Audio Effects and VST Instruments

Cubase SE 3

MUSIC CREATION AND PRODUCTION SYSTEM

Mac

PC VERSION

steinberg

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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 SE, 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.

Delay plug-ins

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

DoubleDelay

This effect provides two separate delays that can be either tempo based or use freely specified delay time settings. Cubase SE automatically provides the plug-in with the tempo currently used in the project.

The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If DoubleDelay is used as a send effect, this should be set to maximum (100%) as you can control the dry/effect balance with the send.
Tempo sync on/off	The buttons above the two Delay Time knobs are used to turn tempo sync on or off for the respective delay. If set to off the delay time can be set freely with the Delay Time knobs, without sync to tempo.
Delay Time 1	This is where you specify the base note value for the delay if tempo sync is on (1/1 - 1/32, straight, triplet or dotted). If tempo sync is off, it sets the delay time in milliseconds.
Delay Time 2	As above.
Feedback	This sets the number of repeats for both delays.
Tempo Sync 1	The note value multiplier (x1 to x10) for the first delay unit.
Tempo Sync 2	As above, but for the second delay unit.

Parameter	Description
Pan1	This sets the stereo position for the first delay.
Pan2	This sets the stereo position for the second delay.

You can also change parameters in the graphic display window. This works as follows:

- If tempo sync is on, you can set the Tempo Sync 1 parameter by dragging the light blue handle left and right.
  When tempo sync is off, this sets the Delay Time 1 parameter.
- You can set the Pan 1 parameter by dragging the light blue handle up and down.
- The dark blue handle works in the same way but for the corresponding second delay parameters.

ModDelay

This is a delay effect that can either be tempo-based or use freely specified delay time settings. The delay repeats can also be modulated.

The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If ModDelay is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Tempo sync on/off	The button above the Delay Time knob is used to turn tempo sync on or off. If set to off the delay time can be set freely with the Delay Time knob, without sync to tempo.
Feedback	This sets the number of repeats for the delay.
Delay Time	This is where you specify the base note value for the delay if tempo sync is on (1/1 - 1/32, straight, triplet or dotted). If tempo sync is off, it sets the delay time in milliseconds.
Tempo Sync knob	This is the note value multiplier (x1 to x10) for the delay when tempo sync is used.
DelayMod.	This controls the pitch modulation rate for the delay effect.

Distortion plug-ins

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

DaTube

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

The parameters are as follows:

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	This 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

The Distortion effect plug-in is capable of producing anything from a soft "crunch" to all-out distortion. There is a selection of factory presets available. Note that these presets are not stored parameter settings, but different basic distortion algorithms. The basic characters of the distortion preset "models" are indicated by their names. The parameters are as follows:

Parameter	Values	Description
Input	-24dB to 0dB	Sets the Input level.
Output	-24dB to 0dB	Sets the Output level. As distortion generates harmonics, it increases the level of the processed signal. You can use the Output fader to compensate for the level increase.
Shapes	Linear, Non-linear 1, Non-linear 2	The Shape parameter determines how much the input signal is affected by the distortion effect. Non-linear 2 will produce the strongest distortion.
Contour	0-100%	This is a selective low-pass filter, altering the tonal quality of the distortion.
Drive	0-100%	Governs the amount of distortion.
Factory Presets	Soft, Crunchy, Dirty, Wracky, Evil	Select one of five presets, which can be used as they are, or as a basis for further "tweaking".

Overdrive

Overdrive is a distortion-type effect, emulating the sound of a guitar amplifier. A selection of factory styles is available. Note that these are not stored parameter settings, but different basic overdrive algorithms, with the style names indicating the basic character of each algorithm.

The parameters are as follows:

Parameter	Description
Input	Sets the input level.
Output	Sets the output level. As overdrive generates harmonics, it increases the level of the processed signal. You can use the Output fader to compensate for the level increase.
Speaker simulation	When this is activated, the effect simulates the sound of a speaker cabinet.
Factory Styles	Select one of six presets, which can be used as they are or as a basis for further “tweaking”.
Bass	Tone control for the low frequencies.
Mid	Tone control for the mid frequencies.
Hi	Tone control for the high frequencies.
Drive	Governs the amount of overdrive. You can also adjust this by clicking and dragging in the display.

Dynamics plug-ins

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

Dynamics

Dynamics is an advanced dynamics processor. It combines three separate processors: AutoGate, 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 by clicking on their labels. Activated processors have highlighted labels.

The AutoGate section

Gating, or noise gating, is a method of dynamic processing that silences audio signals below a certain set threshold level. As soon as the signal level exceeds the set threshold, the gate opens to let the signal through. AutoGate offers all the features of a standard noise gate, plus some very useful additional features, such as auto-calibration of the threshold setting, a look-ahead predict function, and frequency selective triggering.

The available parameters are as follows:

Parameter	Values	Description
Threshold	-60 - 0dB	This setting determines the level where AutoGate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold will close the gate.
Attack	0,1 -100 ms or "Predict mode"	This parameter sets the time it takes for the gate to open after being triggered. If the Predict button is activated, it will ensure that the gate will already be open when a signal above the threshold level is played back. AutoGate manages this by "looking ahead" in the audio material, checking for signals loud enough to pass the gate.
Hold	0 - 1000 ms	This determines how long the gate stays open after the signal drops below the threshold level.
Release	10 - 1000 ms or "Auto"	This parameter sets the amount of time it takes for the gate to close (after the set hold time). If the "Auto" button is activated, AutoGate will find an optimal release setting, depending on the audio program material.

Trigger Frequency Range

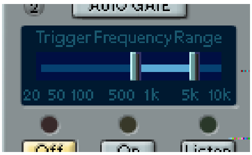
AutoGate has a feature that allows the gate to be triggered only by signals within a specified frequency range. This is a most useful feature because it lets you filter out parts of the signal that might otherwise trigger the gate in places you don't want it to, thus allowing more control over the gate function. The Trigger Frequency Range function is set using the control in the upper part of the AutoGate panel, and the buttons located below it.

The basic operation of the Trigger Frequency Range function is as follows:

1. While playing back audio, click the "Listen" button.
   You will now monitor the audio signal, and the gate will be bypassed.

2. While listening, drag the two handles in the Trigger Frequency Range display to set the frequency range you want to use to trigger the gate.
   You will hear the audio being filtered as you move the handles.

- Dragging the left handle to the right will progressively cut frequencies starting from the low end of the frequency spectrum.
- Dragging the right handle to the left will progressively cut frequencies starting from the high end of the frequency spectrum.



The frequency range between the two handles will be used to trigger the gate.

3. After setting the frequency range, click the "On" button.
   AutoGate will now use the selected frequency range as the trigger input.

4. To disable the Trigger Frequency Range function, click the "Off" button.
   AutoGate will now use the unfiltered audio signal as the trigger input.

The Calibrate function

This function, activated by using the Calibrate button located below the Threshold knob, is used to automatically set the threshold level. It is especially useful for material with consistent inherent background noise, like tape hiss. This may most of the time be masked by the audio content, but becomes noticeable during silent passages.

Use it as follows:

1. Find a part of the audio material, preferably not too short, where only the background noise is heard.
   If you can only find a short background noise section, try looping it.

2. Play it back, and click on the Calibrate button.
   The button will blink for a few seconds, and then automatically set the threshold so that the noise will be silenced (gated) during passages where there is no other signal present.

The Compressor section

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

The available parameters work as follows:

Parameter	Values	Description
Threshold	-60 - 0dB	This setting 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 - 8:1	Ratio determines the amount of gain reduction applied to signals over the set threshold. A ratio of 3:1 means that for every 3 dB the input level increases, the output level will increase by only 1 dB.
Attack	0.1-100 ms	This 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) will pass through unprocessed.
Release	10-1000ms 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.
MakeUp Gain	0 - 24dB	This parameter is used to compensate for output gain loss, caused by compression.
Compressor Mode	RMS/Peak	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.

The Limiter section

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

The available parameters are the following:

Parameter	Values	Description
Threshold	-12 - 0dB	This setting determines the maximum output level. Signal levels above the set threshold are affected, but signal levels below are left unaffected.
Release	10-1000ms or "Auto mode"	This parameter 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, Limiter will automatically find an optimal release setting that varies depending on the audio material.

The Routing section

In the Routing section 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. Beside each processor label is a number. These numbers are used to represent the signal flow options shown in the Routing section. There are three routing options:

- 1-2-3 (Compressor-Gate-Limit)
- 2-1-3 (Gate-Compressor-Limit)
- 1-3-2 (Compressor-Limit-Gate)

MIDI Gate

Gating, in its fundamental form, silences audio signals below a certain set threshold level. I.e. when a signal rises above the set level, the Gate opens to let the signal through while signals below the set level are cut off. MIDI Gate however, is a Gate effect that is not triggered by threshold levels, but instead by MIDI notes. Hence it needs both audio and MIDI data to function.

Setting up

MIDI Gate requires both an audio signal and a MIDI input to function.

To set it up, proceed as follows:

1. Select the audio to be affected by the MIDI Gate.
   This can be audio material from any audio track, or even a live audio input (provided you have a low latency audio card).

2. Select the MIDI Gate as an insert effect for the audio track.
   The MIDI Gate control panel opens.

3. Select a MIDI track to control the MIDI Gate.
   This can be an empty MIDI track, or a MIDI track containing data, it doesn't matter. However, if you wish to play the MIDI Gate in real-time – as opposed to having a recorded part playing it – the track has to be selected for the effect to receive the MIDI output.

4. Open the Output ("out:") pop-up menu for the MIDI track and select the MIDI Gate option.
   The MIDI Output from the track is now routed to the MIDI Gate.

What to do next depends on whether you are using live or recorded audio and whether you are using real-time or recorded MIDI. We will assume for the purposes of this manual that you are using recorded audio, and play the MIDI in real-time.

Make sure the MIDI track is selected and start playback.

5. Now play a few notes on your MIDI keyboard.
As you can hear, the audio track material is affected by what you play on your MIDI keyboard.

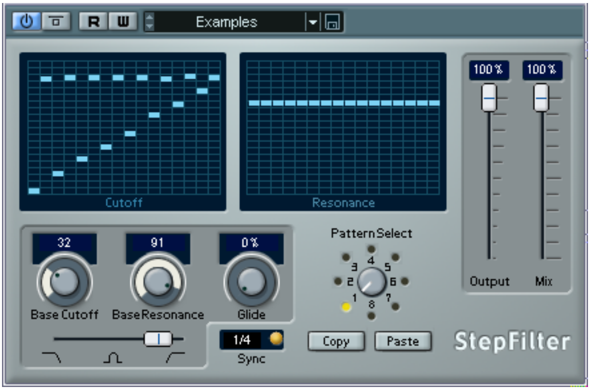
The following MIDI Gate parameters are available:

Parameter	Description
Attack	This is used for determining how long it should take for the Gate to open after receiving a signal that triggers it.
Hold	Regulates how long the Gate remains open after a Note On or Note Off message (see Hold Mode below).
Release	This determines how long it takes for the Gate to close (in addition to the value set with the Hold parameter).
Note To Attack	The value you specify here determines to which extent the velocity values of the MIDI notes should affect the Attack. The higher the value, the more the Attack time will increase with high note velocities. Negative values will give shorter Attack times with high velocities. If you do not wish to use this parameter, set it to the 0 position.
Note To Release	The value you specify here determines to which extent the velocity values of the MIDI notes should affect the Release. The higher the value, the more the Release time will increase. If you do not wish to use this parameter, set it to the 0 position.
Velocity To VCA	This controls to which extent the velocity values of the MIDI notes determine the output volume. A value of 127 means that the volume is controlled entirely by the velocity values, while a value of 0 means that velocities will have no effect on the volume.
Hold Mode	Use this switch to set the Hold Mode. In Note-On mode, the Gate will only remain open for the time set with the Hold and Release parameters, regardless of the length of the MIDI note that triggered the Gate. In Note-Off mode on the other hand, the Gate will remain open for as long as the MIDI note plays, and then apply the Hold and Release parameters.

Filter plug-ins

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

StepFilter



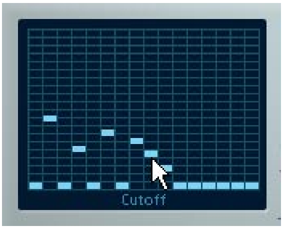
StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

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 will be set to the pointer position.

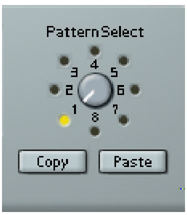


Setting filter cutoff values in the grid window.

- The horizontal axis shows the pattern steps 1-16 from left to right, and the vertical axis determines the (relative) filter cutoff frequency and resonance setting.
  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 directly.

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 patterns are saved together in the 8 Pattern memories.
- To select new patterns you use the pattern selector.
  New patterns are all set to the same step value by default.



Pattern Selector

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 memory location, which is useful for creating variations on a pattern.

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

StepFilter parameters

Parameter/Value	Description
Base Cutoff	This sets the base filter cutoff frequency. Cutoff values set in the Cutoff grid window are values relative to the Base Cutoff value.
Base Resonance	This sets the base filter resonance. Resonance values set in the Resonance grid window are values 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	This slider selects between lowpass (LP), bandpass (BP) or highpass (HP) filter modes (from left to right respectively).
Sync 1/1-1/32 (Straight, Triplet or Dotted)	This sets the pattern beat resolution, i.e. what note values the pattern will play in relation to the tempo.
Mix	Adjusts the mix between dry and processed signal.
Output	Sets the overall volume.

Modulation plug-ins

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

Chorus

The Chorus plug-in adds short delays to the signal, and pitch modulates the delayed signals to produce a “doubling” effect.

The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Chorus is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Shapes	This sets the modulation waveform. Triangle produces smooth modulation, saw produces ramp shaped modulation and pulse waveform produces stepped modulation.
Frequency	This sets the modulation rate.
Delay	This controls the depth of the Chorus effect.
Stages	This adds one to three more delay taps, producing a thicker, multi-layered chorus effect.

- **Note that clicking and dragging in the display allows you to adjust the Frequency and Delay parameters at the same time!**

Flanger

Flanger is a classic flanger effect with stereo enhancement.

The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If the Flanger is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Output	Sets the overall volume.
Tempo sync on/off	The button above the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the flanger sweep (1/1 - 1/32, straight, triplet or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Tempo Sync knob	This is the note value multiplier (x1 to x10) for the flanger sweep when tempo sync is used.
Shape Sync knob	This changes the shape of the modulating waveform, altering the character of the flanger sweep.
Feedback	This determines the character of the flanger effect. Higher settings produce a more “metallic” sounding sweep.
Depth	This sets the depth of the modulation sweep.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Stereo Basis	This sets the stereo width of the effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.

You can also change parameters in the graphic display. This works as follows:

- If tempo sync is on, you can set the base note value by clicking the waveform and dragging left and right.
  When tempo sync is off, this sets the Rate parameter.

- You can set the Depth parameter by clicking the waveform and dragging up and down.
  This means you can freely adjust Rate and Depth at the same time by clicking and dragging.

- By click-dragging the green/blue line in the display left or right you can change the Stereo Basis parameter.

Metalizer

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

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Metalizer is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Output	Sets the overall volume.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 - 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, without sync to tempo.
On button	Turns filter modulation on and off. When turned off, the Metalizer will work as a static filter.
Mono button	When this is on, the output of the Metalizer will be in mono.
Sharpness	Governs the character of the filter effect. The higher the value, the narrower the affected frequency area, producing sharper sound and a more pronounced effect.
Tone	Governs the feedback frequency. The effect of this will be more noticeable with high Feedback settings.
Feedback	The higher the value, the more "metallic" the sound.

- **Note that clicking and dragging in the display allows you to adjust the Sharpness and Tone parameters at the same time!**

Phaser

The Phaser plug-in produces the classic "swooshing" sound that characterizes phasing. It works by shifting the phase of the signal and adding it back to the original signal, causing partial cancellation of the frequency spectrum.

Parameter	**Description**
Mix	Sets the level balance between the dry signal and the effect. If the Phaser is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Output	Sets the overall volume.
Tempo sync on/off	The button above the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the Phaser sweep (1/1 - 1/32, straight, triplet or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Feedback	This sets the amount of feedback. A higher value produces a more pronounced effect.
Tempo Sync knob	This is the note value multiplier (x1 to x10) for the Phaser sweep when tempo sync is used.
Stereo Basis	This sets the stereo width of the effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.

You can also change parameters in the graphic display. This works as follows:

- If tempo sync is on, you can set the base note value by clicking the waveform and dragging left and right.
  When tempo sync is off, this sets the Rate parameter.

- You can set the Feedback parameter by clicking the waveform and dragging up and down.
  This means you can freely adjust the Rate and Feedback at the same time by clicking and dragging.

- By click-dragging the blue/green line in the display left or right you can change the Stereo Basis parameter.

Ringmodulator

The 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.

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

Parameter	Description
Oscillator LFO Amount	LFO Amount controls how much the oscillator frequency is affected by the LFO.

Parameter	Description
Oscillator Env. Amount	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 Wave	Selects the oscillator waveform; square, sine, saw or triangle.
Oscillator Range	Determines the frequency range of the oscillator in Hz.
Oscillator Frequency	Sets the oscillator frequency +/- 2 octaves within the selected range.
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 Waveform	Selects the LFO waveform; square, sine, saw or triangle.
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, with center position representing no modulation. Left of center, a loud input signal will slow down the LFO, whereas right of center a loud input signal will speed it up.
Invert Stereo	This inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.
Envelope Generator (Attack and Decay dials)	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 sets 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	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.
Mix	Adjusts the mix between dry and processed signal.
Output	Sets the overall volume.

Rotary

The Rotary plug-in simulates the classic effect of a rotary speaker. A rotary speaker cabinet features variable speed rotating speakers to produce a swirling chorus effect, commonly used with organs. Rotary features all the parameters associated with the real thing. The included presets provide good starting points for further tweaking of the numerous parameters.

The parameters are as follows:

Parameter	Description
Speed	This controls the speed of the Rotary in three steps: Stop/Slow/Fast.
MIDI Ctrl	Selects the MIDI continuous controller for the Speed parameter. See page 32.
Mode	Selects whether the Slow/Fast speed setting is a switch (left button is lit), or a variable control (right button lit). When switch mode is selected and Pitch Bend is the controller, the speed will switch with an up or down flick of the bender. Other controllers switch at 64.
Overdrive	Applies a soft overdrive or distortion.
Crossover Freq.	Sets the crossover frequency (200-3000Hz) between the low and high frequency loudspeakers.
Mic Angle	Sets the simulated microphone angle. 0 = mono, 180 = one mic on each side.
Mic Distance	Sets the simulated microphone distance from the speaker in inches.

Parameter	Description
Low Rotor Amp Mod.	Adjusts amplitude modulation depth.
Low Rotor Mix Level	Adjusts overall bass level.
Hi Rotor Amp Mod.	High rotor amplitude modulation.
Hi Rotor Freq. Mod.	High rotor frequency modulation.
Phasing	Adjusts the amount of phasing in the sound of the high rotor.
Hi Slow	Fine adjustment of the high rotor Slow speed.
Hi Rate	Fine adjustment of the high rotor acceleration time.
Hi Fast	Fine adjustment of the high rotor Fast speed.
Lo Slow	Fine adjustment of the low rotor Slow speed.
Lo Rate	Fine adjustment of the low rotor acceleration time.
Lo Fast	Fine adjustment of the low rotor Fast speed.
Output	Adjusts the overall output level.
Mix	Adjusts the mix between dry and processed signal.

Directing MIDI to the Rotary

For real-time MIDI control of the Speed parameter, MIDI must be directed to the Rotary.

- Whenever the Rotary has been added as an insert effect (for an audio track or an FX channel), it will be available on the output ("out:") pop-up menu for MIDI tracks.
  If Rotary is selected on the "out:" menu, MIDI will be directed to the plug-in from the selected track.

Symphonic

The Symphonic plug-in combines a stereo enhancer, an auto-panner synchronized to tempo and a chorus-type effect. For best results, apply the Symphonic effect to stereo signals.

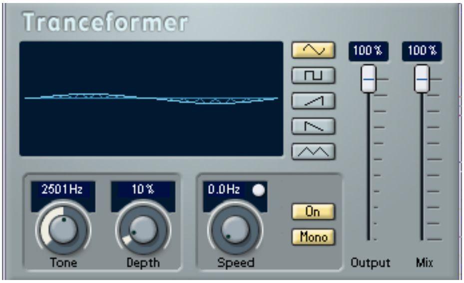
The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Symphonic is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Tempo sync on/off	The button below the Temp sync knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Tempo Sync pop-up	If tempo sync is on, this is where you specify the base note value for tempo syncing the auto-panning (1/1 - 1/32, straight, triplet or dotted).
Tempo Sync knob	This is the note value multiplier (x1 to x10), determining the timing of the auto-panning.
Delay	This determines the delay time and thus the character of the chorus effect, if activated.
Depth	This controls the depth of the chorus effect. If you only want to use Symphonic as an auto-panner or a stereo enhancer, set this to 0%.
Rate	This sets the modulation rate for the chorus effect, if activated.
Stereo Basis	When the Auto-panner is activated, this sets the stereo width of the panning. When the Auto-panner is deactivated (Tempo sync off), this determines the depth of the stereo enhancer effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.
Output	Adjusts the output level of the effect.

You can also change parameters in the graphic display. This works as follows:

- You can set the Rate parameter by clicking the waveform and dragging left and right.
- You can set the Depth parameter by clicking the waveform and dragging up and down.
  This means you can freely adjust Rate and Depth at the same time by clicking and dragging.
- By click-dragging the green/blue line in the display left or right you can change the Stereo Basis parameter.

Tranceformer



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.

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect.
Output	Adjusts the output level of the effect.
Tone	Sets the frequency (pitch) of the modulating oscillator (1 to 5000 Hz).
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 - 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, without sync to tempo.
On button	Turns modulation of the pitch parameter on or off.
Mono button	Governs whether the output will be stereo or mono.
Depth	Governs the depth of the pitch modulation.
Waveform buttons	Sets the pitch modulation waveform.

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

Other plug-ins

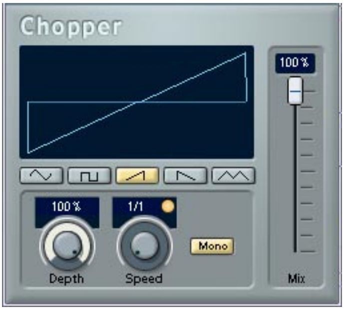
This section contains descriptions of the plug-ins in the “Other” category.

Bitcrusher

If you’re 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 parameters are:

Parameter	Description
Mode	Select one of four operating modes for the Bitcrusher. Each mode will produce a different sounding result. Modes I and III are nastier and noisier, while modes II and IV are more subtle.
Depth	Use this to set the desired bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 will create mostly noise.
Sample Divider	This 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 will be eliminated, turning the signal into unrecognizable noise.
Mix	This slider regulates the balance between the output from the Bitcrusher and the original audio signal. Drag the slider upwards for a more dominant effect, and drag it downwards if you want the original signal to be more prominent.
Output	Governs the output level from the Bitcrusher. Drag the slider upwards to increase the level.

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 parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Chopper is used as a send effect, this should be set to maximum.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on (the button lights up) or off.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 - 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, without sync to tempo.
Stereo/Mono button	Determines whether the Chopper will work as an auto-panner (button set to "Stereo") or a tremolo effect (button set to "Mono").
Waveform buttons	Sets the modulation waveform.
Depth	Sets the depth of the Chopper effect. This can also be set by clicking in the graphic display.

Vocoder

The Vocoder can apply sound/voice characteristics taken from one signal source, called the “modulator” and apply this to another source, called the “carrier”. A typical application of a vocoder is to use a voice as a modulator and an instrument as a carrier, making the instrument “talk”. A vocoder works by dividing the source signal (modulator) into a number of frequency bands. The audio attributes of these frequency bands can then be used to modulate the carrier.

The Vocoder has a built-in carrier (basically a simple polyphonic synthesizer) but you can also use an external carrier, see page 39.

Setting up – using MIDI

In this mode, the Vocoder is set up slightly differently than other plug-in effects. This is because this setup requires both an audio signal (as the modulator source) and a MIDI input (to play the carrier) to function. To set up for using an external carrier, see page 39.

To set up for use, proceed as follows:

1. Select a source for the modulator.
The modulator source can be audio material from any audio track, or even a live audio input routed to an audio track (provided you have a low latency audio card).

- Good modulator source material are talking or singing voices or percussive sounds, e.g. drum loops.
Static pads or soft ambient material are generally less appropriate for use as modulators, but there are no absolute rules as to what could be used as a modulator source.

2. Select the Vocoder as an insert effect for the audio channel with the modulator signal.

3. Make sure that the Vocoder Mode is set to “MIDI”.

4. Select a MIDI track.
This can be an empty MIDI track, or a MIDI track containing data, it doesn’t matter. However, if you wish to play the Vocoder in real-time – as opposed to having a recorded part playing it – the track has to have monitoring activated (or be record enabled) for the Vocoder to receive the MIDI output.

5. Select “Vocoder” from the MIDI “out:” pop-up menu for the MIDI track.
The MIDI Output from the track is now routed to the Vocoder. There is an indicator on the Vocoder panel below the Mode switches that blinks when receiving MIDI.

That concludes setting up – you are now ready to start vocoding!

What you do next depends on whether you are using live or recorded audio as the modulator source and whether you are using real-time or recorded MIDI as the carrier input. We will assume for the purposes of this manual that you are using recorded audio as the modulator, and play the carrier in real-time.

6. Make sure the MIDI track is record enabled and start playback.

7. Now play a few notes on your MIDI keyboard.
As you can hear, the audio track material, or rather its formant characteristics, is now applied to the Vocoder’s built-in sound source!

Setting up – using an external carrier

There are two modes for using an external carrier:

- “Ext” mode is when the carrier and the modulator can be any two audio sources.
The synth section is disabled and grayed out when this mode is selected. MIDI input and the Gap Thru Vocoder parameter are also disabled.

- “MIDI+Ext” mode mixes the audio carrier with the Vocoder’s synth sound.
This is described on page 40.

To use an external carrier instead of the built-in synth ("Ext mode"), you set up as follows:

1. Create a Group channel from the Add Track submenu on the Project menu.
2. Open an audio file you wish to use as the carrier source and place it on an empty audio track.
3. Pan the audio channel full right in the Mixer or in the Inspector.
4. Route the output of the audio channel to the group.
5. Open an audio file you wish to use as the modulator source and place it on another empty audio track.
   Events on the two audio tracks (carrier and modulator) have to play back simultaneously for the Vocoder to work.
6. Pan the modulator audio channel full left in the Mixer or in the Inspector.
7. Route the output of the modulator audio channel to the group.
8. Select the Vocoder as an insert effect for the group channel.
9. Open the Vocoder panel and activate the "Ext." Mode button.
10. If you now start playback, the carrier channel will be modulated by the modulator channel!
    Note that the synth section on the left half of the Vocoder panel and the "Gap Thru" parameter are now disabled.

Setting up – using an external carrier plus MIDI

Setting up is the same as for using an external carrier, except that a MIDI track with its output routed to the Vocoder should also be present. The MIDI track can either play the Vocoder synth in real time or from prerecorded parts. Make sure that monitoring (or record enable) is activated for the track so that the Vocoder synth will receive MIDI played in real time.

- Set up as described, and activate "MIDI+Ext." mode on the Vocoder panel.
  Any incoming MIDI now triggers the Vocoder synth, and the synths output is mixed with the audio carrier signal.

Vocoder parameters

The Vocoder parameters govern the general sound quality of the vocoded sound.

Parameter	Description
Number of Bands	This governs how many frequency bands the modulator signal is divided into (2-24). Fewer bands will provide a thinner more resonant sound, whereas using more bands will make the sound fuller and more intelligible.
Bandwidth	This sets the bandwidth for the frequency bands, which affects the overall timbre. Very narrow bandwidth settings will produce a thin, whistle-like sound.
Min./Max. Freq.	These parameters set the minimum and maximum frequency limits for the Vocoder, respectively.
log/lin	Log/Lin controls how the frequency bands are spaced between the minimum and maximum frequencies. Log = equal spacing in octaves, Lin = equal spacing in Hz. This affects the basic timbre of the Vocoder.
Env.Speed	This determines the attack and release times of the Vocoder envelope. Fast settings will cause the modulator signal to trigger the Vocoder instantly, longer settings will gradually increase the attack/release times, providing a more subtle Vocoder effect. If set to "HOLD" the modulator is "frozen", and doesn't affect the carrier synth at all.
High Thru	This lets through high frequencies around the "S" frequency from the original input signal while notes are played.
Talk Thru	Adjusts the level of the original input signal passed to the Vocoder output while notes are played.
Gap Thru	Gap Thru (only available in MIDI mode) sets the level of the original input signal that is passed to the Vocoder output when no MIDI notes are being played. This lets you apply the Vocoder to a vocal track adding vocoded parts just where you want them.
Output	This controls the output level of the Vocoder.
Emphasis	This is a highpass filter, gradually cutting lower frequencies while letting high frequencies pass.

Vocoder synth parameters

If the built-in synthesizer is the carrier, it is the sound of this instrument that the modulator source is applied to. The synth is polyphonic with up to 8 voices and features 2 oscillators per voice. The synth has the following parameters:

Parameter	Description
Voices	This sets the number of voices for the synth (1-8).
Fine Tune	Tunes the oscillators ± a semitone, in cents (100th of a semitone) steps.
Pitch Bend	Sets the up/down range of the Pitch Bend in semitone steps (1-12).
Noise	Adds white noise to the sound.
NoiseMod	This makes the oscillators modulate the noise level. This gives the noise a rasping sound, turning "sss" into "zzz".
P.Drift	Adds random pitch variation to the oscillators.
P.Glide	This makes the pitch glide between notes played. The parameter controls the time it takes for the pitch to glide from one note to the next.
P.Bright	This is a lowpass filter that can be used to soften the tone of the oscillators. It does not affect the white noise generator.
P.Detune	Allows you to detune one of the oscillators in cent steps.
LFO Rate	Controls the LFO rate (for vibrato).
Vibrato	Adds vibrato to the oscillators. This can also be controlled by using the Mod Wheel.

Restoration

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

Grungelizer

The 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 available parameters are as follows:

Parameter	Description
Crackle	This adds crackle to create that old vinyl record sound. The farther to the right you turn the dial, 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	This dial regulates the amount of static noise added.
Distort	Use this dial to add distortion.
EQ	Turn this dial to the right to cut off the low frequencies, and create a more hollow, lo-fi sound.
AC	This emulates a constant, low hum of AC current.
Frequency switch	This sets the frequency of the AC current (50 or 60Hz), and thus the pitch of the AC hum.
Timeline	This dial regulates the amount of overall effect. The farther to the right (1900) you turn this dial, the more noticeable the effect.

Reverb plug-ins

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

Reverb A

Reverb A is a reverb plug-in which provides smooth, dense reverb effects. Reverb A has the following parameters:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect (wet). If Reverb A is used as a send effect, this should be set to maximum wet, as you can control the dry/wet balance with the send.
Room Size	This setting determines the “size” of the simulated room environment.
Predelay	This parameter sets a delay between the direct sound and the reverb effect output. A short predelay before the reverb reduces reverb “clutter” which blurs the sound, and makes the reverb effect more natural-sounding.
Reverb Time	This parameter sets the length of the reverb time.
Filter HighCut	This filters out high frequencies for the reverb, which can make the reverb sound softer.
Filter LowCut	This filters out the lower frequencies for the reverb. It can be used to reduce low frequency “rumble”.

Reverb B

The Reverb B provides reverb with low processor demands. It has the following parameters:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Reverb B is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Room Size	Governs the "size" of the simulated room environment.
Predelay	This parameter sets a delay between the direct sound and the reverb effect output. A short predelay before the reverb reduces reverb "clutter" which blurs the sound, and makes the reverb effect more natural-sounding.
Reverb Time	This parameter sets the length of the reverb effect.
Damp	This parameter "dampens" the higher frequencies, producing a rounder and smoother sounding reverb.

2

The included VST Instruments

VB-1 Bass Synth

The VB-1 is a virtual bass instrument built on real-time physical modelling principles. It has the following properties:

- VB-1 is polyphonic with up to 4 voices.
- VB-1 receives MIDI In Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to the VB-1.
- VB-1 responds to the following MIDI messages:
  MIDI Note On/Off (velocity governs volume), Volume and Pan.

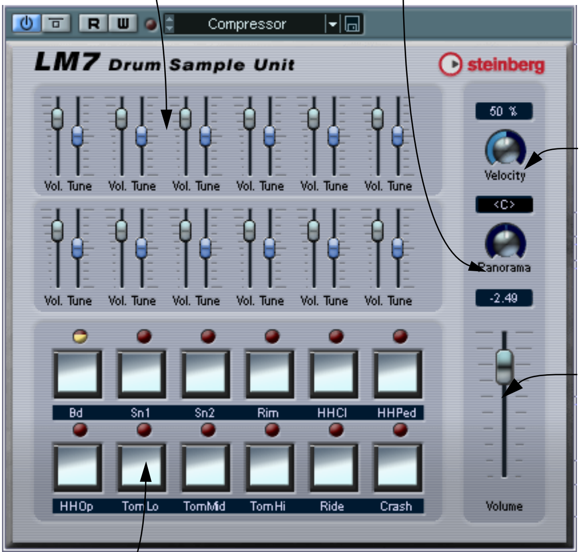
VB-1 Parameters

Parameter	Description
Pick-up	To change the Pick-up position, click and drag the lower end of the Pick-up. Positioning the pick-up position towards the left produces a hollow sound that emphasizes the upper harmonics of the plucked string. When placed towards the right position, the tone is fuller and warmer.
Pick	This determines where along the length of the string the initial pluck is made. This controls the "roundness" of the tone, just like on a real bass. Click-drag the Pick to change position.
Shape	This knob selects the basic waveform used to drive the plucked string model. This parameter can drastically change the sound character. The control smoothly morphs through the waves. It is possible to create sounds that have no relation to a bass guitar with this control.
Volume	This knob regulates the VB-1 volume.
Damper	This parameter controls the length of time the string vibrates after being plucked.

LM-7 Drum Machine

Volume and Tune faders (for each drum sound).

This adjusts the Pan (the position in the stereo image) for the individual drums. The setting is applied to the currently selected drum, indicated by a lit yellow LED over the Pad button.



This sets the global velocity sensitivity for LM-7.

Master Volume

Pad (one for each drum sound). Press to audition the drum sound assigned to the Pad, or to select a sound for adjusting pan.

The LM-7 is a 24-bit drum machine. It has the following properties:

- LM-7 is polyphonic with up to 12 voices.
- LM-7 receives MIDI in Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to LM-7.
- LM-7 responds to the following MIDI messages:
  MIDI Note On/Off (velocity governs volume).

LM-7 Parameters

Parameter	Description
Velocity	This sets the global velocity sensitivity for LM-7. The higher the value, the more sensitive LM-7 will be to incoming velocity data. If set to “0”, the sounds will play back with a fixed velocity value.
Volume sliders	The volume sliders are used to adjust the volume for each individual drum sound.
Tune sliders	The tune sliders are used to tune each individual drum sound, up or down 1 octave.
Pad	The Pads are used for two things: To audition the individual drum sounds, and to select a sound for adjusting pan.
Panorama	This is used to position an individual sound in the stereo image. The setting applies to the currently selected sound, indicated by a lit yellow LED over the Pad button.

Drum sounds

LM-7 comes with six sets of drum sounds. “Compressor”, “909” and “Percussion” are loaded as the default sets when launching LM-7. “Modulation”, “Fusion” and “DrumNbass” can be loaded by selecting “Load Bank” from the File menu and opening the lm7_second_set.fxb file (which is located in the Vstplugins/Drums subfolder).

- You switch between the three loaded sets by using the pop-up menu (just like you switch between effect programs).

MIDI note mapping

The table below shows how the drum sounds are assigned to note values on your MIDI keyboard. The mapping is GM compatible:

Drum sound	Note	Comment
Bd	C1	
Rim	C#1	Compressor only.
Snare	D1	
Clap	D#1	909 only.
Hi-Hat	F#1	
O-Hi-Hat	A#1	
Tom 1	A1	
Tom 2	B2	
Tom 3	D2	
Crash	C#2	
Ride	D#2	Compressor only.
Tambourine	F#2	Percussion only.
Cowbell	G#2	Percussion only.
Hi Bongo	C3	Percussion only.
Lo Bongo	C3#	Percussion only.
Conga Mute	D3	Percussion only.
Conga Open	D#3	Percussion only.
Conga Lo	E3	Percussion only.
Timbale Lo	G3	Percussion only.
Timbale Hi	G#3	Percussion only.
Cabasa	A3	Percussion only.

Universal Sound Module (USM)

The USM is a General MIDI compatible sound module. General MIDI (GM) is a standard set up by the MIDI Manufacturers Association (MMA) and the Japanese MIDI Standards Committee (JMSC).

It defines a standardized group of sounds and the minimum requirements for General MIDI compatible synthesizers or sound modules, so that a specially prepared sequence or MIDI file that is sent to the instrument via MIDI will play back the correct sound types, regardless of make and model of the instrument.

MIDI identifies sounds by their program change number. Before the General MIDI standard was introduced, the same MIDI program change number often addressed totally different types of sound in any two synthesizers or sound modules from different manufacturers, e.g. a flute type sound in one instrument and a piano type sound in the other.

With the introduction of General MIDI standard compatible instruments this changed. These instruments use the same program change numbers for the same types of instruments.

So, if the person that prepared a sequence or MIDI file wants the melody to be played by a "piano", he can use a certain program change command embedded into the sequence to automatically select a piano sound in any GM compatible sound module. The GM standard, however, does not specify in great detail how that piano should sound. It is simply assumed that the manufacturer reproduces an acoustic piano within the capabilities of the instrument. A consequence of this was that, depending on the GM module used, a song would sound very different, even though the instrument sounds were mapped correctly.

This problem is solved by the Universal Sound Module! Cubase users can make sure that their music created using the USM will sound exactly the same when played back on another computer, because the sound reproduction is no longer hardware based.

- The USM features over 70 MB of sampled waveforms and four stereo outputs.
- The USM is polyphonic with up to 96 voices.
- The USM receives MIDI in 16 channel Multi mode (simultaneous multitimbral playback on 16 MIDI channels).
  In other words, one USM unit can play up to 16 MIDI Tracks, each with a different sound.
- The USM responds to the following MIDI messages:
  MIDI Note On/Off (velocity governs volume).
  Volume.
  Pan.
  Pitch Bend (up to ± 12 semitones).
  Modulation (vibrato).

Selecting Sounds

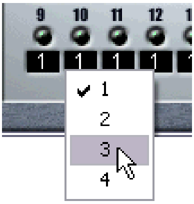
According to the General MIDI Standard, MIDI channel 10 is reserved for drums. This can not be changed.

The USM features 128 different sound patches. Selecting programs is done by sending program change messages by selecting an option from the program ("prg:") pop-up menu in the Inspector.

Selecting Outputs

The USM features four stereo outputs, allowing for flexible routing of sounds to different effect processors etc. By default, all MIDI channels are routed to USM stereo output “1”.

- To select another output, click the “Output” field below the Channel Activity indicator for the MIDI channel you wish to direct to another output.



This opens a pop-up allowing you to select one of the four stereo outputs.

USM Parameters

Parameter	Description
Master Volume	Sets the master output volume for the USM.
Pitchb. Range	Sets the range for incoming Pitchbend messages (selectable between 1 to 12 semitones).
LFO Speed	Governs the speed of the vibrato. The vibrato depth is controlled via MIDI Modulation messages (for example, using the Mod Wheel on your MIDI controller).
MIDI channel activity indicators 1-16.	These light up to indicate activity on the corresponding MIDI channel.
Output 1-16	Clicking in this field opens a pop-up allowing you to direct the corresponding USM MIDI channel to one of the four available stereo outputs.

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