits. You select the desired resolution by clicking the corresponding button.
Hi	Try this first, it is the most "all-round" setting.
Lo	This applies a lower level of dither noise.
Auto black	When this is activated, the dither noise is gated (muted) during silent passages in the material.

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

Modulation plug-ins

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

AutoPan

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

The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the auto-pan speed can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Width	Sets the depth of the auto-pan effect.
Waveform Shape selector	Allows you to select the modulation waveform. A sine and a triangle waveform are available.

Chopper

Chopper is a combined tremolo and autopan effect. It can use different waveforms to modulate the level (tremolo) or left-right stereo position (pan), either using tempo sync or manual modulation speed settings.

The following parameters are available:

Parameter	Description
Waveform buttons	Set the modulation waveform.
Depth	Sets the depth of the Chopper effect. This can also be set by clicking in the graphical display.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the tremolo/auto-pan speed can be set freely with the Speed knob.
Sync button	The button above the Speed knob is used to switch tempo sync on (button lights up) or off.
Stereo/Mono button	Determines whether the Chopper works as an auto-panner (button set to "Stereo") or a tremolo effect (button set to "Mono").
Mix	Sets the level balance between the dry and the wet signal. If Chopper is used as a send effect, this should be set to the maximum value.

Chorus

This is a single-stage chorus effect. It works by doubling whatever is sent into it with a slightly detuned version.

The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the chorus sweep (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Width	Determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Waveform Shape selector	Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.
Spatial	Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
Mix	Sets the level balance between the dry and the wet signal. If Chorus is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Delay	Affects the frequency range of the modulation sweep by adjusting the initial delay time.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

Flanger

Flanger is a classic flanger effect with added stereo enhancement.

The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the flanger sweep (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Range Lo/Hi	Set the frequency boundaries for the flanger sweep.
Feedback	Determines the character of the flanger effect. Higher settings produce a more “metallic” sounding sweep.
Spatial	Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.
Mix	Sets the level balance between the dry and the wet signal. If Flanger is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Waveform Shape selector	Allows you to select the modulation waveform, altering the character of the flanger sweep. A sine and a triangle waveform are available.
Delay	Affects the frequency range of the modulation sweep by adjusting the initial delay time.
Manual knob	Allows you to change the sweep position manually when the Manual button is deactivated. The value range is from 0 to 100.
Manual button	Use this button to activate/deactivate the Manual function. If activated, the flanger sweep is static, i.e. no modulation takes place.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

Metalizer (not in Cubase LE)

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

The following parameters are available:

Parameter	Description
Feedback	The higher the value, the more "metallic" the sound.
Sharpness	Governs the character of the filter effect. The higher the value, the narrower the affected frequency area, producing a sharper sound and a more pronounced effect.
Tone	Governs the feedback frequency. The effect of this will be more noticeable with high Feedback settings.
On button	Turns filter modulation on and off. When turned off, Metalizer works as a static filter.
Mono button	When this is activated, the output of Metalizer is mono.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the modulation speed can be set freely with the Speed knob.
Sync button	The button above the Speed knob is used to switch tempo sync on (button lights up) or off.
Output slider	Sets the overall volume.
Mix slider	Sets the level balance between the dry and the wet signal. If Metalizer is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

Phaser

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

The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the phaser sweep (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Width	Determines the width of the modulation effect between higher and lower frequencies.
Feedback	Determines the character of the phaser effect. Higher settings produce a more pronounced effect.
Spatial	When using multi-channel audio, the Spatial parameter creates a 3-dimensional impression by delaying modulation in each channel.
Mix	Sets the level balance between the dry and the wet signal. If Phaser is used as a send effect, set this to the maximum level as you can control the dry/effect balance with the send.
Manual knob	Allows you to change the sweep position manually when the Manual button is deactivated. The value range is from 0 to 100.
Manual button	Use this button to activate/deactivate the Manual function. If activated, the flanger sweep is static, i.e. no modulation takes place.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

RingModulator (not in Cubase LE)

RingModulator can produce complex, bell-like enharmonic sounds. Ring modulators work by multiplying two audio signals. The ring modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals.

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

The following parameters are available:

Parameter	Description
Oscillator – LFO Amount	Controls how much the oscillator frequency is affected by the LFO.
Oscillator – Env. Amount	Controls how much the oscillator frequency is affected by the envelope (which is triggered by the input signal). Positive and negative values can be set, with center position representing no modulation. Left of center, a loud input signal will decrease the oscillator pitch, whereas right of center the oscillator pitch will increase when fed a loud input.
Oscillator – Waveform buttons	Allows you to select the oscillator waveform; square, sine, saw, or triangle.
Oscillator – Range slider	Determines the frequency range of the oscillator in Hz.
Oscillator – Frequency	Sets the oscillator frequency +/- 2 octaves within the selected range.
Oscillator – Roll-Off	Cuts high frequencies in the oscillator waveform, to soften the overall sound. This is best used when harmonically rich waveforms are selected (e.g. square or saw).
LFO – Speed	Sets the LFO speed.
LFO – Env. Amount	Controls how much the input signal level – via the envelope generator – affects the LFO speed. Positive and negative values can be set, at 0% no modulation is applied. With negative values, a loud input signal slows down the LFO, whereas positive values are used to speed it up at loud input signals.
LFO – Waveform	Allows you to select the LFO waveform; square, sine, saw, or triangle.
LFO – Invert Stereo	Inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.
Envelope Generator section – Attack and Decay	The Envelope Generator section controls how the input signal is converted to envelope data, which can then be used to control oscillator pitch and LFO speed. It has two main controls: Attack controls how fast the envelope output level rises in response to a rising input signal. Decay controls how fast the envelope output level falls in response to a falling input signal.
Lock L<R button	When this button is enabled, the L and R input signals are merged, and produce the same envelope output level for both oscillator channels. When disabled, each channel has its own envelope, which affects the two channels of the oscillator independently.
Output slider	Sets the overall volume.
Mix slider	Adjusts the mix between dry and processed signal.

Rotary (Cubase Elements only)

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

The following parameters are available:

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

Directing MIDI to the Rotary

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

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

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

Tranceformer (not in Cubase LE)

Tranceformer is a ring modulator effect, in which the incoming audio is ring modulated by an internal, variable frequency oscillator, producing new harmonics. A second oscillator can be used to modulate the frequency of the first oscillator, in sync with the Song tempo if needed.

The following parameters are available:

Parameter	Description
Waveform buttons	Allow you to select a pitch modulation waveform.
Tone	Sets the frequency (pitch) of the modulating oscillator (1 to 5000 Hz).
Depth	Governs the depth of the pitch modulation.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the modulation speed can be set freely with the Speed knob.
Sync button	The button above the Speed knob is used to switch tempo sync on (button lights up) or off.
On button	Turns modulation of the pitch parameter on or off.
Mono button	Governs whether the output is stereo or mono.
Output slider	Allows you to adjust the output level of the effect.
Mix slider	Sets the level balance between the dry and the wet signal.

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

Tremolo

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

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the modulation speed can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Depth	Governs the depth of the amplitude modulation.
Spatial	Adds a stereo effect to the modulation.
Output	Allows you to adjust the output volume.

Vibrato

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

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the modulation speed can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Depth	Governs the depth of the pitch modulation.
Spatial	Adds a stereo effect to the modulation.

Pitch Shift plug-ins

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

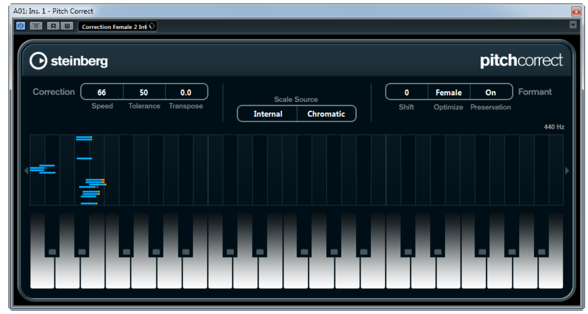
Octaver (not in Cubase LE)

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

The following parameters are available:

Parameter	Description
Direct	Adjusts the mix of the original signal and the generated voices. A value of 0 means only the generated and transposed signal is heard. By raising this value, more of the original signal is heard.
Octave 1	Adjusts the level of the signal that is generated one octave below the original pitch. A setting of 0 means that the voice is muted.
Octave 2	Adjusts the level of the signal that is generated two octaves below the original pitch. A setting of 0 means that the voice is muted.

Pitch Correct (Cubase Elements only)



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

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

The following parameters are available:

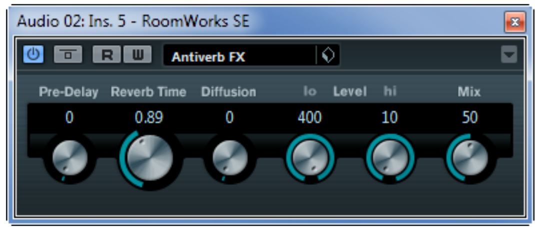
Parameter	Description
Correction – Speed	Determines the smoothness of the pitch change. Higher values cause the pitch shift to occur immediately. 100 is a very drastic setting that is designed mainly for special effects (e.g. the famous "Cher" effect).
Correction – Tolerance	Determines the sensitivity of analysis. A low Tolerance value lets Pitch Correct find pitch changes quickly. When the Tolerance value is high, pitch variations in the audio (e.g. vibrato) will not be immediately interpreted as note changes.
Correction – Transpose (-12 to 12)	With this parameter you can adjust (or "retune") the pitch of the incoming audio in semitone steps. You can set positive and negative values from -12 to 12. A value of zero means that the signal is not transposed.

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

Reverb plug-ins

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

RoomWorks SE



Roomworks SE is a high-quality reverb plug-in.

The following parameters are available:

Parameter	Description
Pre-Delay	Controls how much time passes before the reverb is applied. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.
Reverb Time	Allows you to set the reverb time in seconds.
Diffusion	Affects the character of the reverb tail. Higher values lead to more diffusion and a smoother sound, while lower values lead to a clearer sound.
Hi Level	Affects the decay time of high frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes high frequencies to decay quicker. Values above 100% cause high frequencies to decay more slowly than the mid-range frequencies.
Lo Level	Affects the decay time of low frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes low frequencies to decay quicker. Values above 100% cause low frequencies to decay more slowly than the mid-range frequencies.
Mix	Determines the blend of dry (unprocessed) signal to wet (processed) signal. When using RoomWorks SE inserted in an FX channel, you will most likely want to set this to 100% or use the wet only button.

Spatial + Panner plug-ins

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

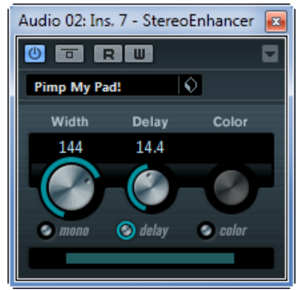
MonoToStereo (not in Cubase LE)

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

The following parameters are available:

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

StereoEnhancer (Cubase Elements only)



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

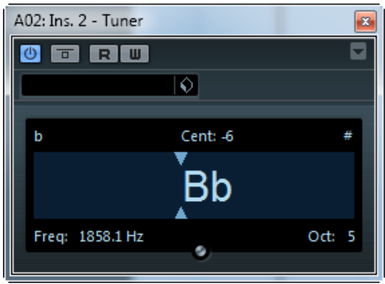
The following parameters are available:

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

Tools plug-ins

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

Tuner (not in Cubase LE)



This is a guitar tuner. Simply connect a guitar or other instrument to an audio input and select the Tuner as an insert effect (make sure you deactivate any other effect that alters pitch, like chorus or vibrato).

When you play a note, the pitch is shown in the middle of the display. In addition, the frequency in Hz is shown in the bottom left corner and the octave range in the bottom right corner.

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

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

  Repeat this procedure for each string.

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

2

The included VST instruments

Introduction

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

Groove Agent ONE (Cubase Elements only)

Groove Agent ONE is an easy-to-use sample-based MPC-style virtual drum machine for creating beats and reconstructing loops.

Audio samples can be associated with the Groove Agent ONE pads. Each pad is associated with a MIDI pitch, allowing you to trigger individual pads via MIDI notes.

To facilitate the creation of your own drum patterns, Groove Agent ONE provides a number of advanced functions.

Groups and pads

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

Groove Agent ONE provides up to 128 pads, organized in eight groups of 16 pads. You can switch between the different groups by clicking on the corresponding group buttons (labeled 1 to 8) above the pads. Each pad is mapped to a particular MIDI note (C-2 to G8, which equals 128 notes).

- The button of the active group is highlighted. If one or more pads of a group have samples mapped to them, an additional red frame is displayed around group buttons.
By default, group 3 is active when you open Groove Agent ONE.

Pad functions

- The pads show the associated MIDI note in the top right corner.
You can change the MIDI note by right-clicking it and selecting a different note from the pop-up menu.

- You can assign up to eight samples to a pad.
See “Drag & drop of audio material” on page 26.

- If one or more samples have been assigned to a pad, the name of the first of these samples is displayed at the bottom of the pad.
To change the name, right-click it, enter a new name and press [Enter]. This allows you, e.g., to indicate that more than one sample is mapped to this pad.

- To remove a sample assignment, click on the pad and drag the associated sample(s) to the trash icon in the LCD display to the left (see “Editing sounds” on page 27).
Note that the trash icon is found only on either the Voice, Filter or Amplifier pages.

- The pad status is indicated by different colors.
During playback, a pad lights up yellow for as long as a sample mapped to this pad is played back. When either the Voice, Filter or Amplifier button is activated in the Pad Edit section and you click on a pad, it turns light green to indicate that it is selected for editing. Unselected pads not playing back any samples are gray.

- To select multiple pads for sound editing, [Ctrl]/[Command]-click on the pads.
The pad that has been selected first lights up light green, the rest of the selected pads turn dark green (see “Editing sounds” on page 27).

- To mute or solo a pad, click the corresponding icon in the upper left corner of a pad.
The icon lights up to indicate that the pad is muted or soloed. If you solo a pad, all other pads are muted automatically. To unmute or unsolo the pad, click once more on the icon.

- You can drag a sample from one pad to another pad.
If the second pad already has a sample mapped to it, the sample assignment is swapped. Note that you can also swap the MIDI notes of the two pads by pressing [Shift] when dropping the sample.

- You can drag and drop samples between groups.
Click on a pad that has a sample mapped to it, keep the mouse button pressed and move the mouse pointer over the button of another group. When the pad display now changes to display the pads of the other group, drag and drop the sample on the desired pad.

Velocity

- The velocity is determined by where on the pad you click: it is lowest at the bottom of the pad and highest at the top.
- You can force all pads to a velocity value of 127 by activating the V-Max button in the Global section in the top right corner of the Groove Agent ONE panel.

Resetting pads

You can find a Reset button in the Global section in the top right corner of the Groove Agent ONE panel. It allows you to clear all pad assignments of the current instance of Groove Agent ONE.

As a safety precaution, the Reset button is locked by default. Clicking the Reset button when it is locked has no effect.

To unlock the Reset button, hold down the [Shift] key while clicking. The button color changes to red. When you click Reset now, all pad assignments are reset.

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

Drag & drop of audio material

Groove Agent ONE provides advanced drag & drop support. You can drag one or more samples at the same time from Cubase onto Groove Agent ONE. Samples can either be mapped to the same pad, or to different pads.

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

- MediaBay
- Project window
- Pool
- Sample Editor (regions)
- Audio Part Editor

Layering samples on the same pad

When you select between one and eight samples and drag them to Groove Agent ONE, dropping them onto a pad (or onto the Layer indicator – see below) automatically creates a corresponding number of layers for this pad.

Drag & drop to several pads

Rather than dropping several samples to the same pad, you can also let Groove Agent ONE distribute samples across the available pads in one or several groups. To do so, select the desired samples, drag them to the Groove Agent ONE window, press [Shift] and drop the samples onto a pad. The samples are mapped to the available pads, starting with the pad on which you initially dropped the samples, and then upwards according to the MIDI pitches of the pads.

How many samples can be dropped to several pads depends on the number of pads available in your current instance of Groove Agent ONE. If Groove Agent ONE cannot supply a sufficient number of free pads for the number of dropped samples, a dialog is displayed in which you can confirm or cancel the operation.

Replacing individual samples

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

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

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

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

Slicing a loop and triggering individual sounds via MIDI

Drag & drop to several pads has a number of uses. For example, it allows you to trigger individual sounds from an audio loop via MIDI. Proceed as follows:

1. Slice up a drum loop using the Sample Editor. Open the resulting audio part in the Audio Part Editor and press [Ctrl]/[Command]-[A] to select all audio events.
See the Operation Manual for details about slicing.

2. In the Audio Part Editor, click on one of the selected events and drag it to the Groove Agent ONE window.

3. Press the [Shift] key.

4. Point the mouse pointer at an empty pad and let go of the mouse button.
The individual samples from the audio part are now mapped to the available pads of Groove Agent ONE.

Now look at the Exchange section (to the left of the pads): the MIDI Export pad (the field displaying a double arrow) at the bottom of the section is lit. When mapping several samples to several pads, Groove Agent ONE creates a MIDI file containing all MIDI information to trigger these pads, and maps this file to the MIDI Export pad.

5. Drag this MIDI file from the MIDI Export pad onto the Cubase Project window.
Dropping the file onto the Project window creates a new MIDI track. You can also drop the MIDI file to an existing MIDI or instrument track.

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

Saving the Groove Agent ONE setup

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

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

Saving plug-in presets

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

1. At the top of the Groove Agent ONE window, click the VST Sound button to the right of the Presets pop-up menu and select “Save Preset”.
The Save Preset dialog opens.

2. Enter a name for the new preset and click OK.
The preset is saved in the User Content folder on your system.

Loading plug-in presets

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

1. At the top of the Groove Agent ONE window, click the VST Sound button and select “Load Preset” from the pop-up menu.
The Presets browser opens.

2. The Presets browser shows all presets it finds in the VST 3 Presets folder for Groove Agent ONE. Double-click the desired preset to load it.
The Presets browser is closed and the preset is loaded into Groove Agent ONE.

- When a sample belonging to a preset cannot be found, Groove Agent ONE prompts you to locate the missing files. You can click either Ignore to skip this message, click Locate File to navigate to a specific folder containing the missing file(s), or click Search Folder to browse a specific folder and any subfolders that might contain the missing file(s).

Saving a GAK archive

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

1. Set up Groove Agent ONE the way you want it.

2. In the Exchange section, click the Export button.
The “Export Groove Agent ONE kit” dialog opens in which you can specify a location and a name for the new archive.

3. Click Save.
The archive is created and the dialog is closed.

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

Loading a GAK archive

To load the GAK file, proceed as follows:

1. In the Exchange section, click the Import button.
Navigate to the GAK file.

2. Click Open.
The saved settings and all samples are imported into Groove Agent ONE.

Editing sounds

All sound editing functions can be found in and below the LCD display in the left half of the panel.

The LCD display can show four different sound editing pages, selected by clicking one of the four buttons in the Pad Edit section.

The information on the Play page refers to this instance of Groove Agent ONE as a whole. When the Play button is activated, the LCD display shows the name of the loaded VST preset and information on the number of samples and pads used by this instance of Groove Agent ONE. The Size parameter indicates the amount of RAM occupied by the currently loaded samples. The Polyphony counter shows the number of pads currently playing.

- Click on a pad for sound editing.

It turns light green and the display shows its sample parameters.

- To adjust a parameter, either use one of the quick controls below the display, or click on the parameter in the display and adjust it by dragging your mouse.

- You can select multiple pads for sound editing by [Ctrl]/[Command]-clicking on them, and adjust their parameters in one go with the quick controls below the display.

The first selected pad lights up light green, all other selected pads turn dark green. The display shows the parameters of the first selected pad.

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

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

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

Parameter	Description
Brightness slider	Use the little slider at the very top of the LCD display to set the display brightness.
VST Preset	The name of a loaded VST Preset is displayed in the top left of the LCD display.
Sample/Pad	The name of the sample (and the pad to which it is assigned).
Trash icon	You can remove the current sample assignment by clicking on a pad or on the Layer indicator (see below) and dragging it onto the trash icon.
MIDI input off	When the MIDI symbol button in the top right corner of the LCD display is activated, the LCD display shows the waveform and parameter values of the currently playing sample. When this button is deactivated, the display shows only the data for the currently edit selected sample.
Layer indicator	The long bar near the top of the LCD display shows the active layer for the current pad. If more than one layer exist for the selected pad, the bar is divided accordingly. You can drag the dividing line between layers to change the velocity ranges of the layers. You can drag a new sample from the MediaBay and drop it directly onto the Layer indicator bar (this is the same as dropping a sample on a pad). You can drag layers to a different position on the bar.
Layer number	The layer number indicates which is the active layer of the current pad.

Parameter	Description
Sample	This is the name of the sample file.
Velocity	Here you can specify a velocity range for the current layer.
Coarse	Here you can tune the sample by up to ±12 semitones.
Fine	This parameter lets you fine-tune the sample by up to ±100 cents.
Volume	Sets the sample volume.
Waveform display	The waveform of the current sample.
s/e locators in waveform display	You can define the sample start and end points by dragging the s and e locators in the waveform display. When you click on a locator and press [Ctrl], this will zoom in on the waveform and center the display around the locator. Note that the locators automatically snap to zero crossings.

Depending on the selected page (Play, Voice, Filter, Amplifier), up to six quick controls with different pad-specific parameter assignments are displayed.

Play parameters

The parameter controls on the Play page are copies of the parameters on the Voice, Filter, and Amplifier pages.

The row of parameter controls below the LCD display shows six parameters:

Parameter	Description
Volume	The volume of the pad currently selected for editing.
Pan	The panorama setting of the pad currently selected for editing.
Coarse	Use this control to tune the pad by up to ±12 semitones.
Cutoff	Sets the filter cutoff frequency.
Q	Sets the filter resonance.
Output	Groove Agent ONE provides up to 16 stereo outputs. You can route pads to individual outputs using this control.

Voice parameters

The row of parameter controls below the LCD display shows six parameters:

Parameter	Description
Mode	Here you can reverse the currently selected sample so that you hear it backwards.
Coarse	Use this control to tune the pad by up to ±12 semitones.
Fine	Use this control to fine-tune the pad by up to ±100 cents.

Parameter	Description
Mute Gr.	With this control you can assign a pad to one of eight mute groups. Pads within a mute group never play back simultaneously. New notes cancel previous notes.
Tr. Mode	The sample of the currently selected pad is played either from start to finish (One Shot) or only for as long as you hold the mouse button/key (Key Hold). Key Hold can also be determined by the length of the corresponding MIDI note on your track.
Output	Groove Agent ONE provides up to 16 stereo outputs. You can route pads to individual outputs using this control. See the Operation Manual for information on how to use multitimbral instruments in Cubase.

Filter parameters

The row of parameter controls below the LCD display shows four parameters used to edit the Groove Agent ONE filter:

Parameter	Description
Type	Sets the filter type: low-pass (LP), high-pass (HP) or band-pass (BP). When you set this knob to OFF, the settings on this editing page have no effect.
Cutoff	Sets the filter cutoff frequency.
Q	Sets the filter resonance.
Mod	This parameter determines the influence that velocity has on the cutoff frequency. When set to 0%, the setting has no effect. When set to any other value, the cutoff frequency changes depending on the velocity.

Amplifier parameters

The row of parameter controls below the LCD display shows six parameters:

Parameter	Description
Volume	The volume of the pad currently selected for editing.
Pan	The panorama setting of the pad currently selected for editing.
Attack	Controls the amplifier envelope attack time.
Release	Controls the amplifier envelope release time. Reduce the release time to shorten the decay of sounds played in one-shot mode.
Amp Mod	This parameter determines the influence that velocity has on the pad volume setting. When set to 100%, the pad sounds louder the higher the velocity. When set to 0%, velocity has no effect on the pad volume.
Attack Mod	This parameter determines the influence that velocity has on the Attack setting. When set to 0%, velocity has no effect on the attack. When set to 100% and playing a pad with high velocity, the Attack time is increased by 50%. The higher the Attack Mod setting, the longer the additional attack time for a pad.

Master volume

In the Master section in the lower left of the Groove Agent ONE panel you can find a master volume slider that sets the output volume of the instrument.

The Exchange section

This section is used to import or export data to/from Groove Agent ONE.

Importing MPC files

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

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

The MIDI Export pad is described in detail in the section "Slicing a loop and triggering individual sounds via MIDI" on page 26.

The function of the Export button is described in detail in the section "Saving a GAK archive" on page 27.

Automation of Groove Agent ONE parameters

When opening an automation subtrack for a track that uses Groove Agent ONE, you can select the following plug-in parameters from the Add Parameters dialog:

- Volume
- Pan
- Mute
- Cutoff
- Resonance

These parameters are available for the pads C1 to B4.

HALion Sonic SE

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

Prologue (Cubase Elements only)

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

- Multimode filter

Variable slope low pass and high pass, plus band pass and notch filter modes – see “About the filter types” on page 34.

- Three oscillators, each with 4 standard waveforms plus an assortment of specialized waveforms.

See “Selecting Waveforms” on page 30.

- Frequency modulation.

See “About frequency modulation” on page 32.

- Ring Modulation.

See “Ring modulation” on page 33.

- Built-in effects.

See “Effects (EFX) page” on page 38.

- Prologue receives MIDI in Omni mode (on all MIDI channels).

You do not have to select a MIDI channel to direct MIDI to the Prologue.

Sound parameters

Oscillator section

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

Selecting Waveforms

Each oscillator has a number of waveforms which are selectable by clicking on the waveform name in the box located in each oscillator section.

The following waveforms are available:

Waveform	Description
Sawtooth	This waveform contains all harmonics and produces a bright and rich sound.
Parabolic	This can be described as a “rounded” sawtooth waveform, producing a softer timbre.
Square	Square waveforms only contain odd number harmonics, which produces a distinct, hollow sound.
Triangle	The triangle waveform generates only a few harmonics, spaced at odd harmonic numbers, which produces a slightly hollow sound.

Waveform	Description
Sine	The sine wave is the simplest possible waveform, with no harmonics (overtones). The sine wave produces a neutral, soft timbre.
Formant 1–12	Formant waveforms emphasizes certain frequency bands. Like the human voice, musical instruments have a fixed set of formants, which give it a unique, recognizable tonal color or timbre, regardless of pitch.
Vocal 1–7	These are also formant waveforms, but specifically vocal-oriented. Vowel sounds (A/E/I/O/U) are among the waveforms found in this category.
Partial 1–7	Partials, also called harmonics or overtones, are a series of tones which accompany the prime tone (fundamental). These waveforms can be described as producing intervals with two or more frequencies heard simultaneously with equal strength.
Reso Pulse 1–12	This waveform category begins with a complex waveform (Reso Pulse 1), that emphasizes the fundamental frequency (prime). For each consecutive waveform in this category, the next harmonic in the harmonic series is emphasized.
Slope 1–12	This waveform category begins with a complex waveform (Slope 1), with gradually decreasing harmonic complexity the higher the number selected. Slope 12 produces a sine wave (no harmonics).
Neg Slope 1–9	This category also begins with a complex waveform (NegSlope 1), but with gradually decreasing low frequency content the higher the number selected.

- To hear the signal generated by the oscillator(s), the corresponding Osc controls in the oscillator sections must be turned clockwise to a suitable value.

OSC 1 parameters

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

Parameter	Description
Osc 1 (0–100)	This controls the output level of the oscillator.
Coarse (±48 semitones)	This determines the base pitch used by all oscillators.
Fine (±50 cent)	Fine tunes the oscillator pitch in cent increments (100th of a semitone). This also affects all oscillators.

Parameter	Description
Wave Mod (±50)	This parameter is only active if the Wave Mod button is activated beside the waveform selection box. Wave modulation works by adding a phase-shifted copy of the oscillator output to itself, which produces waveform variations. For example if a sawtooth waveform is used, activating WM produces a pulse waveform. By modulating the WM parameter with for example an LFO, classic PWM (pulse width modulation) is produced. However, wave modulation can be applied to any waveform.
Phase button (On/Off)	When Phase synchronization is activated, all oscillators restart their waveform cycles with every note played. With Phase deactivated, the oscillators generate a waveform cycle continuously, which produces slight variations when playing as each note starts from a random phase in the cycle, adding warmth to the sound. But when synthesizing bass sounds or drum sounds, it is usually desired that the attack of every note played sounds the same, so for these purposes activate Phase sync. Phase sync also affects the noise generator.
Tracking button (On/Off)	When Tracking is activated, the oscillator pitch tracks the notes played on the keyboard. If Tracking is deactivated, the oscillator pitch remains constant, regardless of what note is played.
Wave Mod button (On/Off)	This switches wave modulation on or off.
Waveform pop-up menu (see “Selecting Waveforms” on page 30)	Sets the basic waveform for the oscillator.

OSC 2 parameters

Oscillator 2 has the following parameters:

Parameter	Description
Osc 2 (0–100)	This controls the output level of the oscillator.
Coarse (±48 semitones)	This determines the coarse pitch for Osc 2. If FM is enabled, this determines frequency ratio of the oscillator regarding Osc 1.
Fine (±50 cent)	Fine tunes the oscillator pitch in cent increments (100th of a semitone). If FM is enabled, this determines the frequency ratio of the oscillator regarding Osc 1.
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.

Parameter	Description
Ratio (1–16)	This parameter (which is only active if the Freq Mod button is activated) adjusts the amount of frequency modulation applied to oscillator 2, see "About frequency modulation" on page 32. Is normally referred to as FM index.
Sync button (On/Off)	When Sync is activated, Osc 2 is slaved to Osc 1. This means that every time Osc 1 completes its cycle, Osc 2 is forced to reset (start its cycle from the beginning). This produces a characteristic sound, suitable for lead playing. Osc 1 determines the pitch, and varying the pitch of Osc 2 produces changes in timbre. For classic sync sounds, try modulating the pitch of Osc 2 with an envelope or an LFO. The Osc 2 pitch should also be set higher than the pitch of Osc 1.
Tracking button (On/Off)	When Tracking is activated, the oscillator pitch tracks the notes played on the keyboard. If Tracking is deactivated, the oscillator pitch remains constant, regardless of what note is played.
Freq Mod button (On/Off)	This switches frequency modulation on or off.
Wave Mod button (On/Off)	This switches wave modulation on or off.
Waveform pop-up menu (see "Selecting Waveforms" on page 30)	Sets the basic waveform for the oscillator.

OSC 3 parameters

Oscillator 3 has the following parameters:

Parameter	Description
Osc 3 (0–100)	This controls the output level of the oscillator.
Coarse (±48 semitones)	This determines the coarse pitch for Osc 3. If FM is enabled, this determines the frequency ratio of the oscillator regarding Osc 1/2.
Fine (±50 cent)	Fine tunes the oscillator pitch in cent increments (100th of a semitone). If FM is enabled, this determines the frequency ratio of the oscillator regarding Osc 1/2.
Ratio (1–16)	This parameter (which is only active if the Freq Mod button is activated) adjusts the amount of frequency modulation applied to oscillator 3, see "About frequency modulation" on page 32. Is normally referred to as FM index.
Sync button (On/Off)	When Sync is activated, Osc 3 is slaved to Osc 1. This means that every time Osc 1 completes its cycle, Osc 3 is forced to reset (start its cycle from the beginning). This produces a characteristic sound, suitable for lead playing. Osc 1 determines the pitch, and varying the pitch of Osc 3 produces changes in timbre. For classic sync sounds, try modulating the pitch of Osc 3 with an envelope or an LFO. The Osc 3 pitch should also be set higher than the pitch of Osc 1.
Tracking button (On/Off)	When Tracking is activated, the oscillator pitch tracks the notes played on the keyboard. If Tracking is deactivated, the oscillator pitch remains constant, regardless of what note is played.
Freq Mod button (On/Off)	This switches frequency modulation on or off.
Wave Mod button (On/Off)	This switches wave modulation on or off.
Waveform pop-up menu (see "Selecting Waveforms" on page 30)	Sets the basic waveform for the oscillator.

About frequency modulation

Frequency modulation or FM means that the frequency of one oscillator (called the carrier) is modulated by the frequency of another oscillator (called the modulator).

- In Prologue, Osc 1 is the modulator, and Osc 2 and 3 are carriers.

Osc 2 could be said to be both carrier and modulator as if Freq Mod is applied to Osc 2 it is modulated by Osc 3. If Osc 2 also uses frequency modulation, Osc 3 is modulated by both Osc 1 and Osc 2.

- The "pure" sound of frequency modulation is output through the modulator oscillator(s).

This means that you should turn off the Osc 1 output when using frequency modulation.

- The Freq Mod button switches frequency modulation on or off.
- The Ratio parameter determines the amount of frequency modulation.

Portamento

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

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

Ring modulation

Ring modulators multiply two audio signals. The ring-modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals. In Prologue, Osc 1 is multiplied with Osc 2 to produce sum and difference frequencies. Ring modulation is often used to create bell-like sounds.

- To hear the ring modulation, turn down the output level for Osc 1 and 2, and turn up the “R.Mod” level all the way.
- If Osc 1 and 2 are tuned to the same frequency, and no modulation is applied to the Osc 2 pitch, nothing much happens.

However, if you change the pitch of Osc 2, drastic changes in timbre can be heard. If the oscillators are tuned to a harmonic interval such as a fifth or octave, the ring modulated output sounds harmonic, other intervals produce inharmonious, complex timbres.

- Deactivate Oscillator Sync when using ring modulation.

Noise generator

A noise generator generates noise (all frequencies at equal levels). Applications include simulating drum sounds and breath sounds for wind instruments.

- To hear only the sound of the noise generator, turn down the output level for the oscillators, and turn up the Noise parameter.
- The noise generator level is routed to Envelope 1 by default.

See “Envelope page” on page 36 for a description of the Envelope generators.

Filter section

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

Parameter	Description
Filter type	Sets the filter type to either low pass, high pass, band pass or notch. The filter types are described in the table below.
Cutoff	This knob controls the filter frequency or “cutoff”. If a low pass filter is used, it could be said to control the opening and closing of the filter, producing the classic “sweeping” synthesizer sound. How this parameter operates is governed by the filter type mode (see the table below).
Emphasis	This is the resonance control for the filter. For low pass and high pass filters, raising the Emphasis value emphasizes the frequencies around the set cutoff frequency. This produces a generally thinner sound, but with a sharper, more pronounced cutoff sweep. The higher the filter Emphasis value, the more resonant the sound becomes until it starts to ring (self-oscillate), generating a distinct pitch. For Band pass or Notch filters, the Emphasis setting adjusts the width of the band. When you raise the value, the band where frequencies are let through (Band pass), or cut (Notch) becomes narrower.
Drive	This can be used to adjust the filter input level. Levels above 0 dB gradually introduce a soft distortion of the input signal, and a decrease of the filter resonance.
Shift	Internally, each filter consists of two or more “subfilters” connected in series. This parameter shifts the cutoff frequency of the subfilters. The result depends on the selected filter type: For Low pass and High pass filter types it changes the filter slope. For Band pass and Notch filter types it changes the bandwidth. The Shift parameter has no effect if either the 12 dB LP or 12 dB HP filter type is selected.
Tracking	If this parameter is set to values over the 12 o’clock position, the filter cutoff frequency increases the further up on the keyboard you play. Negative values invert this relationship. If the Tracking parameter is set fully clockwise, the cutoff frequency tracks the keyboard by a semitone per key.

About the filter types

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

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

Master Volume and Pan

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

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

Modulation and controllers

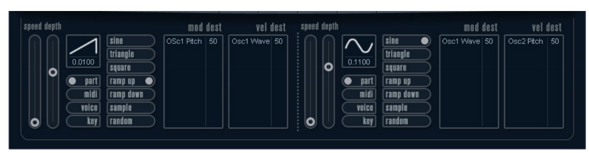
The lower half of the control panel displays the various modulation and controller assignment pages available as well as the effect page. You switch between these pages using the buttons below the Filter section.

The following pages are available:

- The LFO page has two low frequency oscillators (LFOs) for modulating parameters – see below.
- The Envelope page contains the four Envelope generators which can be assigned to control parameters – see “Envelope page” on page 36.
- The Event page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments – see “Event page” on page 38.
- The Effect page has three separate effect types available; Distortion, Delay and Modulation – see “Effects (EFX) page” on page 38.

LFO page

The LFO page is opened by clicking the LFO button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for two independent LFOs.



Depending on the currently selected preset, there may already be modulation destinations assigned, in which case these are listed in the “Mod Dest” box for each LFO – see “Assigning LFO modulation destinations” on page 35. 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 desired.

The two LFOs have identical parameters:

Parameter	Description
Speed	This governs the rate of the LFO. If MIDI Sync is activated (see below), the available rate values are selectable as note values, e.g. beat increments of the sequencer tempo in Cubase.
Depth	This controls the amount of modulation applied by the LFO. If set to zero, no modulation is applied.
Waveform	This sets the LFO waveform.
Sync mode (Part/MIDI/Voice/Key)	This sets the sync mode for the LFO. See below for a description.

About the sync modes

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

Parameter	Description
Part	In this mode, the LFO cycle is free running and affects all the voices in sync. "Free running" means that the LFO cycles continuously, and does not reset when a note is played.
MIDI	In this mode the LFO rate is synced in various beat increments to MIDI clock.
Voice	In this mode each voice in the Part has its own independent LFO cycle (the LFO is polyphonic). These cycles are also free running – each key down starts anywhere in the LFO cycle phase.
Key	Same as Voice except that it is not free running – for each key down the LFO cycle starts over.

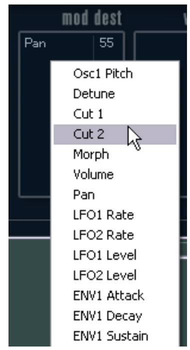
About the waveforms

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

Assigning LFO modulation destinations

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

1. Click in the "Mod Dest" box for one of the LFOs.
A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.



2. Select a destination, e.g. Filter Cut Off.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

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

3. Select a suitable LFO Waveform, Speed, Depth, and Sync mode.
You should now hear the filter cutoff being modulated by the LFO.

4. Using the same basic method, you can add any number of modulation destinations for the LFO.
They are all listed in the "Mod Dest" box.

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

Assigning LFO velocity destinations

You can also assign LFO modulation that is velocity controlled (i.e. governed by how hard or soft you strike a key). Proceed as follows:

1. Click in the “Vel Dest” box for one of the LFOs.
A pop-up menu appears in which all possible velocity destinations are shown.

2. Select a destination.
The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

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

3. Using the same basic method, you can add any number of velocity destinations for the LFO.
They are all listed in the “Vel Dest” box.

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

LFO modulation velocity control – an example:

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

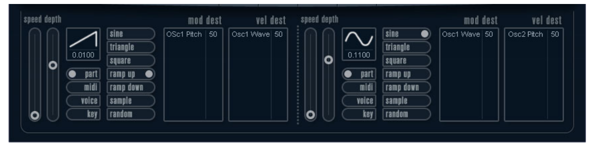
- The harder you strike the key, the more the filter cutoff parameter is modulated by the LFO.

- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the filter cutoff is modulated by the LFO.

Envelope page

The Envelope page is opened by clicking the ENV button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for the four independent envelope generators.

Envelope generators govern how a parameter value changes when a key is pressed, when a key is held and finally when a key is released.



On the Envelope page, the parameters for one of the four envelope generators is shown at a time.

- You switch between the four envelopes in the section to the left.
Clicking on either of the four mini curve displays 1 to 4 selects it and displays the corresponding envelope parameters to the right. The mini curve displays also reflect the envelope settings for each corresponding envelope.

- Envelope generators have four parameters; Attack, Decay, Sustain, and Release (ADSR).
See below for a description of these.

- You can set envelope parameters in two ways; either by using the sliders or by click-dragging the curve in the Envelope curve display.
You can also do this in the mini curve displays.

- By default Envelope 1 is assigned to the master volume, and therefore acts as an amplitude envelope. The amplitude envelope is used to adjust how the volume of the sound changes from the time you press a key until the key is released.
If no amplitude envelope were assigned, there would be no output.

The Envelope parameters are as follows:

Attack

The attack phase is the time it takes from zero to the maximum value. How long this takes is governed by the Attack setting. If the Attack is set to “0”, the maximum value is reached instantly. If this value is raised, it takes time before the maximum value is reached. Range is from 0.0 milliseconds to 91.1 seconds.

Decay

After the maximum value has been reached, the value starts to drop. How long this takes is governed by the Decay time parameter. The Decay time has no effect if the Sustain parameter is set to maximum. Range is from 0.0 milliseconds to 91.1 seconds.

Sustain

The Sustain parameter determines the level the envelope rests at after the Decay phase. Note that Sustain represents a level, whereas the other envelope parameters represent times. Range is from 0 to 100.

Release

Release determines the time it takes for the value to fall back to zero after releasing the key. Range is from 0.0 milliseconds to 91.1 seconds.

Punch

When Punch is activated, the start of the decay phase is delayed by a few milliseconds (i.e. the envelope remains at the top level for a moment before moving on to the decay phase). The result is a punchier attack similar to a compressor effect. This effect is more pronounced with short attack and decay times.

Retrigger

When Retrigger is activated, the envelope re-triggers each time you play a new note. However, with certain textures/ pad sounds and a limited number of voices it is recommended to leave the button deactivated, due to click noises that might occur, when the envelope is ended up abruptly. This is caused by the incoming re-trigger that forces the envelope to start over again.

Assigning Envelope modulation destinations

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

1. Click in the "Mod Dest" box for one of the Envelopes.
A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, e.g. Filter Cut Off.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

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

3. Select a suitable envelope curve for the modulation.
You should now hear the filter cutoff being modulated by the envelope as you play.

4. Using the same basic method, you can add any number of modulation destinations for the envelope.
They are all listed in the "Mod Dest" box.

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

Assigning Envelope velocity destinations

You can also assign Envelope modulation that is velocity controlled (i.e. governed by how hard or soft you strike a key). Proceed as follows:

1. Click in the "Vel Dest" box for one of the envelopes.
A pop-up menu appears in which all possible velocity destinations are shown.

2. Select a destination.
The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

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

3. Using the same basic method, you can add any number of velocity destinations for the Envelope.
They are all listed in the "Vel Dest" box.

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

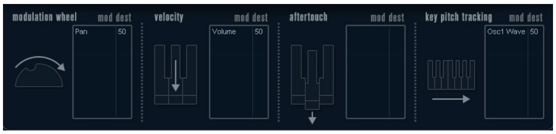
Envelope modulation velocity control – an example:

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

- The harder you strike the key, the more the filter cutoff parameter is modulated by the Envelope.
- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the filter cutoff is modulated by the Envelope.

Event page

The Event page is opened by clicking the EVENT button at the top of the lower half of the control panel. This page contains the most common MIDI controllers and their respective assignments.



The following controllers are available:

Controller	Description
Modulation Wheel	The modulation wheel on your keyboard can be used to modulate parameters.
Velocity	Velocity is used to control parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder.
Aftertouch	Aftertouch, or channel pressure, is MIDI data sent when pressure is applied to a keyboard after the key has been struck, and while it is being held down or sustained. Aftertouch is often routed to control filter cutoff, volume, and other parameters to add expression. Most (but not all) MIDI keyboards send Aftertouch.
Key Pitch Tracking	This can change parameter values linearly according to where on the keyboard you play.

To assign any of these controllers to one or several parameters, proceed as follows:

1. Click in the “Mod Dest” box for one of the controllers.
A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination.
The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount when the controller is at its full range.

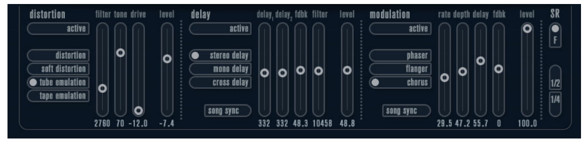
- You can set positive and negative modulation values by clicking on the value in the list, typing in a new value and pressing the [Enter] key.
To enter negative values type a minus sign followed by the value.

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

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

Effects (EFX) page

This page features three separate effect units: Distortion, Delay and Modulation (Phaser/Flanger/Chorus). The Effect page is opened by clicking the EFX button at the top of the lower half of the control panel.



- Each separate effect section is laid out with a row of buttons that determine the effect type or characteristic and a row of sliders for making parameter settings.
- To activate an effect, click the “Active” button so that a dot appears.
Clicking again deactivates the effect.

Distortion

You can select between 4 basic distortion characteristics:

- Distortion provides hard clipping distortion.
- Soft Distortion provides soft clipping distortion.
- Tape Emulation produces distortion similar to magnetic tape saturation.
- Tube Emulation produces distortion similar to valve amplifiers.

The parameters are as follows:

Parameter	Description
Filter	This parameter sets the crossover frequency of the distortion filter. The distortion filter consists of a low pass filter and a high pass filter with a cutoff frequency equal to the crossover frequency.
Tone	This parameter controls the relative amount of low pass and high-pass filtered signal.
Drive	Sets the amount of distortion by amplifying the input signal.
Level	This controls the output level of the effect.

Delay

You can select between 3 basic delay characteristics:

- Stereo Delay has two separate delay lines panned left and right.
- In Mono Delay the two delay lines are connected in series for monophonic dual tap delay effects.
- In Cross Delay the delayed sound bounces between the stereo channels.

The parameters are as follows:

Parameter	Description
Song Sync	This switches tempo sync of the delay times on or off.
Delay 1	Sets the delay time ranging from 0ms to 728ms. If MIDI sync is activated the range is from 1/32 to 1/1; straight, triplet or dotted.
Delay 2	Same as Delay 1.
Feedback	This controls the decay of the delays. With higher settings the echoes repeat longer.
Filter	A low pass filter is built into the feedback loop of the delay. This parameter controls the cutoff frequency of this feedback filter. Low settings result in successive echoes sounding darker.
Level	This controls the output level of the effect.

Modulation

You can select between 3 basic modulation characteristics:

- The Phaser uses an 8-pole allpass filter to produce the classic phasing effect.
- The Flanger is composed of two independent delay lines with feedback for the left and the right channel respectively. The delay time of both delays is modulated by one LFO with adjustable frequency.
- Chorus produces a rich chorus effect with 4 delays modulated by four independent LFOs.

The parameters are as follows:

Parameter	Description
Song Sync	This switches tempo sync of the Rate parameter on or off.
Rate	Sets the rate of the LFOs modulating the delay time. If Song Sync is activated the rate is synced to various beat increments.
Depth	This parameter controls the depth of the delay time modulation.
Delay	This parameter sets the delay time of the four delay lines.
Feedback	The feedback parameter controls the amount of positive or negative feedback for all four delay lines. The adjustable range is from -1 to 1.
Level	This controls the output level of the effect.

SR parameters

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

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

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

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