Audio Effects and VST Instruments

◀ Cubase • SX/SL

MUSIC CREATION AND PRODUCTION SYSTEM

PC
VERSION

X
Built for Mac OS X

▶ steinberg

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1
# The included effect plug-ins

Introduction

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

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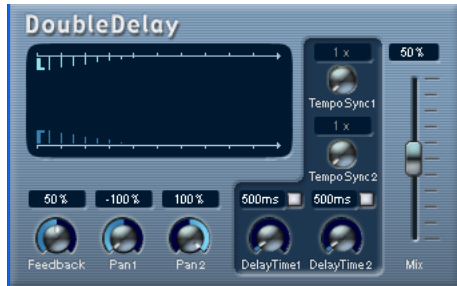
This effect provides two separate delays that can be either tempo based or use freely specified delay time settings. Cubase SX/SL automatically provides the plug-in with the tempo currently used in the project.

The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Double-Delay 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 buttons are gray) 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.

Parameter	Description
Tempo Sync 2	As above, but for the second delay unit.
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 (gray button) 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
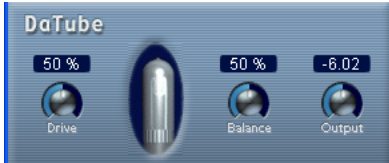
This effect emulates the characteristic warm, lush sound of a tube amplifier.

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".

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.

QuadraFuzz

QuadraFuzz is a high-quality distortion effect divided into four frequency bands allowing for control over the level both before and after distortion. This high level of control can create a very wide selection of distortion effects, ranging from subtle to extreme. The user interface consists of two windows.

• The main window features four Filterbank controls, the master Gain and Output controls and a preset selector.

• In the editor window (which is opened by clicking the "Edit" button in the lower right corner) the main feature is a frequency band display.
This is where you set the width of the frequency bands as well as their level before distortion.

How does QuadraFuzz work?

Here's a short description of the three major factors that determine how QuadraFuzz sounds, and where you find the corresponding controls:

• The signal volume control *before* distortion.
You can use the Gain control on the left side of the QuadraFuzz main window to control the overall input level of the signal that is fed into the distortion stage. The signal is split up into four frequency bands in the editor window, with adjustable width and level controls. These control the input level before distortion.

• The distortion type, based on a selectable distortion characteristic.

• The signal volume control *after* distortion.
The Output control on the right side of the QuadraFuzz main window controls the overall output level. In addition, the Filterbank controls in the same window allow you to raise or lower the output volume of each separate frequency band that was defined in the editor window.

Editing in the frequency band display

The signal is divided into four frequency bands before being passed to the distortion stage, as explained earlier. You adjust the level and width of these bands in the frequency band display.
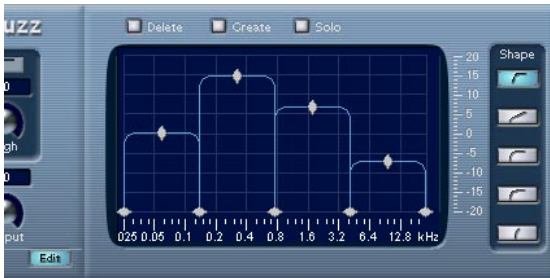
The frequency band display

Two value scales as well as a number of rhomb- and diamond-shaped handles are available.

•   The diamond-shaped handles at the bottom are used to define the corner frequencies of the different frequency bands.
•   By using the rhomb-shaped handles on top of each frequency band you determine its relative level before distortion.
•   The horizontal value scale below the Frequency band display indicates frequency. The maximum value on this scale corresponds to half the sample rate of the audio file used (Nyquist theorem).
•   The vertical value scale to the right shows the approximate level of an edited frequency band.
•   If you click and hold on one of the handles, its current value is displayed. Depending on the handle type, corner frequency or level is shown.
•   The corner frequency handles can be moved by dragging horizontally. The level handles can be moved by dragging them up or down.
•   To reset a level handle to 0 dB, hold down the [Shift] key on your computer keyboard and click on the handle.
•   If you hold down the [Ctrl]/[Command] key and move a handle, the values will change in smaller steps.
•   The "Solo" button above the frequency band display allows you to monitor individual frequency bands.
    If Solo is activated, one of the four bands is highlighted indicating the selected band. You select other bands by clicking on them.

The parameters

The following tables list all parameters available in QuadraFuzz.

The parameters in the main window are as follows:

Parameter	Description
Gain dial	This dial can be found in the lower left corner of the QudraFuzz window. You can use it to control the level of the overall input signal before distortion.
Filterbank dials: Low/Low Mid/ High Mid/High	These dials are used to control the output level of the corresponding frequency band *after* distortion. Values between +/- 12 dB can be set for each band.
Presets fader	This is used to select one of the available presets. To select a new preset, click on the fader handle and drag horizontally.
Output dial	This controls the overall output level.
Over LED	When lit, this indicates that the total input signal level exceeds 0 dB. This LED does not refer to the output level but solely to the input level before distortion. Levels above 0 dB are subject to strict limiting and cause signal clipping. As this is sometimes what you want, QuadraFuzz also offers this option.
Edit button	By clicking on this button, located in the lower right corner of the main window, you can open or close the editor window.

The parameters in the editor window are as follows:

Parameter	Description
Create	If you click on this button, a dialog will open where you can add (and name) a new preset to the preset set currently in memory. The presets are stored with the project – to make a preset available in other projects you use the File pop-up menu as usual.
Delete	This deletes the selected preset from the preset set currently in memory. If you click on the button, a dialog appears where you can confirm or cancel the action.
Solo	This mutes all frequency bands except the selected band.
Shape buttons	The available distortion characteristics (from bottom to top) create effects from a slight distortion up to a trashy hardcore sound.
Frequency band display	Here you control the level and bandwidth for the four bands, see above.

Dynamics plug-ins

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

SPL DeEsser (Cubase SX only)
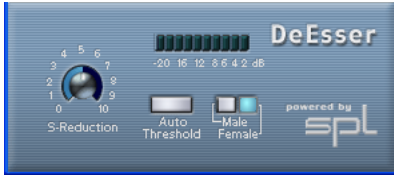
A de-esser is used to reduce excessive sibilance, primarily for vocal recordings. Basically, it is a special type of compressor that is tuned to be sensitive to the frequencies produced by the "s" sound, hence the name de-esser. Close proximity microphone placement and equal- izing can lead to situations where the overall sound is just right, but there is a problem with sibilants. Conventional compression and/or equalizing will not easily solve this problem, but a de-esser can.

The SPL DeEsser has the following parameters:

Parameter	Description
S-Reduction	Controls the intensity of the de-essing effect. We recommend that you start with a value between 4 and 7.
Level display	Indicates the dB value by which the level of the sibilant or s-frequency is reduced. The display shows values between 0 dB (no reduction) and minus 20 dB (the s-frequency level is lowered by 20 dB). Each segment in the display represents a level reduction of 2 dB.
Auto Threshold	See separate description below.
Male/Female	This sets the s-frequency and sibilant recognition to the characteristic frequency ranges of the female or male voice. The center frequency of the bandwidth at which the SPL DeEsser operates is located in the 7 kHz range for the female voice and in the 6 kHz range for the male voice.

About the Auto Threshold function

Conventional de-essing devices all have a threshold parameter. This is used to set a threshold for the incoming signal level, above which the device starts to process the signal. The SPL DeEsser however has been designed for utmost ease-of-use. With Auto Threshold on (the button is blue) it automatically and constantly readjusts the threshold to achieve an optimum result. If you still wish to determine for yourself at which signal level the SPL DeEsser should start to process the signal, deactivate the Auto Threshold button. The SPL DeEsser will then use a fixed threshold.

When recording a voice, usually the de-esser's position in the signal chain is located after the microphone pre-amp and before a compressor/limiter. This is useful, as it keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics by reacting to excessive sibilants and s-frequencies.

The Auto Threshold function keeps the processing on a constant level. The input threshold value is automatically and constantly adjusted to the audio input level. Even level differences of say 20 dB do not have a negative impact on the result of the processing. The input levels may vary, but processing remains constant.

Dynamics

Compressor

AutoGate

Limiter

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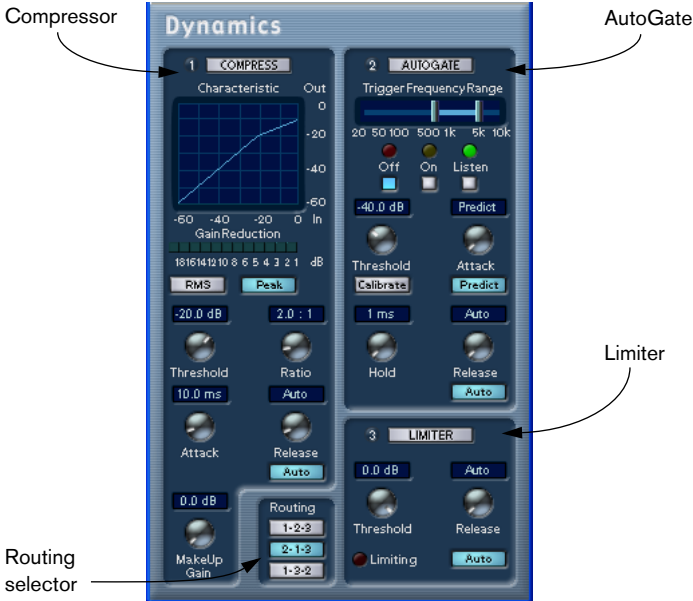
Dynamics is an advanced dynamics processor. It combines three separate processors: AutoGate, Compressor and Limiter, covering a variety of dynamic processing functions. The window is divided into three sections, containing controls and meters for each processor.

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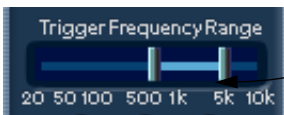
The frequency range between the two handles will be used to trigger the gate.

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)

Magneto

Magneto brings the positive qualities of analog recording to your digital system, by providing the following capabilities:

• Simulates "tape saturation" and "tape overdrive" in a very realistic manner.
• Adds warmth, punch, and brilliance to a sound.
• Allows you to emphasize the "small details" in the sound.
• Works great on bass and guitar recordings as well as on drums, including individual samples and drum loops.
• Makes sampled drums and percussion sound much more "natural" and "warm".
• Removes the "hardness" otherwise associated with digital audio recording.

All this makes Magneto suitable for processing both single sounds and complete recordings. In other words; practically any recording that you want to make sound warmer or more "natural".

The algorithm behind Magneto is based on extensive studies and measurements of analog tape recorders. Special care has been taken to transfer the results of these studies into the digital domain.

• If your audio material has been recorded digitally with Emphasis, it contains a disproportionate amount of high frequencies. This will disturb the audio analysis in Magneto.
We recommend that you convert such material (removing Emphasis) before processing it with Magneto.

About the Drive parameter and Magneto output levels

• Magneto is different from analog tape recorders in one respect: On an analog tape machine, you will get a lower output level when overdriving the tape "too far". This is known as the "saturation" effect. In Magneto, high Drive settings do not have this effect on the Output level.

• Magneto needs headroom to perform its "magic". For this reason you may note a decrease in Output level (compared to the Input level) when using very low Drive parameter settings (when the onTape meter shows levels below approximately +10dB). Since low Drive settings is not a normal situation (since the plug-in then practically doesn't have any audible effect), this is not something you would normally encounter. However, if for some application a low Drive setting is required, you can compensate for the loss in level with the Output Level parameter, see later in this text.

Metering Switch

Use the "Level" buttons to switch the meters between three modes:

• Input
In this mode, the level of the input signal is shown. This should never exceed 0dB, as mentioned above and described in more detail below.

• onTape
In this mode, the meters show an equivalent of the level recorded on the simulated "tape". See the description of the Drive parameter for more details.

• Output
This shows the output level for the entire plug-in. This should never exceed 0dB, see below.

Clip LEDs

The Input and Output Clip LEDs, located on the corresponding "Level" meter buttons, show if the signal is too "hot" (clipping occurs) at the input or output. The advantage of these is that they indicate clipping regardless of the mode the meters are switched to.

Input Level

This is used to make sure the input signal is strong enough, without exceeding full level (so that clipping is avoided).

• If your input is already normalized, or sufficiently hot, leave this knob at 0.0 dB.
• If you need to adjust the input level, switch the Level metering to Input. Then adjust the knob until the signal peaks are as close as possible to 0dB without *ever* exceeding that level!

Output Level

• Under normal conditions, the Output Level control should be left at 0.0dB. The DSP algorithm in Magneto includes an "auto-gain" function which tries to keep the output level as close as possible to 0dBfs, at high Drive settings.
• At very low drive settings (if onTape metering indicates peak levels at 7dB or less – see the Drive parameter description for more info) you might need to amplify the signal using the Output Level control. However, always do this with the Level metering Output button activated, so that you can check that clipping doesn't occur.
• At very high HF-Adjust settings, you might need to back off a bit on the Output level. Again, use Output metering to check.

If "digital clipping" occurs

If clipping occurs, (if the sound is heavily distorted), start by switching to input metering and check the input levels. If the input levels seem OK, switch to Output metering and adjust the Output Level as needed.

The main parameters

You can change the Magneto parameters in realtime – i.e. while the audio material is played back – and the changes take effect more or less immediately (depending on your system). This allows you to experiment to get a feeling for how the settings interact.

Input level, Output Level, "Level" buttons and Meters

These are used to adjust the level throughout the signal chain as described on the previous pages.

Drive

This is the main parameter. It is used to set the simulated analog tape "recording level". The value corresponds to how far above normal working level (0dB) you want to "record" on the "analog tape". For example, a setting of 7 means the "tape" is "overdriven" by 7dB.

The higher you set this, the more of the "tape saturation" effect you will get.

Please use the following guidelines:

- Start out with a Drive setting of 10dB. Then adjust to taste.
- The effect of this parameter varies drastically with the frequency content and other characteristics of the material. There is no "best setting" for all types of recordings.
- If the material you are processing is already compressed or has been recorded on analog tape, a high Drive setting is not recommended, since it will give the sound an unnatural character.
- When processing complete mixes, you will have to be more careful with the Drive settings than when processing individual recordings. If all you want is to add some "warmth" or "punch" to a complete mix, adjust the Drive setting carefully.
- Always use the onTape meter to check out the effect of the setting on the material. This meter has to go pretty far above the 0dB level for Magneto to have any audible effect on the sound. If the meter displays levels close to, or even below 0dB, you get no "overdrive" or tape saturation effect at all! If this occurs, you need to raise the Drive setting or adjust the input level.

Characteristics

This affects the tonal characteristics of the "tape saturation" effect controlled by the Drive parameter, as described above.

Tape Speed

This switches the tape simulation between 15 and 30 ips (inches per second) tape speed. There are slight differences in the harmonic character of the two. How much you will actually be able to hear of this difference depends on the frequency content of the material.

HF-Adjust

Various types of tape, recording and playback equalizers and the general design of various tape machines has an overall impact on the character of the sound. This control is used to adjust the High frequency content of the material to simulate those differences. It also has an effect on the perceived "warmth" of the sound.

This parameter can be used to compensate for the loss in high frequency that the overdrive effect introduces. Unlike on a real tape recorder it can also be used to boost the high frequency contents, compared to the original!

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.

MultibandCompressor (Cubase SX only)
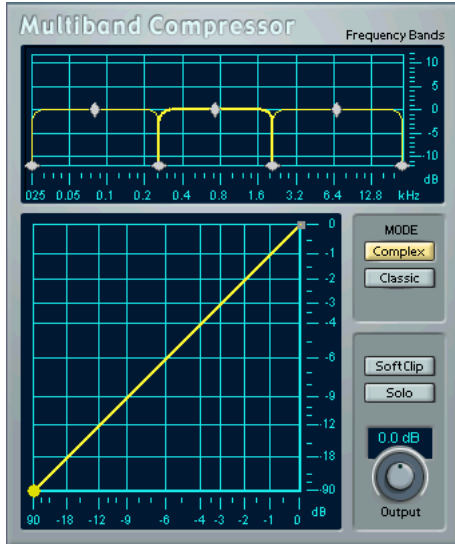
The MultibandCompressor allows a signal to be split in up to five frequency bands, each with its own freely adjustable compressor characteristic. The signal is processed on the basis of the settings that you have made in the Frequency Band and Characteristics editors. You can specify the level, bandwidth and compressor characteristics for each band by using the various controls.

The Frequency Band editor

The Frequency Band editor is where you set the width of the frequency bands as well as their level before compression. Two value scales and a number of diamond-shaped handles are available. The vertical value scale to the right gives you a clue to the approximate input gain level of each frequency band.

The diamond-shaped handles provided in the Frequency Band editor can be dragged with the mouse. You use them to set the corner frequencies and the input gain levels for up to five frequency bands. The width of each frequency band can be adjusted by dragging horizontally.

The Level handles can be moved by dragging them up or down. If you click and hold on a handle, its current value is displayed. Depending on the handle type, corner frequency or level is shown.

• The diamond-shaped handles at the bottom are used to define the corner frequencies of the different frequency bands.
• By using the diamond-shaped handles on top of each frequency band you can cut or boost the input gain by +/- 12dB before compression.
• To reset a Level handle to 0 dB, hold down the [Shift] key on your computer keyboard and click on the handle.
• If you hold down the [Shift] key and click on the corner frequency handles, they will be set to the same bandwidth (in octaves). The exact bandwidth they will be set to is dependent on the number of bands currently used.

If you hold down [Ctrl] (Win) or [Command] (Mac) and move a handle, the values will change in smaller steps.

Adding and removing frequency bands

To add a frequency band, drag the leftmost or rightmost corner frequency handle towards the middle of the window, and a new band will automatically appear (given that you have less than the maximum number of five bands active). To remove a frequency band, drag the *second* leftmost or *second* rightmost handle out of the left or right edge of the window respectively.

About the Frequency scale

The horizontal value scale below the Frequency band display indicates frequency. The maximum value on this scale corresponds to half the sample rate of the audio file used. Hence, if a 44.1 kHz soundfile is used, the highest frequency will be 22 kHz.

❐ **In the digital domain, only frequencies of up to half the sample rate used can be reproduced (Nyquist theorem). The values available in the Frequency band display do therefore depend on the sample rate of the audio material used.**

The Solo button

The Solo button in the lower right part of the MultibandCompressor panel can be used to separately monitor each of the frequency bands. This function is useful both when editing bandwidth settings and compressor characteristics.

• To select another band while solo is active, click somewhere in the (dark) area of the frequency band that you wish to monitor.

Using the Characteristics editor

By adding breakpoints and drawing curves you set the compressor characteristic. Before you start using the Characteristics editor, you have to select the frequency band you want to process. This is done in the Frequency Band editor by clicking in the area inside the frequency band.

• A selected band is highlighted for editing both in the Frequency Band and the Characteristics editors.
If you select another frequency band, the previously edited band characteristic is still shown in the Characteristics window, but it is no longer highlighted or editable until you select it again.

About breakpoints

• Clicking anywhere on the line will add a breakpoint.
• To remove a breakpoint, hold down [Shift] and click on it.
• The first breakpoint from which the line deviates from the straight diagonal will be the threshold point.
• Creating a curve in the area *below* the diagonal input/output line will cause compression. Compression decreases the output level in relation to the input level.
• Creating a curve in the area *above* the diagonal input/output line will cause expansion. Expansion increases the output level in relation to the input level.

About the Compressor type (MODE)

• Classic mode works like a standard compressor with fixed attack and release parameters.
• Complex mode features a new compression approach with a program adaptive circuit. The program adaptive compression automatically optimizes parameters according to the audio material.

The Output dial

The Output dial controls the total output level that the MultibandCompressor passes on to Cubase SX/SL. The range available is +/- 12 dB. If the SoftClip function (see below) is active, the Output dial instead controls the amount of soft clipping.

The SoftClip function

The SoftClip function is positioned at the very last stage of the internal signal path, right after the Output dial. When active, it will ensure that the total output to Cubase SX/SL never exceeds 0 dB. It works by clipping the signal gently, generating harmonics which add a warm, tube-like characteristic to the signal.

VST Dynamics

The VST Dynamics plug-in is similar to the Dynamics plug-in (see page 18), but with the following differences:

- VST Dynamics has two additional modules: Auto Level and Soft Clip.
- The signal flow is fixed, in the order AutoGate-AutoLevel-Compressor-SoftClip-Limiter.

Activating the individual processors

You activate the individual processors by clicking on their labels. Activated processors have highlighted labels. You can activate as many processors as you want, but remember that not all processors are designed to work together. For example, "Limit" and "SoftClip" are both designed to ensure that the output never exceeds 0dB, but achieves this in different ways. To have both of them activated would be unnecessary.

- To turn off all activated VST Dynamics processors, click the lit On button to the right in the panel.
Clicking the button again activates the same configuration of processors.

Auto Gate section

This is exactly the same section as the AutoGate in the Dynamics plug-in. See page 19 for details.

Auto Level section

Auto Level reduces signal level differences in audio material. It can be used to process recordings where the level unintentionally varies. It will boost low levels and attenuate high level audio signals. Only levels above the set threshold will be processed, so low level noise or rumble will not be boosted. If the input level is greater than 0dB, Auto Level will react very fast, because it "looks ahead" in the audio material for strong signal levels and can attenuate levels before they occur, thus reducing the risk of signal clipping. Auto Level has the following parameters:

Parameter	Description
Threshold	Only levels stronger than the set threshold will be processed.
Reaction time buttons (Slow, Mid, Fast)	Here you can set the amount of time it takes for Auto Level to adjust the gain. Set this according to whether the program level changes suddenly or over a length of time.

Compressor section

This is exactly the same section as the Compressor section in the Dynamics plug-in. See page 22 for details.

Soft Clip section

Soft Clip is designed to ensure that the output level never exceeds 0dB, like a limiter. Soft Clip, however, acts differently compared to a conventional limiter. When the signal level exceeds -6dB, SoftClip starts limiting (or clipping) the signal "softly", at the same time generating harmonics which add a warm, tubelike characteristic to the audio material. Soft Clip is simplicity itself to use as it has no control parameters. The meter indicates the input signal level, and thus the amount of "softclipping". Levels in the green area (weaker than -6dB) are unaffected, while levels in the yellow-orange-red area indicate the degree of "softclipping". The deep red meter area to the right indicates input levels higher than 0dB.

• **Avoid feeding Soft Clip with excessively high signal levels as audible distortion may occur, although the output level will never exceed 0dB.**

Limiter section

This is exactly the same section as the Limiter in the Dynamics plug-in. See page 23 for details.

Filter plug-ins

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

Q (Cubase SX only)

Q is a high-quality 4-band parametric stereo equalizer with two fully parametric midrange bands. The low and high bands can act as either standard shelving filters or fixed-gain high/low-cut filters.

Making settings

1. Click the corresponding On button below the EQ curve display to activate any or all of the Low, Mid 1, Mid 2 or High equalizer bands.
   When a band is activated, a corresponding eq point appears in the EQ curve display.

2. Set the parameters for an activated EQ band.
   This can be done in several ways:

• By using the knobs.
• By clicking a value field and entering values numerically.
• By using the mouse to drag points in the EQ curve display window.
   By using this method, you control both the Gain and Frequency parameters simultaneously. The knobs turn accordingly when you drag points. In addition, if the Mid 1 and Mid 2 bands (M1 and M2) are activated there will be two points on each side of the Gain/Frequency point that control the width (Q) parameter.
   If you press [Shift] while dragging, values can be set in finer increments.

Parameters

Parameter	Description
Low Freq (20-2000Hz)	This sets the frequency of the Low band.
Low Gain (-20 to +20 dB)	This sets the amount of cut/boost for the Low band.
Low Cut	If this button is activated for the Low band, it will act as a Low Cut filter. The Gain parameter will be fixed.
Mid 1 Freq (20-20000Hz)	This sets the center frequency of the Mid 1 band.
Mid 1 Gain (+/- 20dB)	This sets the amount of cut/boost for the Mid 1 band.
Mid 1 Width (0.05-5.00 Octaves)	This sets the width of the Mid 1 band, in octaves. The lower this value, the "narrower" the bandwidth.
Mid 2 Freq (20-20000Hz)	This sets the center frequency of the Mid 2 band.
Mid 2 Gain (-20 to +20 dB)	This sets the amount of cut/boost for the Mid 2 band.
Mid 2 Width (0.05-5.00 Octaves)	This sets the width of the Mid 2 band, in octaves. The lower this value, the "narrower" the bandwidth.
High Freq (200-20000Hz)	This sets the frequency of the High band.
High Gain (-20 to +20 dB)	This sets the amount of cut/boost for the High band.
High Cut	If this button is activated for the High band, it will act as a High Cut filter. The Gain parameter will be fixed.
Output (-20 to +20 dB)	This parameter allows you to adjust the overall output level.
Left/Stereo/Right/Mono Modes	For stereo signals you can set independent curves for the left and right channels by clicking the corresponding button. If the Stereo mode is activated, the curve will be applied to both channels. When channel independent curves have been set, the left/right channel curves will be colored green and red, respectively. The currently non-selected channel is shown with a dotted curve. If you activate Stereo mode after independent curves have been set, the currently active curve will be applied to both channels. Mono mode is automatically activated for mono signals and is otherwise unavailable.

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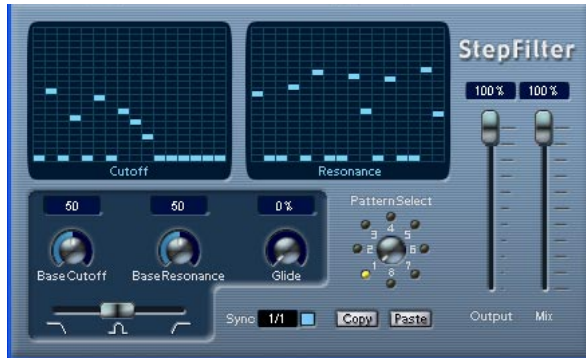
StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

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.
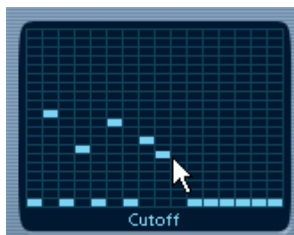
Setting filter cutoff values in the grid window.

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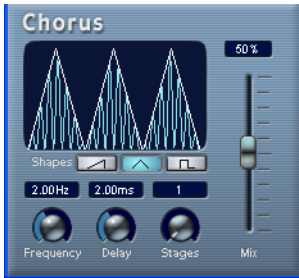
The Chorus plug-in adds short delays to the signal, and pitch modulates the delayed signals to produce a "doubling" effect.

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 yellow when tempo sync is on, and gray when it is off.
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
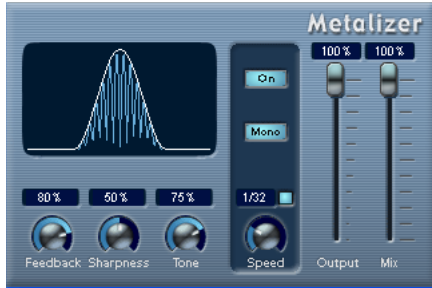
The Metalizer feeds the audio signal through a variable frequency filter, with tempo sync or time modulation and feedback control.

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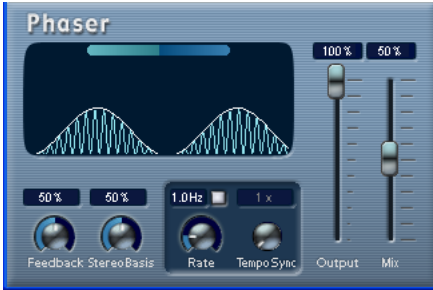
The Phaser plug-in produces the classic "swooshing" sound that characterizes phasing. It works by shifting the phase of the signal and adding it back to the original signal, causing partial cancellation of the frequency spectrum.

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 yellow when tempo sync is on, and gray when it is off.
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.
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.

Parameter	Description
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 .
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.
Low Rotor Amp Mod.	Adjusts amplitude modulation depth.

Parameter	Description
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 yellow when tempo sync is on, and gray when it is off.
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.

Parameter	Description
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 yellow when tempo sync is on, and gray when it is 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 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 (yellow button) or off (gray button).
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.

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.

Apogee UV 22 HR (Cubase SX only)

The UV22 HR is a dithering plug-in, based on an advanced algorithm developed by Apogee (for an introduction to the concept of dithering, please refer to the chapter "Audio Effects" in the Operation Manual). You can use the UV22 HR plug-in for all dithering situations, except when working with surround audio. This is because the UV22 HR is a standard "stereo in" – "stereo out" plug-in (as opposed to the SurroundDither plug-in, see page 70).

The following options can be set in the UV 22 HR control panel:

Option	Description
Normal	Try this first, it is the most "all-round" setting.
Low	This applies a lower level of dither noise.
Autoblack	When this is activated, the dither noise is gated (muted) during silent passages in the material.
Bit Resolution	The UV22 HR supports dithering to multiple resolutions: 8, 16, 20 or 24 bits. You select the desired resolution by clicking the corresponding button.

❐ **Dither should always be applied post output bus fader.**

Apogee UV22

The UV22 is a stereo in-stereo out dithering plug-in, based on an advanced algorithm developed by Apogee (for an introduction to the concept of dithering, please refer to the chapter "Audio Effects" in the Operation Manual). The UV22 plug-in always dithers to 16 bit resolution, which means you should only use it when exporting to 16 bit audio format.

The following options can be set in the control panel:

Option	Description
Normal	Try this first, it is the most "all-round" setting.
Low	This applies a lower level of dither noise.
Autoblack	When this is activated, the dither noise is gated (muted) during silent passages in the material.

❒ **Dither should always be applied post output bus fader.**

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 .

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 .

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 .

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.

SMPTE Generator (Cubase SX only)
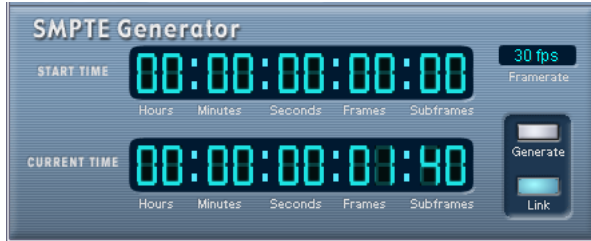
This plug-in is not an effect device. It sends out SMPTE time code to an audio output, allowing you to synchronize other equipment to Cubase SX (provided that the equipment can sync directly to SMPTE time code). This can be very useful if you don't have access to a MIDI-to-time code converter.

The following items and parameters are available:

• Generate Button
Activate this to make the device generate SMPTE time code.

• Link Button
This synchronizes the time code output to the Transport time positions.
When Link is activated, the time code output will exactly match the play position in Cubase SX.
Activating the Generate button makes the device send the SMPTE time code in "free run" mode, meaning that it will output continuous time code, independently from the transport status in Cubase SX. If you wish to "stripe" a tape with SMPTE, you should use this mode.

• Start Time
This sets the time at which the SMPTE Generator starts, when activated in "free run" mode (Link button off). To change the Start time, click on a digit and move the mouse up or down.

• Current Time
When Link is on this shows the current position in Cubase SX. If Link is off it shows the current time of the SMPTE Generator in "free run" mode. This cannot be set manually.

- Framerate
  This defaults to the frame rate set in the Project Setup dialog. If you wish to generate time code in another frame rate than the Project is currently set to (for example to stripe a tape), you can select another format on the Framerate pop-up (provided that "Link" is off).
  Note, however, that for the other device to synchronize correctly with Cubase SX, the framerate has to be the same in the Project Setup dialog, the SMPTE Generator and in the receiving device.

Example - Synchronizing a device to Cubase SX

Proceed as follows:

1. Connect the SMPTE Generator as an insert effect on an audio channel, and route the output of that channel to a separate output.
   Make sure that no other insert or send effects are used on the time code channel. You should also disable EQ, if this is active.

2. Connect the corresponding output on the audio hardware to the time code input on the device you wish to synchronize to Cubase SX.
   Make all necessary settings in the other device, so that it is set to synchronize to incoming timecode.

3. Adjust the level of the time code if needed, either in Cubase SX or in the receiving device.
   Activate Generate button (make the device send the SMPTE time code in "free run" mode) to test the level.

4. Make sure that the frame rate in the receiving device matches the frame rate set in the SMPTE Generator.

5. Activate the Link button.
   The SMPTE Generator will now output time code that matches the position of the Cubase SX Transport panel.

- Press Play on the Cubase SX Transport panel.
  The other device is now synchronized and will follow any position changes set with the Cubase SX transport controls.

Reverb plug-ins

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

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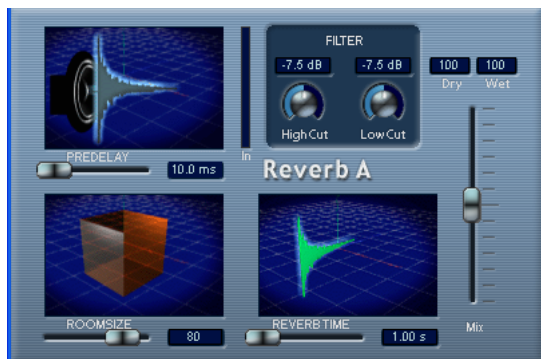
Reverb A is a reverb plug-in which provides smooth, dense reverb effects. Reverb A has the following parameters:

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.

Surround plug-ins (Cubase SX only)

This section describes the plug-ins in the "Surround" category.

Mix6To2 (Cubase SX only)

The Mix6To2 effect allows you to control the levels of up to six surround channels, and to mix these down to a stereo output. The pop-up menu contains a number of speaker arrangement presets that correspond to some default surround formats. The Mix6To2 lets you quickly mix down your surround mix format to stereo, and to include parts of the surround channels in the resulting mix.

• Note that Mix6To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. Also note that the Mix6To 2 should be placed in one of the post fader insert effect slots for the output bus.

Each of the surround channels has the following parameters:

• Two volume faders that govern the levels of the surround bus to the left and right side of the (master) bus.
• A Link button that links the two volume faders.
• Two Invert buttons allow you to invert the phase of the left and right side of the surround bus.

The Master bus has the following parameters:

• A Link button that links the two Master faders.
• A Normalize button. If activated, the mixed output will be normalized, i.e. the output level will automatically be adjusted so that the loudest signal is as loud as possible without clipping.

SurroundDither (Cubase SX only)

SurroundDither is not an "effect" as such. Dithering is a method for controlling the noise produced by quantization errors in digital recordings. The theory behind this is that during low level passages, only a few bits are used to represent the signal, which leads to quantization errors and hence distortion. For example, when "truncating bits", as a result of moving from 24- to 16-bit resolution, quantization errors are added to an otherwise immaculate recording. By adding a special kind of noise at an extremely low level, the effect of these errors is minimized. The added noise could be perceived as a very low-level hiss under exacting listening conditions. However, this is hardly noticeable and much preferred to the distortion that otherwise occurs.

When should I use SurroundDither?

• Basically anytime you mix down to a lower resolution, either in real-time (playback) or with the Export Audio Mixdown function, you should consider dithering.

• Since SurroundDither is capable of dithering up to six channels at the same time, it is recommended if you're using surround channels.
  If not, you may want to use the UV22 HR instead, see .

  The following options can be set in the SurroundDither control panel:

Dithering Type

There are no hard and fast rules for the following options, it all depends on the type of material you are processing. We recommend that you experiment and let your ears be the final judge:

Option	Description
Off	No dithering is applied.
Type 1	Try this first, it is the most "all-round" type.
Type 2	This method emphasizes higher frequencies more than Type 1.

Noise Shaping Options (Off, Type 1 - 3)

This parameter alters the character of the noise added when dithering. Again, there are no fixed general rules, but you may notice that the higher the number selected here, the more the noise is moved out of the ear's most sensitive range, the mid-range.

Ditherbits

This is used to specify the intended bit resolution for the final result.

• The section has six buttons, one for each channel.

• Above each button there are six corresponding value fields that display the bit resolution the files will be converted to.
Clicking a button several times cycles through the available bit resolution values.

An Example

Say you have set up a project to record 24-bit files. After completion, you want to create a digital 16-bit master for CD burning. Proceed as follows:

1. Add SurroundDither to a post fader insert effect slot for the output bus.
I.e. in one of the last two slots.

2. Open the control panel for SurroundDither, and select the Dithering and Noise Shaping Type.

3. Set the Ditherbit destination to "16" for all the master mix outputs currently used, as defined in the VST Connections dialog.
If you are not using Surround channels, this will be Channel 1 and 2.

4. When you now play back the Project, the digital outputs of your audio hardware will output the mix with 16-bit resolution, with dithering applied.

SurroundPan (Cubase SX only)

The SurroundPan plug-in provides a graphical overview representing the speaker arrangement and the sound source, allowing you to dynamically position the audio in the surround field.

This plug-in is described in detail in the Operation Manual chapter "Surround Sound".

2

The included VST Instruments

A1 Synthesizer

The A1 is a dual oscillator software synthesizer with the following main features:

- The A1 is polyphonic with up to 16 voices.

- Multimode filter.
  Lowpass, bandpass, highpass and notch filter types are available.

- PWM (Pulse Width Modulation).

- FM (Frequency Modulation).

- Ring Modulator.

- Built-in stereo chorus/flanger effect.

- The A1 receives MIDI in Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to the A1.

- The A1 responds to MIDI Controller messages.
  See .

A1 Parameters

Oscillator 1 and 2 section

This section contains parameters affecting the oscillators.

Parameter	Description
Octave	Clicking on the outer ring of the dial allows you to tune the oscillator in octave steps.
Semitone	Clicking on the inner ring of the dial allows you to tune the oscillator in semitone steps.
Detune	Tunes the oscillator in cent (100th of a semitone) steps.
Shape	This sets the waveform for the oscillator (sine, triangle, sawtooth or pulse).
PW	Sets the width of the waveform when a Pulse waveform is selected. Turning the dial clockwise gradually produces a narrower pulse waveform. Note that a PW setting of 100% will lead to complete cancellation of the waveform (i.e. silence), if no modulation (see PW Mod) is applied.
PW Mod	This parameter determines the amount of Pulse Width Modulation (PWM) by the LFO. Positive and negative values can be set. A Pulse waveform must be selected for PW Mod to function.
Pitch Mod	This parameter determines the amount of oscillator 1 pitch modulation (or vibrato) by the LFO. Positive and negative values can be set.
FM (Oscillator 1 only)	Governs the amount of frequency modulation. See page 84.
FM Env (Osc 1 only)	This governs how much the Filter Envelope parameters affects the FM amount. Positive and negative values can be set. See page 84.

LFO section

This section contains the LFO (Low Frequency Oscillator) parameters. LFOs are used to modulate parameters like pitch (vibrato) or the filter cutoff.

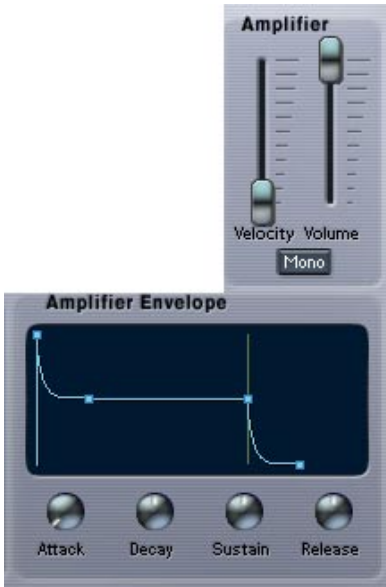| Parameter	Description
LFO Wave	This sets the LFO waveform for modulating parameters: Sine and triangle waves have a smooth waveform, suitable for normal vibrato. Saw produces a ramp up or down cycle. S&H produces stepped random modulation. Square waves produce cycles that abruptly change between two values. Random produces smooth random modulation.
LFO Sync	If this is activated, the LFO rate will be synchronized to the sequencer tempo in various bar/beat divisions that can be set with the LFO Speed parameter.
LFO Speed	Governs the modulation rate of the LFO.
LFO Speed (tempo sync on)	If the "LFO Sync" parameter is activated, the LFO rate will be synchronized to the sequencer tempo, according to the different beat divisions that can be specified here.

Filter section

This section contains the filter parameters:

Parameter	Description
Filter type	Sets the filter type to either lowpass, highpass, bandpass or notch. The filter types are described on page 83.
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 83).
Resonance	The Resonance control for the filter. Raise this for a more pronounced filter sweep effect.
Drive	This parameter can overdrive the filter to produce distortion effects.
Filter Envelope	Controls how much the filter cutoff should be affected by the Filter Envelope. Negative values will invert the filter envelope settings.
Filter Velocity	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.
Filter Envelope Attack/Decay/ Sustain/ Release	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. Values can be changed using the dials or by dragging the breakpoints in the graphic display.
Cutoff Mod	This controls how much the filter cutoff is modulated by the LFO (low frequency oscillator).
Keytrack	If this parameter is set to values over 0, the filter cutoff frequency will increase the further up on the keyboard you play. Negative values invert this relationship.

Amplifier section

This section contains the Amplifier parameters:

Parameter	Description
Amplifier Attack/ Decay/Sustain/ Release	The Amplifier Attack, Decay, Sustain and Release parameters. Use these parameters to determine how the volume should change with time, when a note is played. Values can be changed using the dials or by dragging the breakpoints in the graphic display.
Velocity	This determines how much the Amplifier Envelope should be affected by velocity, i.e. by how hard or soft you strike a note on the keyboard.
Mono	When this is activated, the A1 will be monophonic, i.e. only play one voice at a time.

The Chorus/Flanger section

Adding chorus will introduce a wide stereo effect and generally "fatten" sounds. With higher Feedback settings, more metallic sounding flanging effects are produced. The section contains the following parameters.

Parameter	Description
Speed	Controls the modulation rate of the effect.
Feedback	Increasing the Feedback parameter value results in a more pronounced sweeping metallic sound. Positive and negative feedback values can be set.
Depth	Controls the depth of the modulation.
Quad	Adds more delay taps, producing richer chorus/flanger effects.
On	This turns the chorus/flanger effect on or off.

The Glide section

This section contains the glide parameters.

Parameter	Description
On	If set to "On" the pitch will glide up or down between notes played.
Speed	Controls the time it takes for the pitch to glide from one note to the next when using Glide.

The Mixer section

This section controls the relative levels of Oscillator 1 and 2. Here you also set the levels of the Ring Modulator and Noise Generator outputs.

| Parameter	Description
Osc 1	Sets the volume of oscillator 1.
Ring Mod	Controls the level of the ring modulator. See page 83.
Osc 2	Sets the volume of oscillator 2.
Noise	Noise is commonly used to create wind and percussion type sounds. To hear the noise generator output on its own, turn down the osc 1 and 2 output in the Mixer.

Mod Wheel section

This section controls how the modulation wheel affects certain parameters. Positive and negative values can be set. For example, this can be used to set up so that moving the mod wheel gradually removes Filter Cutoff LFO modulation and instead introduces vibrato.

Parameter	Description
Pitch Mod	Governs the amount of LFO modulation of the oscillator frequency (vibrato) using the mod wheel.
Cutoff Mod	Governs the amount of LFO modulation of the Filter Cutoff parameter using the mod wheel.
Cutoff	Governs how much the mod wheel affects the Filter Cutoff frequency. Positive values raise the cutoff frequency when moving the mod wheel forward. Negative values invert this relationship.

The Parameter display
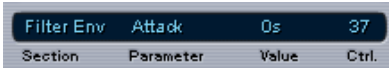
The Parameter display is located in the middle of the A1 panel. The Parameter display shows information about any A1 parameter control when you point at it with the mouse. The information is presented in the following way (from left to right):

- "Section" shows what A1 section the parameter belongs to.
- "Parameter" shows the name of the parameter.
- "Value" shows the current value of the parameter.
- "Ctrl" shows the MIDI Controller number assigned to the parameter, see page 84.

Setting the number of voices

A1 can have up to 16 voices, but you can freely set the number of voices for each program by changing the value in the "Voices" field.

Keyboard section

The keyboard shows incoming MIDI note data as played by "invisible hands". The keyboard can be "played" by clicking on it with mouse. Note that the velocity produced will be fixed and that you cannot record anything by clicking the keyboard.

- "Bend Range" is the only parameter that can be set in this section.
  A value of "1" equals a semitone bend range, "2" equals a range of two semitones etc.

About the Filter types

The A1 features a multimode filter. The various filter modes are selected with the Filter Type buttons, and are as follows:

• Lowpass
Lowpass filters let low frequencies pass and cuts out the high frequencies. This is the most commonly used filter type in analog synthesizers.

• Bandpass
A bandpass filter cuts frequencies above and below the cutoff frequency, allowing a specific range of frequencies to pass while attenuating all others.

• Highpass
A highpass filter is the opposite of a lowpass filter, cutting out the lower frequencies and letting the high frequencies pass.

• Notch
A notch filter cuts out frequencies in a narrow midrange band, letting the frequencies below and above through.

Filter Slope

You can also select between 12 or 24 dB filter slopes for all filter types. A 12 dB Lowpass filter leaves more of the harmonics in the filtered sound compared to a 24 dB Lowpass filter.

Ring Modulator

Ring modulators basically multiply two audio signals together. In the A1, Oscillator 1 is multiplied with Oscillator 2 to produce sum and difference frequencies. Ring modulation can be used to create complex, bell-like sounds.

• To hear the output of Ring Modulator on its own, turn down the osc 1 and 2 output in the Mixer.

• If the oscillators are tuned to the same frequency, and no modulation is applied to either the oscillator 1 or 2 frequency, the ring modulated output will sound fairly similar to the "normal" sound of the oscillators. It is when the frequencies of osc 1 and osc 2 *differ*, that you get the more complex timbres associated with ring modulation.

About FM

Frequency Modulation, or FM, is when the frequency of one oscillator (called the "carrier") is modulated by the frequency of another oscillator (called the "modulator"). Using FM can produce a wide range of harmonic and non-harmonic timbres.

• In the A1, Oscillator 1 is the carrier and Oscillator 2 the modulator.
When using FM, you should turn the master volume for Oscillator 2 down to zero in the Mixer to hear the "pure" sound of FM. The output of oscillator 2 is internally routed to oscillator 1 anyway when using FM.

• Changing the frequency of Oscillator 2 also changes the timbre of the FM sound.
The waveform selected for both oscillators also affects the timbre.

MIDI Controller Messages

The A1 responds to MIDI Controller Messages. All A1 parameters are assigned controller numbers. To find out what controller number is assigned to a parameter, simply point at the parameter and you can see the associated controller number assigned to it in the Parameter display (see).

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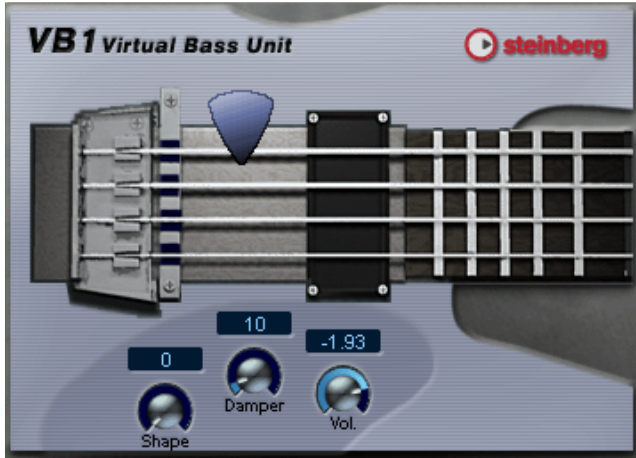
The VB-1 is a virtual bass instrument built on real-time physical model-ling principles. It has the following properties:

• 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.

3

**VST plug-ins from previous
Cubase versions**

Introduction

This chapter describes the VST effects in the "Cubase 5 Plugins" and "Earlier Plugins" subfolders on the effect pop-up menu, and the VST Instruments in the "Cubase 5 Instruments" subfolder on the instruments pop-up menu. These are plug-ins from old versions of Cubase VST, 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 wish to apply it on several channels at once, you can use it as an insert effect on a group channel and then 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:

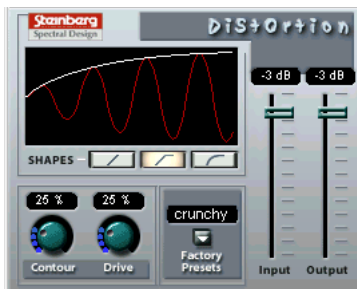| Parameter	Description
Frequency	This slider controls the speed of the LFO. The farther to the right you drag the slider, the faster the oscillation will be.
Modulation	Use this slider to control how the speed of the LFO should be modulated by the Envelope Generator (and thus by the level of the input 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 (see) 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 each of 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. For parameter descriptions, see .
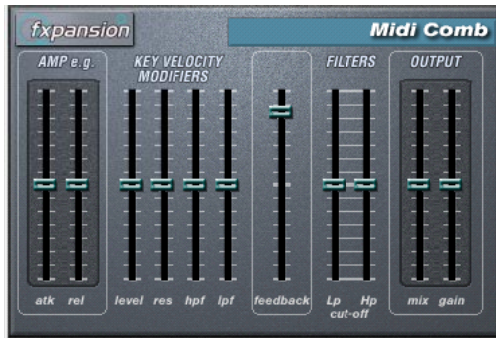
MIDIComb

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. Click in the Output column for the MIDI Track.

The Output pop-up menu appears, with the MIDI Comb as one of the items.

5. Select the MIDI Comb from the Output pop-up menu.

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.

6. 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 Gate 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
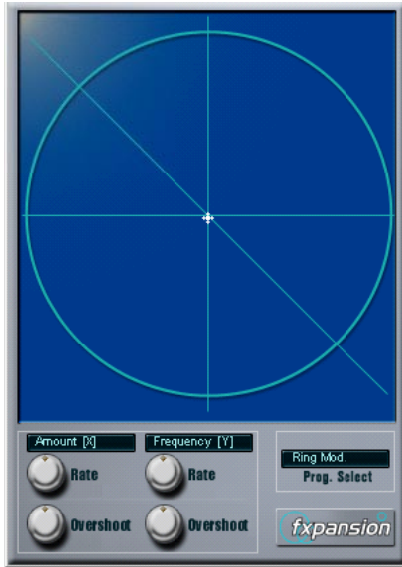
The Mysterizer is a multi-effect plug-in with a unique hands-on user interface. It can be used as an insert effect or a send effect, and allows you to choose between eight different effects. For each effect, you can control two parameters by clicking and dragging in the display, allowing for continuous 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.
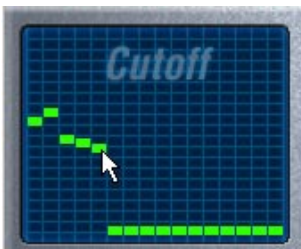
Setting filter cutoff values in the grid window.

- 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

This is an earlier version of the Reverb B plug-in, included for reasons of compatibility. See page 68 for parameter descriptions.

Reverb 32

This is an earlier version of the Reverb A plug-in. See page 67 for parameter descriptions.

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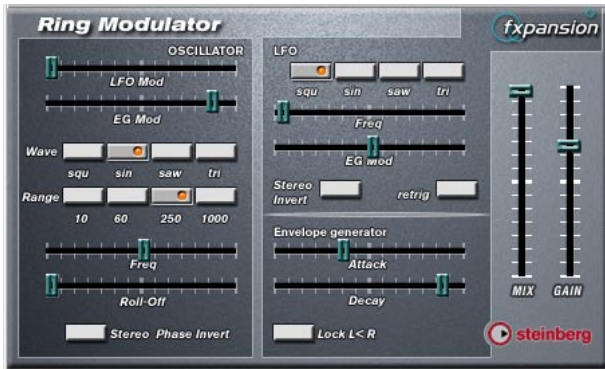
This is an earlier version of the Ringmodulator plug-in (see), with slightly different panel layout and parameters.

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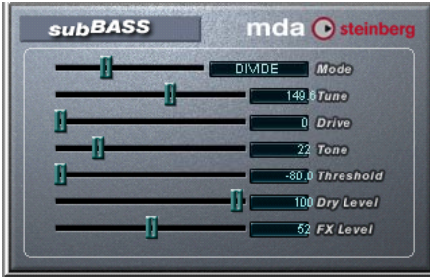
The Ring Modulator has a built-in oscillator that is multiplied with the input signal to produce the effect.

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 (see page 54) 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.

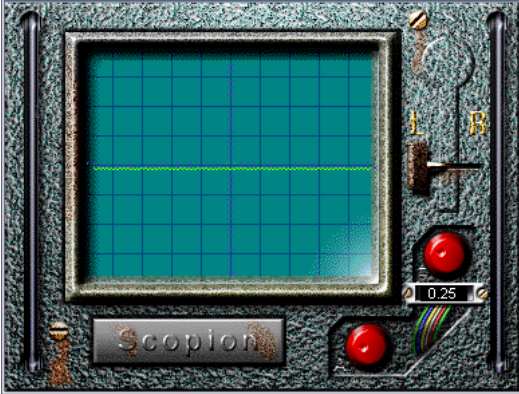If you distort a sine wave, by raising the Boost parameter...   ...it will be clipped like this.
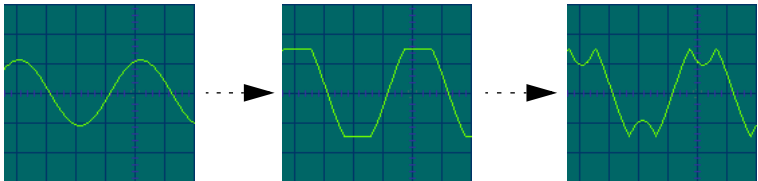
Increasing the Clipback value...   ...will invert the clipped signal peaks, adding harmonics.

- 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 is 500 ms, unless you link both channels for mono operation, in which case the maximum delay time is 1000 ms - see below (1000ms = 1 second).
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 .

- 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 .

❒ **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 127 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 130).
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.
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:
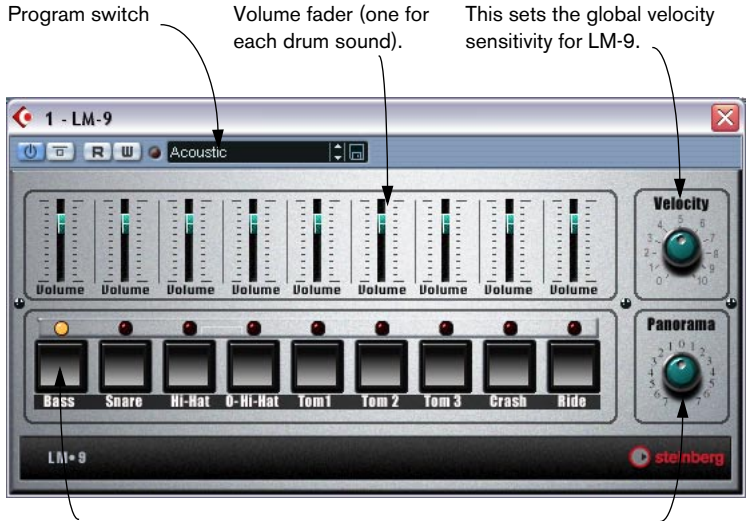
Parameter	Description
VCA Att/Dec/Sus/Rel	The VCA Attack, Decay, Sustain and Release parameters. Use these parameters to determine how the volume should change with time, when a note is played.
VCA Velocity	This determines whether the VCA Envelope should be affected by velocity, i.e. by how hard or soft 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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