VST plug-ins from previous Cubase VST versions

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Introduction

This document describes the VST effects from old versions of Cubase VST (Cubase VST 5 and earlier), included for reasons of compatibility.

Cubase 5 audio effect plug-ins

Autopole

The Autopole is a filter effect containing two separate filters capable of operating in four different modes, an Envelope Generator and an LFO with four different waveforms. It also lets you choose between three different Signal Routing modes to control how an incoming signal should be sent through the filters.

The Autopole should be used as an insert effect. If you want to apply it on several channels at once, you can use it as an insert effect on a group channel and route the desired channels to the group channel.

The parameters for the different “sections” of the Autopole are the following:

The Filters

Parameter	Description
Filter Mode buttons (LP, BP, HP, Notch)	These buttons let you decide in which mode the Filter should operate: LP: This is a Low-Pass Filter that “filters out” the high frequency content of the incoming signal, according to a certain set threshold level. Only signals below the threshold will pass through. BP: This is a Band-Pass filter that only lets signals around the set frequency through, filtering out all other content. HP: This is a High-Pass Filter that “filters out” the low frequency content of the incoming signal, according to a certain set threshold level. Only signals above the threshold will pass through. Notch: This is a filter that cuts off the signals around the set frequency, leaving all other content unaffected.
Cutoff	This is used for setting the Cutoff frequency, i.e. the threshold at which the filter should “kick in”. The farther to the right you drag the sliders, the higher the frequency.
Resonance	This affects the resonance of the filter. Increasing the resonance gives a more pronounced, lively filter sound. Be wary of extremely high levels of resonance since they might induce unpleasant distortion.
LFO Mod	These sliders govern how the filter cut-off frequencies are affected by the LFO (see below). The sliders are “zero-centered”, meaning that in the middle position (zero) no LFO modulation will be applied. By dragging the sliders to the left or right, you cause an increasing amount of modulation to the cut-off frequency. The difference is that if you drag the sliders to the left, the waveform of the LFO is inverted, creating a different effect.
EG Mod	These sliders work in conjunction with the Envelope Generator settings (see below). They control to which extent the cut-off frequencies of the filters should be affected by the Envelope Generator. Drag the sliders to the right if you want to raise the cut-off frequencies and if you want to lower the frequencies, drag the sliders to the left. Leave the sliders in the middle position if you don’t want Envelope data to affect the cut-off frequencies.

Signal Routing

By clicking one of the three buttons, you choose how an input stereo signal will pass though the filters. The signal flow chart to the left of the buttons indicates the path:

- Option # 1 will have the signal from each channel pass through both of the filters in series (one after the other).
- With option # 2, the signal from each channel will pass through both of the filters in parallel, and then be mixed at the output.
- Finally, option # 3 causes the signals from both channels to each pass through a separate filter. I.e. the left signal only passes through Filter A, and the right signal only passes through Filter B.

When using the Autopole with mono material, options 1 and 2 are the best choices (sending the signal through the filters in series or in parallel, respectively).

Envelope Generator

This section controls how the input signal is converted into Envelope data. This, in its turn, affects the EG Mod sliders in the Filter sections and the Modulation slider in the LFO section:

Parameter	Description
Attack	This regulates how fast the Envelope Generator will respond to an input signal as it rises in sound level. The farther to the left you drag the slider, the faster the response will be.
Release	This governs how fast the Envelope Generator will respond to an input signal as it drops in sound level. The farther to the left you drag the slider, the faster the response will be.

LFO

These are the controls for the Low Frequency Oscillator, used for adding continuous filter movement, wah-wah effects, etc:

Parameter	Description
Frequency	This slider controls the speed of the LFO. The farther to the right you drag the slider, the faster the oscillation will be.
Modulation	Use this slider to control how the speed of the LFO should be modulated by the Envelope Generator (and thus by the level of the input signal). If you drag the slider to the left, a loud input signal will cause the LFO to slow down and if you drag to the right, the LFO will speed up. In the middle position, the speed of the LFO is unaffected.
Waveform Buttons	These buttons are used for choosing a waveform for the LFO. You can choose between Square, Sine, Saw and Triangle.

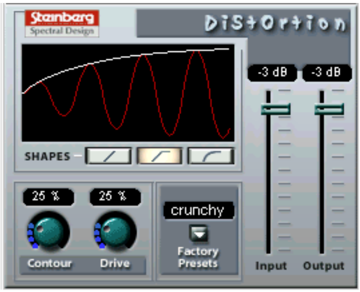
Output Controls

Parameter	Description
Mix	This controls the balance between the output from the Autopole and the input signal. In the middle position, both signals are equally mixed. The higher you drag the slider, the more dominant the effect will be. Conversely, with lower settings the unaffected original signal will be more pronounced.
Gain	This slider regulates the output level from the Autopole. The higher you drag the slider, the higher the level.
Sync	When this is activated, the LFO will restart in intervals according to the current Song tempo, which is useful for tempo sync and special effects. Click the button to activate sync, and then click in the small display to the right to select at which note values the LFO should be restarted: 1/1, 1/2, 1/4, 1/8 or 1/16. For example, setting this to 1/4 will make the LFO restart on each beat (quarter note) according to the current tempo.

Chopper2

Chopper2 is an earlier version of the Chopper plug-in, included for reasons of compatibility. It has the same parameters as Chopper with the addition of independent input and output level settings.

Distortion



The Distortion effect plug-in is capable of producing anything from a soft “crunch” to all-out distortion.

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	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”. Note that these presets are not stored parameter settings, but different basic distortion algorithms.

Karlette

The Karlette is a four-channel delay that emulates a "tape-loop" echo. The four "tape-heads" can be set to a certain note value or a certain time, depending on whether Tempo Sync is activated or not.

For the four "tape-heads", you can set the following parameters:

Parameter	Description
Delay	With the sync button activated, the delay can be set to a note value synchronized to the Cubase SX/SL tempo. If the sync button is deactivated, the delay can be freely set to a time value.
Volume	The amplitude of the delay. With the knob turned all the way to the left, the delay is muted.
Damp	The higher the value, the more the delay is dampened (the high frequencies are attenuated) to produce a more subtle effect.
Pan	Sets the stereo position for the delay.
Feedback	Sets the number of delay repeats.

In addition, the following global parameters are available:

Parameter	Description
Dry/Wet	Sets the level balance between the dry signal and the effect. If Karlette is used as a send effect, this should be set to maximum as you can instead control the dry/effect balance with the send.
Sync	Turns Tempo Sync on or off.

Metalizer2

The Metalizer2 is an older version of the Metalizer plug-in, included for reasons of compatibility.

MIDIComb

This is a comb filter, which can be described as one or several very short delays with high feedback, causing resonating peaks at certain frequencies. To operate, the MIDI Comb needs both audio and MIDI input. While the MIDI Comb is used as an insert effect on an audio channel, the signals that actually trigger it are the ones sent from a MIDI track.

Setting Up

The MIDI Comb 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 Comb.
   This can be audio material from any Audio Track, or even a live audio input routed to a Audio Track (provided you have a low latency audio card). If a live audio input is used, monitoring must be set to input (the "In" buttons in the Inspector must be lit).

2. Select the MIDI Comb as an Insert effect for the Audio channel.
   Click the Edit button to open the MIDI Comb panel.

3. 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 MIDI Comb 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 column for the MIDI Track pop-up menu and select MIDI Comb.
The MIDI Output from the Track is now routed to the MIDI Comb.

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 MIDI Comb is polyphonic with up to 8 voices, i.e. you can play up to 8 MIDI notes at once and each tone will produce a separate resonating tone.

You can now make settings for the MIDI Comb using the following parameters:

Amp e.g.

Parameter	Description
Atk	Use this slider to set the attack time of the resonant tones created by the comb filter - i.e how soon they will start to resonate after being triggered by MIDI notes. The farther down you drag the slider, the slower the attack.
Rel	This controls the release time of the resonant tones created by the comb filter - i.e. how soon the sound will be cut off. The farther up you drag the slider, the longer the sound will resonate.

Key Velocity Modifiers

Parameter	Description
Level	This determines how the filter responds to MIDI notes with different velocity values. At the middle setting, all tones produced by the filter will sound at an equal level regardless of the velocity values of the MIDI notes that trigger them. If you move the slider upwards, MIDI notes with higher velocity values will produce louder comb filter tones. Conversely, moving the slider downwards causes the level of the filter tones to increase with lower MIDI note velocities.
Res	This affects the resonance (feedback) of the produced tones depending on the velocity value of the MIDI notes that trigger them. In the middle position, the resonance is unaffected regardless of velocity. By dragging the slider upwards, tones triggered by MIDI notes with a high velocity value will get increased resonance. By dragging the slider downwards, tones triggered by MIDI notes with a low velocity value will become more resonant.
HPF & LPF	The MIDI Comb features both a High-Pass filter and a Low-Pass filter (see "Filters" below) that can be used for "filtering out" certain frequencies of the resonating tones according to a certain set filter cutoff frequency. These two sliders determine how much the High-Pass and Low-Pass filters should be affected by the MIDI note velocity values. Positive values cause higher velocities to increase the effect of the filters, negative values cause higher velocities to decrease the effect.

Feedback

Parameter	Description
Feedback	This slider governs the amount of effect output from the MIDI Comb that is fed back in again. The more effect feedback, the more complex the sound. Drag the slider upwards to increase feedback.

Filters

Parameter	Description
LP Cut-off	Use this to set the frequency threshold of the Low-Pass Filter. This filter cuts off all of the high frequencies relative to the set threshold. The farther up you drag the slider, the more of the high frequencies will be allowed to pass through.
HP Cut-off	Use this to set the frequency threshold of the High-Pass Filter. This filter cuts off all of the low frequencies relative to the set threshold. The farther down you drag the slider, the more of the low frequencies will be allowed to pass through.

Output

Parameter	Description
Mix	Use this to set the balance between the original, unprocessed signal and the signal affected by the MIDI Comb. In the middle position, they are equally mixed. Drag the slider upwards for a more dominant effect sound and vice versa.
Gain	This controls the output level from the MIDI Comb. Drag the slider upwards to increase the level.

Mysterizer

The Mysterizer is a multi-effect plug-in with a unique hands-on user interface. It can be used as an insert effect or a send effect, and allows you to choose between eight different effects. For each effect, you can control two parameters by clicking and dragging in the display, allowing for continuous real-time effect manipulation, subtle sweeping changes or weird, wild mutations.

Here's how to use the Mysterizer:

1. Play back some audio and route the audio channel through the Mysterizer (either as an insert or a send effect).
2. Open the Mysterizer effect control panel and click the Prog Select field to the right to select the desired effect.
   Each time you click, the next effect is selected. For a list of the effects, see below.
3. When you have selected an effect you want to use, the two text fields to the left show you which parameters are controlled on the X-axis and Y-axis respectively.
   In the figure above, the Ring Mod effect is selected, with Amount controlled on the X-axis and Frequency on the Y-axis.

4. Click in the display and drag the hair cursor to change the parameter settings.
The X-axis goes from left to right and the Y-axis goes from top to bottom, which means that the "zero setting" for both axes is in the upper left corner of the display.

5. Experiment!

The Rate and Overshoot knobs

When you move the hair cursor, you will see how the small white dot moves to follow your adjustments. This represents the actual parameter settings. The Rate and Overshoot controls at the bottom of the window control how quickly and accurately the white dot follows your movements - in other words how your mouse movements are "interpreted" by the effect.

- The Rate knobs determine how fast the Mysterizer will respond when you move the hair cursor to a new position.
You can make independent settings for the X- and Y-axis.

- The Overshoot knobs determine how far from "the target position" the white dot will be allowed to stray along the corresponding axis when moving the hair cursor.
Moderate settings can give a more natural feel when a parameter is changed. Maximum Overshoot settings (turning the knob all the way to the right) will cause constant movement back and forth along the corresponding axis relative to the target position, because the white dot will never "reach the target" and come to rest. This can create an undulating, LFO-like special effect, the speed and range of which can be controlled with the corresponding Rate knob.

The Effects

The following effects are available:

- Ring Modulator
  An effect with which the incoming audio is ring modulated by an internal, variable frequency oscillator, thereby producing new harmonics.
  X-axis governs the amount of effect, Y-axis the frequency of the built-in oscillator.
- Comb Delay
  A delay with high feedback, causing resonating peaks at certain frequencies.
  X-axis governs the feedback amount, Y-axis the manual delay time (pitch).
- Mono Delay
  A monaural delay. X-axis controls the delay feedback, Y-axis the delay time.
- Stereo Delay
  A stereo delay with which the repeats are heard in both the left and right channels.
  X-axis controls the delay feedback, Y-axis the delay time.
- Low-Pass Filter (LP)
  A filter that cuts off high frequencies according to a set frequency threshold. Only signals below the cut-off frequency will be heard.
  X-axis governs the filter resonance, Y-axis the cutoff frequency.
- High-Pass Filter (HP)
  A filter that cuts off low frequencies according to a set frequency threshold. Only signals above the cut-off frequency will be heard.
  X-axis governs the filter resonance, Y-axis the cutoff frequency.
- Band-Pass Filter (BP)
  A filter that cuts off all frequencies except those around the set cut-off frequency.
  X-axis governs the filter resonance, Y-axis the cutoff frequency.
- Distortion
  A standard distortion effect.
  X-axis controls the drive (distortion) amount, Y-axis serves as a tone control.

PhatSync

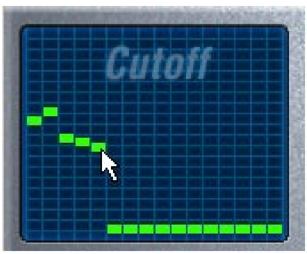
PhatSync is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

General Operation

PhatSync can produce two simultaneous 16-step patterns for the filter cutoff and resonance parameters, synced 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 show 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 affects the sound source connected to PhatSync directly.

Selecting New Patterns

- Created patterns are saved with the song, 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, then 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.

PhatSync Parameters

Parameter/Value	Description
Base Cutoff	This sets the base filter cutoff frequency. Cutoff values set in the Cutoff Grid windows are values relative to the Base Cutoff value.
Base Resonance	This sets the base filter resonance. Resonance values set in the Resonance Grid windows 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 (LP, BP, HP)	This selects between lowpass (LP), bandpass (BP) or highpass (HP) filter modes.
Sync (1/32, 1/16, 1/8, 1/4)	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.
Gain	Sets the overall volume.

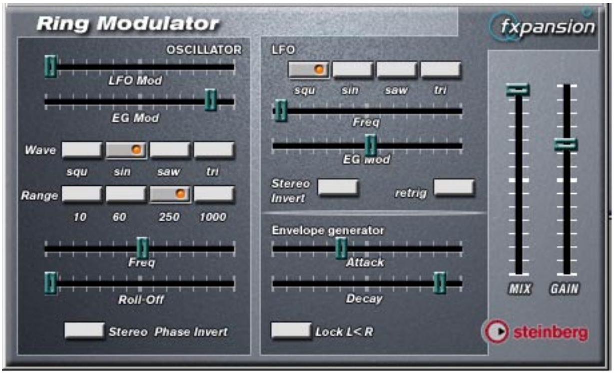
Reverb (PC only)

This is an earlier version of the Reverb B plug-in, included for reasons of compatibility.

Reverb 32 (PC only)

This is an earlier version of the Reverb A plug-in.

Ring Modulator



This is an earlier version of the Ringmodulator plug-in, with slightly different panel layout and parameters.

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

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

Parameters

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

Parameter	Description
Freq	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).
Stereo Phase Invert	Flips the phase of the oscillator waveform on the right channel.
LFO Waveform	Selects the LFO waveform; square, sine, saw or triangle.
LFO Freq	Sets the LFO Speed.
EG Mod	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.
Stereo Invert	This inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.
Retrig	Causes the LFO cycle to reset itself at the start of each bar during playback, which can be used for certain LFO effects synced to the tempo.
Envelope Generator	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 EG output level rises in response to a rising input signal. Decay controls how fast the EG output level falls in response to a falling input signal.
Lock L<R	When this switch is enabled, the L and R input signals are merged, and produce the same EG output level for both oscillator channels. When disabled, each channel has its own EG, which affect the two channels of the oscillator independently.
Mix	Adjusts the mix between dry and processed signal.
Gain	Sets the overall volume.

subBASS

The subBASS is a bass synthesizer that can generate low frequency content and track the pitch from the audio material for deep, sub-sonic bass effects.

The parameters are as follows:

Parameter	Description
Mode	There are three modes of operation: Boost produces a warm bass boost to the signal. Divide generates a pitch tracking signal an octave below the input signal. Trigger adds a decaying "boom" produced by an oscillator, typically triggered by a kick drum.
Tune	This sets the maximum frequency to be affected (20-500Hz). Set as low as possible to avoid unwanted distortion. In "Trigger" mode this sets the oscillator frequency.
Drive	In Boost mode, raising the Drive parameter adds "crunch" to the effect. In Divide mode, increasing Drive to 50% overdrives the sub-octave signal producing a square wave, and increasing Drive to 100% produces a square wave one octave above (i.e. at the original input frequency). In Trigger mode this changes the tone of the oscillator, with higher settings producing a thinner sound.
Tone	This is a lowpass filter that can be used to change the brightness of the signal. In "Trigger" mode this sets the decay time of the generated oscillator boom.
Threshold	This sets the threshold for the effect. Increase to "gate" the effect and to cut out unwanted background rumble.
Dry Level	Sets the level of the original, unprocessed signal.
FX Level	Sets the level of the processed signal.

Tranceformer2

Tranceformer2 is a previous version of the Tranceformer plug-in. The parameters are the same with the addition of an input level control.

Earlier audio effect plug-ins

Autopan

This makes the sound move automatically between the left and right channel.

The parameters are as follows:

Parameter	Explanation
LFO Freq	This sets the speed of the panning effect.
Width	This sets the depth of the effect, that is, how far out to the left/right speaker the sound should move.
Waveform	This sets the shape of the LFO producing the effect. Sine and Triangle both produce a smooth sweep, but with different characteristics. Sawtooth creates a ramp (sweep from one speaker to the other and then a quick jump back). Pulse makes the signal jump back and forth between the speakers.
Output Level	The stereo output level of the effect.

Choirus and Choirus 2

For some computer configurations, the original Choirus effect gave rise to clicks and distorted sound. The Choirus2 effect solves this problem. It is identical to the “Choirus Classic” featurewise, but draws slightly more computer power.

Choirus is a chorus and flanger effect which adds “depth” and “animation” to a sound. It basically works as follows: The original signal is delayed and the amount of delay is continuously varied by an “LFO”. This delayed signal is then added back in with the original.

Parameter	Explanation
Time	This is the basic amount of delay applied to the signal. The larger the value, the richer the sound (up to a certain extent). For flanger types of effects, use the lower range of values.
Width	Sets the amount of variation in the delay of the signal. The larger the value, the more drastic the effect. This value should be balanced with the Time setting for optimal results.
LFO Freq	This is the speed of the LFO “sweep”. The larger the value, the faster sweep.
Feedback and Feed Bal	These control the amount of output signal re-routed back to the input of the effect. For soft and wide chorus effects, keep these values low. For flanger sounds and special fx, raise these values.
Glimmer and Glimmer 2	Progressively add more “voices” making the chorus richer and more animated. The Glimmer parameter also adjusts the stereo spread.
Out Levl	The stereo output level of the effect.

- **The Choirus plug-ins are mono in-stereo out effects.**
  When using them as insert effects for stereo channels, only the left or right channel will be processed (depending on the routing settings you make for the insert slot).

Espacial

This is a basic reverb effect with dedicated control over early reflections. As there is no Mix control, you should use this as a send effect (an insert on an FX Channel track). Please note though that the Espacial is a mono in-stereo out effect - when used on a stereo channel, only the left or right channel will be processed (depending on the routing settings you make for the insert slot).

The parameters are:

Parameter	Explanation
Size	Affects the apparent size of the simulated room.
Width	This parameter also affects the impression of the size and shape of the simulated room. It also affects the “density” and clearness of the reverb.
Time	The decay time of the reverberation.
ER Start	The start time of the Early Reflections - the first “echo” from the walls in a simulated room.
ER Width	Early Reflection “density” and clearness.
ER Gain	The balance between Early Reflections/direct sound in the input to the actual reverb. When this parameter is fully raised, no Early Reflections will be heard at all.
ER Decay	Determines the gradual attenuation of Early Reflections.
ER Outp	The level of Early Reflections in the Effect Output.
Output Level	The stereo output level of the effect.

Electro Fuzz

This is a simulation of the good old transistor distortion stomp box. It accepts a mono input and is used as an Insert or Send Effect.

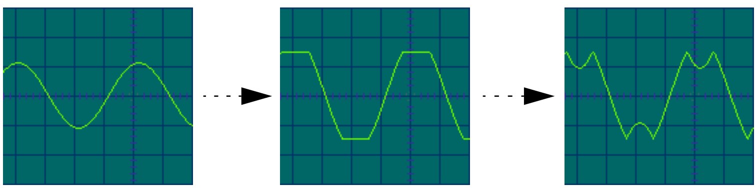
The Electro Fuzz has the following parameters:

- Boost
  This governs the amount of distortion. If you want to increase the distortion without raising the signal level, you may have to adjust the Volume knob as well.

- Clipback
  Raising this parameter will “invert” the part of the signal that is above the clipping level, instead of employing hard clipping. The result is that more 2nd order harmonics are added, changing the character of the distortion.

If you distort a sine wave, by raising the Boost parameter...

...it will be clipped like this.

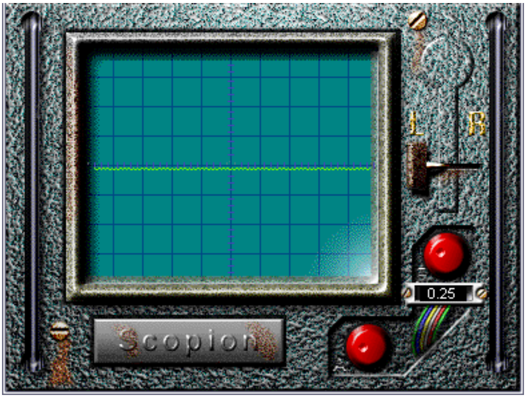


Increasing the Clipback value...

...will invert the clipped signal peaks, adding harmon-

- Volume
  This is a volume control for the output signal from the Electro Fuzz.

Scopion



The Scopion is an basic on-board oscilloscope that analyzes the left or right side of a stereo input signal and displays the waveform contents in real time. There are three parameters:

Parameter	Description
L/R Switch	Clicking this switch allows you to choose between displaying the left and right side of the stereo input signal.
Frequency	This knob (directly below the L/R switch) allows you to scale the waveform horizontally.
Amplitude	This knob (at the bottom of the Scopion window) allows you to scale the waveform vertically.

- If you click the Scopion label plate below the display, a help screen will be shown, explaining the functionality of the parameters in the window.

Stereo Echo

The Stereo Echo is a delay with separate settings for the left and right channel. It can also be used as a single mono delay, in which case the maximum delay time will be doubled.

The Stereo Echo accepts a mono input only. It is normally used as a Send Effect.

The Stereo Echo has the following parameters:

Parameter	Explanation
Delay1	The delay time for the left channel. The maximum delay time depends on the sample rate setting. If you link both channels for mono operation (see below), the maximum delay time will double.
Feedbck1	The delay feedback for the left channel. Higher values result in a higher number of echo repeats.
Link 1-2	Activating this switch turns the effect into a mono delay. When Link is on, only the left channel parameters will be available (Delay1, Feedback1, etc).
Delay 2	The delay time for the right channel.
Feedbck2	The delay feedback for the right channel.
Del2 Bal	This parameter determines how much of the left channel output is sent to the right channel input. When set to 0.0 (fully left), then none of the left channel output is added to the right channel input; when it is set to 1.0 (fully right), the right input receives both its normal source and the complete output of the left channel.
Volume L	The output level of the left channel delay.
Volume R	The output level of the right channel delay.

Stereo Wizard

The Stereo Wizard is a stereo width enhancer that takes a stereo input signal and makes it sound “wider”. StereoWizard will give best result if you use “real” stereo material (as opposed to mono channels panned to different positions in the stereo image), but you could also apply stereo ambience or reverb to a mono signal, and then use Stereo Wizard to enhance the stereo width of the reverb.

The Stereo Wizard has the following parameters:

Parameter	Explanation
Amount	Higher values result in a greater stereo width. Normally, you should set this to values between 0.00 - 0.20; higher values can be used for special effects.
Reverse	Reverses the left and right channel.

WunderVerb 3

WunderbVerb 3 is a reverb plug-in which provides natural sounding reverb effects, and still uses very little processor power.

- **The WunderVerb 3 plug-in is a mono in-stereo out effect.**
  When using it as an insert effect for a stereo channel, only the left or right channel will be processed (depending on the routing settings you make for the insert slot).

Use the Program pop-up to select one of ten Reverb Types:

Hall	The reverberation of a medium-sized hall.
Large Hall	The reverberation of a larger hall.
Large Room	The reverberation of a large room.
Medium Room	The reverberation of a medium-sized room.
Small Room	The reverberation of a very small room.
Plate	The slightly metallic effect of a plate reverb.
Gated	A special effect, where the reverb is abruptly cut off.
Effect 1	A special “bouncing” effect.
Echoes	An echo (delay) effect.
Effect 2	A special, resonant effect, suitable for “ringing” metal sounds.

You can adjust the following three parameters:

Size

This is the size of the simulated room. Changing this will affect the density and character of the reverb. If you have selected a Reverb Type where you can hear the individual “bounces” (Effect 1, Echoes, etc), raising the Size will increase the time between each “bounce”, like the time control on a delay effect.

Decay

This is the decay time for the reverb. The higher the value, the longer the reverb.

Damp

Raising this value will cause the high frequency contents of the reverb sound to die out quicker. This results in a softer, darker reverb.

Cubase 5 VST Instruments

CS40

The CS40 is a straightforward software synthesizer with the following main features:

- The CS40 is polyphonic with up to 6 voices.
- The CS40 receives MIDI in Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to the CS40.
- The CS40 responds to the following MIDI messages:
  MIDI Note On/Off (velocity governs volume).
  Volume.
  Pan.
  Pitch Bend (± 2 semitones).
  Modulation (vibrato).

CS40 Parameters

Parameter	Description
Oscillator 1 Range	Selects an octave range for oscillator 1; 32, 16, 8 or 4 feet.
Oscillator 1 Waveform	The basic waveform for oscillator 1; Triangle, Sawtooth, Square or Pulse.
Oscillator 1 Tune	Detunes Oscillator 1 ± 7 semitones.
Oscillator 2 parameters	Same as Oscillator 1.
Oscillator Blend	Adjusts the relative volume mix between oscillator 1 and 2.
LFO Speed	Governs the speed of the LFO. If LFO Sync is activated, this parameter sets the LFO speed in various beat increments to the sequencer tempo.
LFO Sync	Syncs the LFO speed to the set sequencer tempo.
LFO Amount	This governs the amount of LFO modulation applied to the destination parameters.
LFO Destination	This sets the destination parameter(s) for the LFO. You can apply modulation to the VCF cutoff frequency, the VCA amplitude, or both.
Vibrato Speed	Governs the speed of the Vibrato LFO. The Vibrato amount is controlled by the Mod Wheel.
VCF Cutoff	The Cutoff Frequency for the filter, governing the amount of high frequencies in the sound.
VCF Resonance	The Resonance control for the filter. Raise this for a more hollow, pronounced filter effect.
Filter Mod ADSR	This controls how much the VCF cutoff is affected by the VCF Envelope. Negative values invert the Envelope settings.
VCF Attack, Decay, Sustain, Release	The Filter Envelope. Use these parameters to determine how the filter should open and close with time, when a note is played.
VCA Attack, Decay, Sustain, Release	The Amplitude Envelope. Use these parameters to determine how the volume should open and close with time, when a note is played.
MonoMode	When activated the CS40 will be monophonic.
Volume	Governs the overall volume.

JX16 Synthesizer

The JX16 is a dual oscillator software synthesizer with the following main features:

- The JX16 is polyphonic with up to 16 voices.
  The polyphony setting for each patch is user programmable.
- Low CPU load and high quality sound (low aliasing distortion).
- Multimode Filter.
  Lowpass, Bandpass and Hipass filter modes are available.
- Oscillator Lock function enables the creation of pulse and square waveforms with classic PWM (Pulse Width Modulation).
  See page 36.
- Built-in stereo chorus effect.
- The JX16 receives MIDI in Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to the JX16.
- The JX16 responds to MIDI Controller messages.
  See page 40.

All parameters can be automated as described in the Operation Manual.

Osc 1+2 Section

This section contains parameters affecting both oscillators.

Parameter	Description
Octave	Tunes the oscillators in octave steps.
Fine Tune	Tunes the oscillators in cent (100th of a semitone) steps.
Vibrato	Governs how much the LFO should modulate the pitch of the oscillators (vibrato). The Vibrato parameter is also controllable via MIDI by using the Mod Wheel.
Noise	This parameter produces white noise mixed with the oscillators. By using the "OSC lock" parameter you can "cancel out" the oscillators and use pure noise as the sound source. This is described below.
OSC lock	See below.

The Oscillator 2 Section

This section contains parameters that affect oscillator 2 only.

Parameter	Description
OSC Mix	Controls the level of oscillator 2. 100 produces equal level to oscillator 1, which has a fixed output level.
Coarse	Tuning of Oscillator 2, in semitone steps.
Fine Tune	Fine tuning of Oscillator 2, in cent (=100th of a semitone) steps.
Vibrato	This lets you apply vibrato on the second oscillator only. This can be useful for creating PWM effects - see page 36 for a further description. Both positive and negative values can be set.

About the "Oscillator Lock" parameter

JX16 features two oscillators per voice, with fixed sawtooth waveforms. You can, however, generate square waves and PWM (pulse width modulation) with the JX16, by combining the two oscillators using the "OSC lock" and Oscillator 2 "Vibrato" parameters. The following applies:

- "OSC lock" allows the phase of Oscillator 2 to be fixed relative to OSC 1, producing pulse waves when Oscillator 2 has the same pitch and level as OSC 1.

- If the oscillators are tuned to the same pitch and level, an "OSC lock" setting of 50% produces a square wave with higher and lower settings producing progressively narrower pulse waveforms.
  With an "OSC lock" setting of 0% the two oscillators cancel out completely, which is useful if you only want to use the noise generator as a sound source.

- By applying the Oscillator 2 "Vibrato" parameter when OSC lock is set to around 50%, classic PWM is produced.
  You can also detune Oscillator 2 for even richer modulation effects.

- In "Free" mode the oscillator phase is allowed to drift, producing a random timbre change.

By experimenting with these parameters, many different timbres and modulation effects can be produced.

The Glide/Chorus Section

This section contains Glide parameters, and also the Polyphony and Chorus parameters.

Parameter	Description
Mode	If set to "On", the pitch will glide up or down between notes played. If set to "Held", Glide will only be applied when you press a key while another key is held.
Rate	Controls the time it takes for the pitch to glide from one note to the next when using Glide. If Bend (see below) is used, this parameter controls the time it takes for the pitch bend to "land" at the correct pitch.
Bend	Applies an initial pitch bend to the notes played. Negative values causes the pitch to slide up to the pitch of the note played, and vice versa.
Polyphony	This sets the polyphony, i.e. the number of voices a patch can use.
Chorus	This adds a stereo chorus effect. The values set different modulation rates and depths for the effect.

The LFO Section

This is where you set up the LFO (Low Frequency Oscillator). LFOs are used to modulate parameters like pitch (vibrato) or the filter cutoff.

Parameter	Description
LFO Wave	This sets the LFO waveform for modulating parameters: Sine produces smooth modulation suitable for vibrato, Square produces stepped modulation between two alternating values, Saw+/- produces ramp up/down values respectively, and Random will produce random stepped modulation.
LFO Sync	If this is activated, the LFO rate will be synced to the sequencer tempo in various beat divisions that can be set with the LFO Rate parameter.
LFO Rate	Governs the modulation rate of the LFO.
LFO Rate (tempo sync on)	If the "LFO Sync" parameter is activated, the LFO rate will be synced to the sequencer tempo, according to the different beat divisions that can be specified here.
LFO Velocity	This allows you to control the LFO Rate parameter with velocity, i.e. by how hard or soft you strike a note on the keyboard. The harder you play the faster the LFO rate.

The VCF Section

This section contains the filter parameters:

Parameter	Description
VCF Mode	Sets the filter mode to either lowpass (LP), highpass (HP), bandpass (BP) or off. The filter modes are described below this table.
VCF Freq (Cutoff)	Controls the filter frequency or "cutoff". If a lowpass 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 mode (see page 39).
Resonance	The Resonance control for the filter. Raise this for a more pronounced filter sweep effect. If set to 100, the filter will self-oscillate and produce a pitch. See the "VCF Key" parameter below for a description of how this can be used.
VCF Env	Controls how much the filter cutoff should be affected by the VCF Envelope parameters. Negative values will invert the filter envelope settings.

Parameter	Description
VCF Vel	Determines how the filter cutoff will be affected by velocity, i.e. how hard or soft you strike a key. Positive values will increase the cutoff frequency the harder you strike a key. Negative values will invert this relationship.
VCF Att/Dec/ Sus/Rel	The Filter Envelope Attack, Decay, Sustain and Release parameters. Use these parameters to determine how the filter cutoff should open and close with time, when a note is played.
VCF LFO	This controls how much the filter cutoff is modulated by the LFO (low frequency oscillator).
VCF Key	If this parameter is set to values over 0, the filter cutoff frequency will increase the further up on the keyboard you play. If set to 100, it will track the notes on the keyboard, enabling you to “play” the filter as an extra sound source, as the filter self-oscillates and produces a pitch when the resonance is set to 100.
VCF Touch	This sets the amount the VCF cutoff parameter should be affected by Aftertouch. If positive values are set, the filter cutoff is raised the harder you press. Negative values invert this relationship.
LFO Touch	This sets the amount the VCF LFO parameter should be affected by Aftertouch. If positive values are set, the modulation increases the harder you press. Negative values invert this relationship.

About the filter modes

The JX16 features a multimode filter. The various filter modes are selected with the VCF Mode parameter, and are as follows:

- Lowpass (LP)
  Lowpass filters lets low frequencies pass and cuts out the high frequencies. This is the most commonly used filter type in analog synthesizers.
- Bandpass (BP)
  A bandpass filter cuts frequencies above and below the cutoff frequency, allowing a specific range of frequencies to pass while attenuating all others.
- Highpass (HP)
  A highpass filter is the opposite of a lowpass filter, cutting out the lower frequencies and letting the high frequencies pass.

The VCA Section

This section contains the VCA Envelope parameters, governing the amplitude (volume) of the sound:

Parameter	Description
VCA Att/Dec/Sus/Rel	The VCA Attack, Decay, Sustain and Release parameters. Use these parameters to determine how the volume should change with time, when a note is played.
VCA Velocity	This determines whether the VCA Envelope should be affected by velocity, i.e. by how hard or soft you strike a note on the keyboard.

MIDI Controller Messages

The JX16 responds to the following MIDI Controller Messages:

Controller	Parameter/Value
Pitch Bend	+/- 2 Semitones
CC1 (Mod Wheel)	Vibrato
Aftertouch	Can control filter cutoff and filter cutoff modulation (by the LFO).
CC2 / CC3	Increases and decreases filter frequency, respectively.
CC7	Volume
CC16	Increase filter resonance
Program Change #	1-64

LM-9

Program switch

Volume fader (one for each drum sound).

This sets the global velocity sensitivity for LM-9.

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

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.

The LM-9 is a basic drum machine. It has the following properties:

- LM-9 is polyphonic with up to 9 voices.
- LM-9 receives MIDI in Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to LM-9.
- LM-9 responds to the following MIDI messages:
  MIDI Note On/Off (velocity governs volume).

Furthermore, all parameters can be automated as described in the Operation Manual chapter "VST Instruments".

LM-9 Parameters

Parameter	Description
Velocity	This sets the global velocity sensitivity for LM-9. The higher the value, the more sensitive LM-9 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.
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-9 comes with two sets of drum sounds; "Acoustic" and "Beat Box". Acoustic features samples of an acoustic drum kit and Beat Box features classic analog drum machine sounds. 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 value
Bass	C1
Snare	D1
Hi-Hat	F#1
O-Hi-Hat	A#1
Tom 1	D2
Tom 2	B1
Tom 3	A1
Crash	C#2
Ride	D#2

Switching the sets

Use the Program menu to switch between the two supplied drum sets, just like you switch between effect programs.

The Neon

The Neon is a simple software synthesizer. It has the following properties:

- The Neon is polyphonic with up to 16 voices.
  However, since each added voice consumes CPU power, the maximum polyphony may be limited by the speed of your computer.

- The Neon receives MIDI in Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to the Neon.

- The Neon responds to the following MIDI messages:
  MIDI Note On/Off (velocity governs volume).
  Volume.
  Pan (remember to pan the two Instrument channels hard Left/Right if you want to use MIDI Pan messages).
  Pitch Bend (± 2 semitones).
  Modulation (vibrato).

Furthermore, all parameters can be automated as described in the Operation Manual chapter "VST Instruments".

Neon Parameters

Parameter	Description
Range	Selects an octave range for the oscillators, 16, 8 or 4 feet.
Waveform	The basic waveform for the oscillators, Triangle, Sawtooth or Square.
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).
Osc 2 Detune	Allows you to detune the “second oscillator” ± 7 semitones. By setting this to a value close to “twelve o’clock”, you will get fine detuning, for a warmer, fatter sound.
VCF Cutoff	The Cutoff Frequency for the filter, governing the amount of high frequencies in the sound. On the Neon, the Cutoff control also serves as a Depth control for the Filter Envelope (VCF Attack, Decay, Sustain, Release), so that the lower the setting of the Cutoff parameter, the more will the filter be affected by the Filter Envelope.
VCF Resonance	The Resonance control for the filter. Raise this for a more hollow, pronounced filter effect.
VCF Attack, Decay, Sustain, Release	The Filter Envelope. Use these parameters to determine how the filter should open and close with time, when a note is played.
VCA Attack, Decay, Sustain, Release	The Amplitude Envelope. Use these parameters to determine how the amplitude (volume) should change with time, when a note is played.

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