




Logic Pro 7

Plug-In Reference

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Introducing Logic's Plug-ins

The professional Logic music and audio production software features a comprehensive collection of powerful plug-ins.

These include; innovative synthesizers, high quality effect plug-ins and authentic recreations of vintage instruments. Logic also supports the use of Audio Unit plug-ins in Mac OS X and also supports TDM plug-ins for users of TDM systems.

Given a fast enough computer, you could conceivably arrange and mix an entire song using several software instruments, such as Logic's ES1, ES2, EVP88, or EXS24, amongst others. These instruments have the added benefits of superior sound quality and timing as the audio signal never leaves the digital domain, and you can freely edit these software instrument parts, change the tempo and more, right up to the final mix.

Don't worry if you're unfamiliar with the terminology used here—this manual will explain everything. It covers all of the general things you need to know about plug-ins and will introduce you to the individual effects and instruments and their parameters. We've included a few tutorial chapters, which will explain how to program sounds using several of Logic's instrument plug-ins.

Using plug-ins is much easier if you are familiar with some of Logic's basic functions. You should be acquainted with Logic's Audio Mixer before going further. Information about it can be found in the Audio Mixer section of the Logic reference.

The *Bounce* buttons found on the Master Audio Objects allow you to write submixes of plug-in tracks—as an audio file—to disk at any time. For details please refer to the Logic reference.

Whatever you play on your instruments can be recorded by simply pressing Logic's Record button. Your performances can be freely edited in any of Logic's MIDI editors. Further details about this can be found in the Logic reference

Logic's plug-ins include the following features:

- Real-time processing of audio.
- Support for sample rates up to 192 kHz.
- AltiVec optimizations for the Power Macintosh G4 and G5 processors which increase the number of software effects and instruments that can be run simultaneously.
- A sophisticated, intuitive, real-time graphical editing interface for most Logic plug-ins.
- A consistent window interface for Logic, Audio Unit and TDM plug-ins.
- The ability to save and load individual plug-in effect and instrument settings or entire channel strip configurations, including those from Apple's *GarageBand* application.
- Almost all plug-in parameters can be automated via Logic's total recall mix automation system.

About This Manual

This guide covers all areas of plug-in usage in Logic. All plug-in parameters are discussed in detail.

The Basics section discusses the most essential aspects of plug-in usage, the Plug-in window interface and global plug-in commands and menus.

The Instruments and Effects chapter covers the differences between effect and instrument plug-ins.

Ensuing chapters discuss the parameters of individual plug-in effects and instruments. The instrument chapters include a number of tutorials that will help you to make the most of your new instrument.

The Onscreen Help system—accessible from Logic's Help menu—is fundamentally the Reference Manuals in electronic form. It has the advantage of being at your fingertips when you need it, and is also searchable.

Even if you're the type who just doesn't like reading manuals, we ask that you read the next section. It will provide you with essential information on the basic use of Logic's plug-ins.

Please note that all topics described herein were accurate at the date of printing. For up to date information on changes or additions made after printing, please refer to the *Late Breaking News* on the Logic DVD, and/or to the *Update Info*, included with each Logic update.

Conventions of this Guide...

Before moving on to the Basics section, we'd like to cover the following conventions used in this manual.

Menu Functions

For functions that can be reached via hierarchical menus, the different menu levels are described as follows: *Menu > Menu entry > Function*.

Important Entries

Some text will be shown as follows:

Important: Information on function or parameter.

These entries discuss a key concept or technical information that should, or must, be followed or taken into account. Please pay special attention to these entries.

Notes

Some sections provide additional information or tips that will assist your use of the effect or instrument plug-in. These are displayed as shown below:

Note: Information on function or parameter.

Key Commands

Several plug-in functions can be activated or accessed with key commands—computer keyboard shortcuts. The key commands mentioned in this guide are based on the standard Key Command Set, assigned by the Logic Setup Assistant. Where possible, we have also included the standard Key Commands for PowerBook users. These are based on the PowerBook Key Command Set, assigned in the Logic Setup Assistant.

This chapter covers all important steps required for plug-in use in Logic.

The steps include:

- Inserting, deleting, and bypassing plug-ins.
- Operating plug-ins in the Plug-in window.
- Managing plug-in settings.
- Automating plug-ins.

Using Plug-ins

Inserting and Deleting Plug-ins

Plug-ins can be either; software instruments, which respond to MIDI note messages, or audio effects, which do not respond to MIDI note messages.

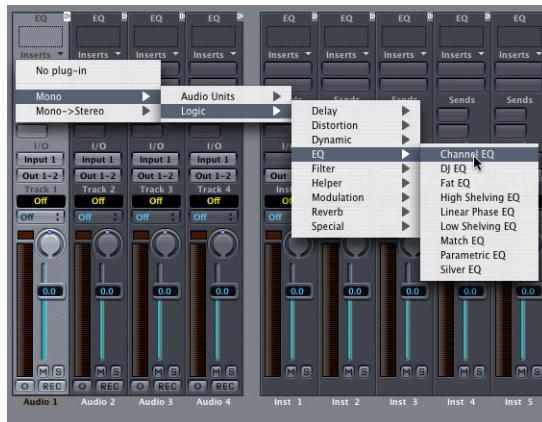
- All plug-ins can be added via the plug-in menu of an Audio Object.
- Effect plug-ins can be inserted into the *Insert* slots of all Audio Objects.
- Software-based instruments can only be inserted into special Audio Objects, called Audio Instruments. These Audio Instrument Objects have a special *Instrument* slot, directly above their Output slots.

To add a plug-in:

- 1 Click-hold on an Audio Object's Insert/Instrument slot.



- 2 The plug-in-menu appears, showing all available plug-ins. Move the mouse through the different levels of the hierarchical menu and choose a plug-in name, then release the mouse button.



The Plug-in window is launched automatically. If you do not want the Plug-in window to open automatically after insertion, uncheck the *Preferences > Audio > Display > Open Plug-in window on insertion* preference.

You can open a closed Plug-in window by double-clicking on an assigned Insert/Instrument slot.

You can set all plug-in parameters in the Plug-in window. For further information please read “The Plug-in Window” on page 16. Closing the Plug-in window leaves the plug-in active.

To remove a plug-in:

- 1 Click-hold the corresponding Insert/Instrument slot.
- 2 The plug-in menu is opened. Select the *No Plug-In* menu option.

Inserting Mono/Stereo Plug-ins

You can insert mono and stereo effects into Logic's mono objects. If you use a stereo effect in a mono object, the plug-in menu is limited to stereo effects from this insert point onwards.

Note: In general, stereo effects require twice as much processing power as their mono counterparts.

In stereo objects, the plug-in menu only shows effects with stereo inputs and stereo outputs. If you hold the Option key while opening the plug-in menu on stereo objects, you can also select mono effects.

Logic automatically inserts conversion modules (in the background) to handle stereo → mono and mono → stereo transitions. This enables you to use plug-ins in any order. Please keep the following in mind when doing so:

- These conversion modules require extra processing power.
- During a stereo → mono conversion, all spatial information is lost.
- During a mono → stereo conversion, no spatial information is added—the same mono signal is sent to both outputs.

Bypassing Plug-ins

If you want to deactivate a plug-in, but don't want to delete it, you can bypass it. Bypassed plug-ins do not drain system resources.

To bypass a plug-in:

- Option-click the appropriate plug-in insert/instrument slot on the desired Audio Object.

The insert slot of the bypassed plug-in turns from blue to gray, indicating that the plug-in is currently bypassed.

You can also use bypass a plug-in from within the Plug-in window. Further information on this can be found in the following section.

The Plug-in Window

Hands-on operation of plug-ins is performed in the Plug-in window. This window allows access to all plug-in parameters. The Plug-in window can be opened by double-clicking on the blue plug-in label on an Audio Object. Each instance of a plug-in has its own Plug-in window, allowing each to have discrete settings.

Operation of Built-in Plug-ins

Adjusting Parameters

To toggle a Plug-in window's buttons:

- Click on the button. It toggles to the next/previous option, or will be enabled/disabled.

To adjust a slider:

- Click-hold anywhere on the slider and drag up/down or left/right.

To adjust rotary knobs:

- Click-hold on the center of the rotary knob and drag the mouse up and down. You can also move the mouse in a circular motion. Fine-tuning of values is easier when using a larger radius for this circular motion.

To adjust numerical panels:

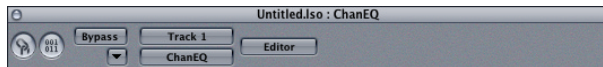
- Click-hold on the panel's numerical value and drag up/down. If there are up/down arrows alongside such panels, you can use them to increment/decrement the value by one step.

Note: You can reset any parameter to its default value by Option-clicking on it.

Note: If you hold Shift before clicking and moving a control, its value can be fine-tuned.

Common Plug-in Window Parameters

The gray area at the top of the Plug-in window is common to all Logic plug-ins. It offers a number of important functions for plug-in use.



Link

The button to the extreme left (with a chain on it) is called the *Link* button. If the Link button is switched on, a single Plug-in window will be used to display all opened plug-ins. Each time you launch a new plug-in, the window will update to reflect the new selection. By default, the Link button is switched off, allowing you to open several Plug-in windows simultaneously. This is handy if you want to compare the settings of two plug-ins or adjust several plug-ins at the same time.

When changing the Arrange track, an open Plug-in window will update to display the corresponding slot's plug-in on the newly-selected track. As an example, if the ES1 was loaded on Audio Instrument channel 1, and an EXS24 instance was loaded on Audio Instrument channel 1, switching between these tracks would automatically update the Plug-in window to show the ES1/EXS24, respectively.

Bypass

The *Bypass* button allows a plug-in to be deactivated, but not removed from the insert/instrument slot. You can also bypass the effect directly in the Audio Object by Option-clicking on the corresponding insert slot.

Settings Menu (Arrow)

Clicking the *Arrow* to the right of the *Bypass* button accesses the Settings menu. Further information on this can be found in “Plug-in Settings” on page 19.

Switching the Contents of the Plug-in Window

You can reassign any open Plug-in window—in two different ways—via the two pull-down menus to the right of the Settings menu (the *Arrow*):

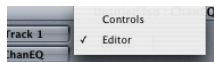
- Use the upper pull-down menu (*Track 1* in the diagram) to switch the Plug-in window between all channels that use the same plug-in. If you have inserted the EVB3 on tracks 1 and 6, for example, you can switch between these channels and adjust the parameters of each EVB3 instance, respectively.
- In the lower pull-down menu you can switch between the plug-in slots of the selected channel. As an example, if a particular channel uses an Equalizer and an EVB3 plug-in, you can switch the Plug-in window between these plug-ins.

Editor—Controls View

The plug-in parameters can be viewed in two forms: Controls and Editor. The Editor view shows the plug-in's graphical interface, if it offers one. The Controls view displays all plug-in functions as a set of horizontal sliders, with numerical fields to the left of each parameter. These fields are used for both the display and entry of data values.

To switch the view modes:

- 1 Click-hold the *Editor* button in the gray area at the top of the Plug-in window.



- 2 Choose *Controls* from the pull-down menu.

Some Logic plug-ins may have additional parameters that don't show up on the Editor control panel. This is indicated by an additional *001/011* button next to the *Link* button.



Activate this button to reveal sliders for the extra parameters at the bottom of the Plug-in window.

Plug-ins With Side Chain Input

All plug-ins that support side chain inputs, feature an additional *Side Chain* pull-down menu in the gray area at the top of the Plug-in window. This facilitates the routing of any Audio track, Input channel or Bus Object into the plug-in via a side chain.

You can also route an Instrument channel as side chain signal, if you follow these steps:

- 1 Create a Send, using a Bus on the Instrument channel.
- 2 Choose the selected Bus as a Side Chain input for the plug-in.

Once the Side Chain input is selected, the plug-in processes the audio of the channel it is inserted in, using the trigger impulses provided by the Side Chain. The signal peaks of the Side Chain input, combined with the Threshold parameter of the plug-in, determine when the plug-in is triggered.

Examples for Side Chaining

- A sustained pad sound is sent through a noise gate, which is triggered by a drum track being used as the Side Chain input signal. This results in a rhythmic pad sound which follows the signal peaks of the drum track.
- A noise gate inserted into a bass guitar channel is triggered by the kick drum track via the Side Chain. This can “tighten” the timing of the bass guitar, as it follows the kick drum signal.
- Side Chains can also be used to blend a music mix with a voice-over. To achieve this, the mix needs to be routed through a compressor which, in turn, is side chained, using the voice-over track. In this type of setup, the music becomes softer when the narrator is speaking, and louder, when not. The effect is also known as ducking. Please note that in order for this to function, the automatic gain make-up or Auto Gain (if applicable to the compressor plug-in) must be disabled.

Plug-in Settings

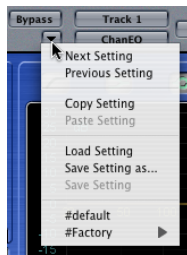
Logic's plug-ins ship with a library of ready-to-play preset sounds, known as Settings. These Settings can be found in the Logic > Plug-In Settings subfolder, following the installation procedure.

Note: It is strongly recommended that you do not attempt to change the Logic > Plug-in Settings folder structure. Within the Plug-in Settings folder you are, however, free to sort your settings into sub folders. This folder structure is reflected in a hierarchical menu, shown each time you load a plug-in setting.

All current plug-in settings are stored with the song file, and are automatically recalled the next time you load the song. You can also recall and save individual settings via the Settings menu functions. The Settings pull-down menu can be opened by clicking on the *Arrow* in the gray area at the top of the Plug-in window.

Functions of the Settings Menu

In the gray area at the top of each Plug-in window is an Arrow button. Clicking on it opens the Settings menu, which features the following functions:



Copy Setting

Choose this entry to copy all parameter settings into a special Settings clipboard, which is independent from the global Logic clipboard.

Paste Setting

If you have opened a plug-in of the same type (two SilverVerb instances, for example), you can use this command to paste the parameter set from one to the other via the Settings clipboard.

Save Setting

This allows you to name and save a setting.

Note: If you save a Setting with the name of *#default* in a plug-in's Settings folder, it will be loaded as the default plug-in Setting.

Load Setting

This function can be used to load a setting. The file selector box only shows settings for compatible plug-in types. Each plug-in has its own set of parameters, and therefore its own file format.

Note: Proprietary plug-in-settings created in Logic for Windows can be read by Logic for Mac OS, and vice versa. Plug-in settings files created on the Mac must be saved with a .pst file extension in order for them to work in Logic for Windows.

Note: Some plug-ins allow you to load Settings files by dragging and dropping them from the Finder. This poses a problem as float windows will disappear once Logic is “in the background,” and the Finder becomes the active application. To circumvent this issue, you can hold Option when inserting a plug-in, making it a non-floating window.

Next/Previous Setting

These functions allow you to load the next/previous setting in the folder. You can also make use of the *Next/Previous Plug-In Setting* (or the *Next/Previous Plug-In Setting or EXS Instrument*) key commands. These are not set by default, so you will need to assign them. Once assigned, you can simply press the appropriate key command to step forwards/backwards through your plug-in settings. In Logic Pro, *Previous/Next Setting* can be assigned to almost any MIDI message, such as Control Change or Program Change commands.

Settings of other Manufacturers

Logic can read the most common settings files used by Audio Unit plug-ins.

Loading and Saving Multiple Plug-ins

Logic’s Mixer windows allow you to save and load multiple plug-ins (inclusive of their Settings files) via the arrow pull-down menu alongside the word *Inserts* on channel strips. The entire channel strip can be stored and recalled for use on any suitable Audio Object, allowing common chains of effects such as Reverb, Chorus, and Delay to be loaded far more quickly than individually inserting each plug-in. Further details can be found in the Logic reference.

Plug-in Automation

Almost all Logic plug-ins can be fully automated, which means that you can record, edit, and play back almost any movement of any knob, switch or fader in any plug-in. For more information, please read the Automation chapter in the Logic reference.

Plug-ins From Other Manufacturers

Audio Unit Support

Correctly installed third-party Audio Unit plug-ins (Effects and Instruments) can be used in Logic. Clicking on an Audio object insert/instrument slot will launch the hierarchical Plug-In menu. A separate Audio Units submenu displays all installed Audio Unit plug-ins.

TDM Plug-ins

Users of a Digidesign TDM system can utilize TDM plug-ins in Logic.

This chapter explains the difference between effect and instrument plug-ins.

Instrument plug-ins respond to MIDI note messages, effect plug-ins do not. Therefore instrument plug-ins can only be inserted into special Audio Objects, called Audio Instruments.

Effect Plug-ins

Logic's effects can be installed into all insert slots of all Audio Object types (See "Inserting and Deleting Plug-ins" on page 13.). This allows processing of all audio and instrument signals.

There are two ways of sending audio to effects: via an insert, or via a bus (also known as an "aux send").

Insert Effects

With insert effects, all of the signal is processed. This means that 100% of the signal flows through the effect. This is suitable for equalizers or dynamic effects. This also typically applies to pan knobs and faders.

If you have enough processing capacity, you can use up to 15 insert effects per audio object. An extra blank insert is created, as soon as all the currently displayed inserts are used, up to the maximum allowed.

Bus Effects

When you use bus effects, a controlled amount of the signal is sent to the effect. Buses are typically used for effects that you want to apply to several signals at the same time.

Within Logic, the effect is placed in an insert slot of a bus object. The signals of the individual tracks can each be sent to the bus, controlled by a *Send* knob.



The audio signal is then processed with the effect, and mixed with the stereo output.

The advantage of this “bussed” approach, over inserting effects on tracks, is efficiency. This method allows as many tracks as you like to be processed by one inserted plug-in, massively saving CPU power when compared to insertion of the same effect directly into multiple tracks.

For computationally-intensive effects such as reverb, it’s always advisable to insert them into a bus. Chorus, Flanger, and Delay effects should also always be inserted into a bus, if they are going to be used on more than one track.

In some cases, it may make sense to patch an effect such as a delay, directly into the insert of an individual track. There are no restrictions in Logic as to where effects may be used.

Instrument Plug-ins

The Audio Instrument Object Type

Unlike effect plug-ins, instrument plug-ins respond to MIDI note messages. Instrument plug-ins can only be inserted into special Audio Objects, called Audio Instruments. Audio Instruments feature a special instrument slot, directly above their Output slot.

An Audio Instrument is an Audio Object with its *Channel* parameter switched to one of the *Instruments*. Any audio object can be switched to operate as an Audio Instrument, by changing this parameter (*Channel*) in the Object Parameter box.

To create an Audio Instrument Object:

- 1 Open Logic’s Audio Mixer, by choosing Audio > Audio Mixer.
- 2 In the Audio mixer window select New > Audio Object to create a new Audio Object.

- 3 Double click the newly-created Audio Object icon, so that the (grayed out) channel strip appears.



- 4 Now, go to the Object Parameter box, and set the *Channel* parameter to an Instrument. The generic Audio Object will now operate as an Audio Instrument, allowing you to insert any Instrument plug-in into the instrument slot.

The default song—the song that opens if you move the Autoload Song away from the Logic folder—features a number of ready-configured Instruments, that can be accessed via the Track Mixer or Audio Mixer.

The output signal of a software instrument plug-in is fed into the input (the instrument slot) of the Instrument channel strip, where it can be processed via inserted plug-ins and/or sent to busses.

Logic supports up to 64 discrete Audio Instruments. The number of instrument instances which can be run simultaneously is dependent on the availability of computer processing resources.

Following the insertion of an instrument, the Audio Instrument Object can be used just like a MIDI track in the Arrange window. The Audio Instrument Object can also receive MIDI notes from standard MIDI instrument objects via Environment cables. This is useful for creating layered sounds with “real” MIDI instruments and virtual instruments. Please note that the *Options > Preferences > MIDI > Use Unified Virtual and Classic MIDI Engine* setting needs to be switched on for these features to work.

When an Audio Instrument track is selected, it is ready to be played in real-time and consequently produces some system load. Normally, Logic releases system resources used by the Audio Engine when the sequencer is stopped. This is not the case, however, if an Audio Instrument track is selected in the Arrange window, and is therefore available for real-time playing. Selecting a MIDI track or a standard Audio track exits this Audio Instrument “stand by” mode, and releases reserved system resources when the sequencer is stopped.

Note: Muting an Audio Instrument track in the Arrange does *not* reduce system load.

Logic's *Bounce* function allows the entire Audio Instrument track to be recorded as an audio file. This "*Bounced*" audio file can then be assigned (as an audio region) to a standard Audio track, allowing you to reassign the available processing (CPU) power for further synthesizer tracks. For details, please refer to the *Bounce* chapter in the Logic Reference manual.

You can also make use of the Freeze function to capture the output of an Audio Instrument track, again saving processing power. For details please refer to the Freeze section, in the Logic Reference manual.

Accessing Multiple Outputs

Logic supports the multiple outputs of the EXS24 and all Audio Unit (AU) compatible instruments. In addition to the *Mono* and *Stereo* submenus of the Audio Instrument plug-in menu, a *Multi Channel* submenu lists all Instruments that offer multiple outputs. A plug-in needs to be inserted from the Multi Channel submenu, in order to access its individual outputs.

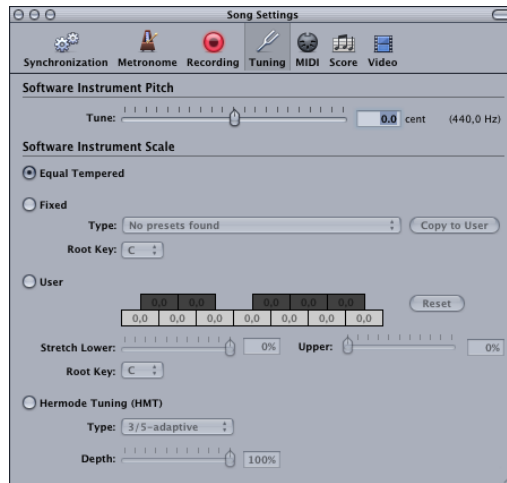


Note: Not all plug-ins (both Logic and third-party) are multi-output capable. If the Instrument does not appear in the Multi Channel submenu, it is not equipped with multiple output facilities.

The first two outputs of a multiple output instrument are always played back as a stereo pair by the Instrument channel in which the plug-in is inserted. Additional outputs (3 and 4, 5, and 6, and so on) are accessed via the Aux Objects.

Software Instrument Pitch

The *Song Settings > Tuning > Software Instrument Pitch > Tune* parameter remotely controls the main tuning parameter for all software instruments (plug-in synthesizers, such as the ES1 or EXS24 sampler and others) by ± 100 cents.



Note: Some instruments do not recognize this remote command.

No Hermode Tuning

Logic allows all software instruments to be globally tuned to different tempered scales, including Hermode Tuning (see the Tuning section of the Logic reference manual for details). There may, however, be occasions where you want individual Software Instruments to be exempt from this global tuning system.

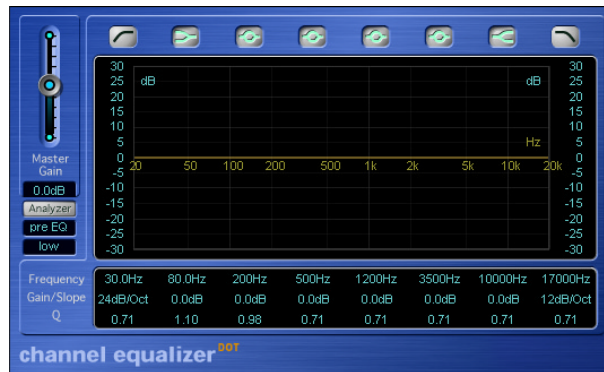
When *File > Song Settings > Tuning > Hermode Tuning* is active, a *No HMT* checkbox is visible in the Object Parameter boxes of all Audio Instrument channels. Simply click in this box to prevent the selected software instrument from following the global Hermode Tuning scale.

Any software instrument with an active *No HMT* checkbox will be played back at equal temperament.

This chapter covers all Logic equalization effects. Equalizers allow you to increase or decrease the level of selected components in the overall audio spectrum.

Logic's built-in equalizers include the Channel EQ, Linear Phase EQ, Match EQ, Fat EQ, Silver EQ, DJ EQ, High/Low Pass Filters, High/Low Cut EQ, Parametric EQ and High/Low Shelving EQ plug-ins.

Channel EQ



The extremely high-quality Channel EQ offers eight frequency bands and an integrated FFT analyzer.

EQ Parameters

The *Band Type* buttons above the display activate the Channel EQ's bands individually; inactive bands do not use any computer resources.

Band 1 is a lowpass filter and band 8 is a highpass filter.

Note: The Q-parameter of band 1 and band 8 will have no effect when using a slope of 6 dB/Oct.

Bands 2 and 7 are defined as shelving equalizers.

Note: When the Q parameter of band 2 and 7 is set to an extremely high value (to 100, for example), the equalizers only apply to a very narrow band, and can work in a fashion that is similar to notch filters.

Bands 3 to 6 are bandpass filters.

You can set the band parameters either in the parameter area or directly in the central EQ display. Move the mouse horizontally over the display. When your mouse cursor is in the access area of a band, its individual curve and parameter area will be highlighted and a pivot point appears. When you click-hold the mouse button directly on the (illuminated) pivot point of a band, vertical movements (up/down) will change its Q value. Horizontal movements (left/right) change the *Frequency* of the band. When you click-hold the mouse button on the display background, horizontal movements will again change the *Frequency* of the band. Vertical mouse movements will change the *Gain* of band 2 to 7. The slope values of the highpass and lowpass filters (bands 1 and 8) can only be changed in the parameter area below the graphic display. Click-hold on the parameter: Moving up increases, and down decreases, the value.

After boosting or cutting frequency bands, you can use the *Master Gain* fader to readjust the output level of the Channel EQ.

Channel EQ—Analyzer

The precision analyzer of the Channel EQ uses Fast Fourier Transformation (FFT) to show the energy of all frequency components of the signal. The central display of the Channel EQ fulfills multiple display functions: it shows both the curve of the FFT analyzer and the EQ curve. An identically scaled frequency axis is shown for both. This allows you to easily recognize unwanted frequencies in the analyzer curve, while using the EQ to edit them accordingly.

A click on the *Analyzer* button activates/deactivates the FFT analyzer. The display directly under the button determines the location of the Analyzer. You can switch the Analyzer *pre EQ* or *post EQ* (default) in order to compare the original signal with your edits.

Click into the display to open a pull-down menu that defines the resolution of the FFT analyzer—or more accurately, the number of frequency bands. The higher the precision of measurements, the more CPU power is needed.

High resolutions are necessary whenever you need reliable results in the very low bass frequency area. The bands derived from FFT analysis are divided in accordance with the frequency linear principle—non-technically, this means that there are far more bands in the highest octave than in the lowest.

Use the *Scales* to the left and right of the EQ display, to change the vertical scale of the EQ and analyzer curves.

To increase the resolution of the EQ *Gain* parameter (*dB Warp*) in the most interesting area around the zero line, click-hold in the green dB Scale on the left side of the graphic display, and move the mouse up. Moving the mouse down, will decrease the parameter value. The overall range is always ± 30 , but small values will be easier to recognize.

As soon as the Analyzer is activated, you can change the *Analyzer Top* parameter, which alters the scaling of the FFT analyzer, on the right side of the graphic display. The visible area represents a dynamic range of 60 dB, but by click-holding and vertically dragging, you can adjust the maximum value between +20 dB and -40 dB. The Analyzer display is always dB-linear.

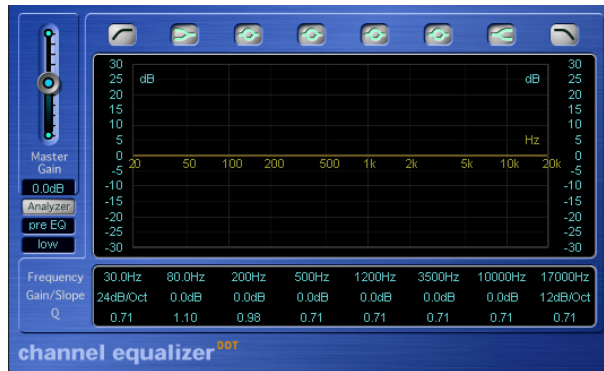
Two additional Analyzer parameters are available via the 001/100 view. *Analyzer Mode* allows you to switch between Peak and RMS. *Analyzer Decay* allows you to adjust the decay rate (in dB per second) of the Analyzer curve (peak decay in Peak mode or an averaged decay in RMS mode)

Note: The FFT analyzer needs additional CPU resources. In fact, resource consumption increases significantly at higher resolutions! We recommend that you disable the Analyzer or close the Channel EQ window after setting the desired EQ parameters. This will free up CPU resources for other tasks.

Using the Channel EQ as the Default EQ

The Channel EQ replaces the Track EQ of older Logic versions. It is inserted into the first available insert slot by double-clicking the EQ area on the upper portion of mixer channel strips. This area will change to a thumbnail view of the Channel EQ display. The thumbnails provide an overview of the EQ settings used in each individual channel.

Linear Phase EQ



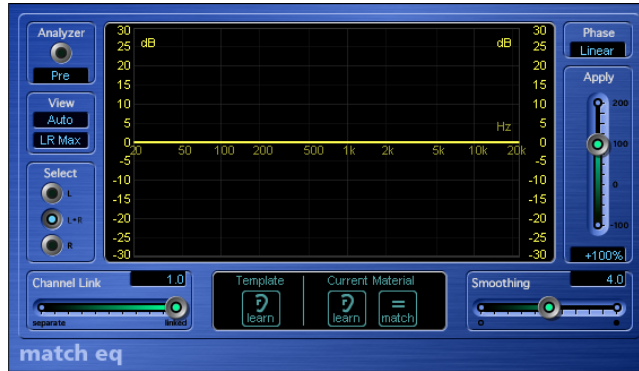
The extremely high-quality Linear Phase EQ plug-in is almost identical to the Channel EQ. With the exception of the different name and a few different colors, it uses the same familiar eight-band layout, and method of operation, as the Channel EQ.

Under-the-hood, however, the Linear Phase EQ uses completely different technology which preserves the phase of the audio signal 100%—even if the wildest EQ curves are applied to the sharpest signal transients!

As with all good things in life, there is a catch. The Linear Phase EQ uses more CPU power than the Channel EQ. Another factor is the inherent amount of latency introduced by this technology. Logic's plug-in delay compensation will successfully prevent the worst of these latency artefacts in mixdown situations—but don't even think about playing software instruments live when using the Linear Phase EQ.

As the parameters of the Channel EQ and Linear Phase EQ are almost identical, you may freely copy settings between them. For more information on the parameters of the Linear Phase EQ, read up on the "Channel EQ" on page 29.

Match EQ



The Match EQ plug-in allows you to “match,” and transfer the frequency spectrum from one signal to another, or to store it as a spectral template file. In this way, you can acoustically match the sound of various songs for an album, or impart the “sound” of any reference source onto your own recordings. The alignment of signals is automatic, but you can also manually draw or modify the filter curve to alter the sound as required.

Note: Match EQ acoustically matches two audio signals. It does not, however, match any dynamic differences in the two signals.

Description of the basic functions

Match EQ is a learning equalizer that reads the frequency spectrum of any reference source, including: the input signal, an audio file, or a template. Alternatively, you can load a setting file or import the settings of another Match EQ instance via a copy and paste operation.

You can analyze the average audio spectrum of the track the plug-in is assigned to or load another setting file or template. By matching both spectra, a filter curve is generated. This generated curve adapts the track signal to match the sound of the template. If required, you can modify the filter curve by boosting or cutting gain in different frequencies, or inverting it. Further to this, you can manually modify the curve by creating a virtually infinite number of peak filters, and adjust them as required. In this way, you can draw your own filter curve to optimize the sound as required. The internal analyzer allows you to visually check the frequency spectrum of the original data and the resulting curve, making manual corrections at specific points within the spectrum easier.

Parameters

The *View* pull-down menu allows you to select the type of information shown on the analyzer display in the center. The following options are available:

- **Automatic:** Depending on the selected function, the analyzer view is automatically toggled between the three following options.
- **Template:** The analyzer display shows the average frequency curve, which is generated by analyzing the input signal or loading a template.
- **Current Material:** The analyzer visualizes the average frequency curve, which is generated by analyzing the track signal or loading a Setting file or template.
- **Filter:** The analyzer displays the filter curve, which is generated by matching the *Template* and the spectra of the *Current Material*.

Independent of the selection, the analyzer can be activated/deactivated via the *On/Off* button. The *Analyzer Position* pull-down menu allows you to place the analyzer tap before (*Pre*: unchanged) or behind the filter circuit (*Post*: behind the Match EQ).

Note: Deactivating the analyzer frees up processing power for other applications.

On stereo channels, the view mode is configured via the lower *View* toggle menu. You can select whether the analyzer displays both audio channels via separate curves (L&R) or the summed maximum level (LR Max).

You can manually modify the filter curve generated via matching the Template with the Current Material. The buttons in the *Select* section let you choose whether the modifications are applied only to the left, right, or both channels.

You can refine this selection via the *Channel Link* slider. If the slider is set to 1.0, the *L* and *R* buttons for the single channels will have no effect, because both channels are represented via a common EQ curve. At the minimum value of 0.0, two separate filter curves are displayed, each of which can be selected for editing via the *L* and *R* buttons. The intermediate settings of the Channel Link slider allow you to blend these extreme values as required. As a result, any modification to either of the filter curves is transferred to both channels, dependent on the Channel Link setting.

Note: In the mono version of the plug-in, the parameters in the View Mode, Select, and Channel Link sections have no effect.

The *Template* and *Current Material* buttons perform the spectral analysis of the audio signals, and match the resulting curves. Clicking the *Learn* button in the Template section starts and stops measurement of the average frequency spectrum in the reference signal.

Clicking the *Learn* button in the Current Material section starts and stops measurement of the average frequency spectrum in the audio material of the track.

Note: Audio files can also be dragged onto the Template or Current Material Learn buttons to generate template or current spectra. A progress bar displays the progress of the analysis process.

If you right-click (or Control click) on either of the *Learn* buttons, a context menu opens. This menu allows the spectrum of the template or the track signal (Current Material) to be:

- cleared
- copied to the Match EQ clipboard, which is common to all Match EQ instances in the current song.
- pasted from the Match EQ clipboard to the active instance.
- loaded from a stored Setting file.
- generated from an audio file (chosen in the File Selector). This is done by choosing the *Generate Template/Current Material Spectrum from audio file* option, and selecting an appropriate file in the file selector that appears. A progress bar displays the status of the analysis process.

Note: When you activate the *Learn* button in either the Template or Current Material section, the *View* parameter is set to Auto, and the analyzer will display the current status of the spectral analysis, indicated by a progress bar.

The *Match* button in the Current Material section allows you to write the differences between the learned or loaded Template and the learned or loaded spectrum of the Current Material to a filter curve. Differences in gain are automatically compensated for, with the resulting EQ curve referenced to the 0 dB line.

The filter curve is updated automatically each time a new template or current material spectrum is learned or loaded, when the Match button is enabled. You can toggle between the matched (and possibly scaled and/or manually modified) filter curve and a flat response by activating/deactivating the Match button.

Note: Each time a new audio spectrum is matched—either by loading/learning a new spectrum while *Match* is activated or by activating *Match* after a new spectrum has been loaded—existing manual modifications are discarded, and *Apply* is set to 100%.

Basically, only one of the *Learn* buttons may be active at a time. As an example, if the Learn button in the Template section is active and you press the Learn button in the Current Material section, the analysis of the template file stops, and the current status is used as the spectral template. Analysis of the track (Current Material) will then begin.

Note: If you have manually modified the filter curve, the original (or flat) curve can be restored by Option-clicking on the background of the analyzer display. A second Option-click restores the most recently modified curve.

The filter curve can be edited via the *Smoothing* slider. At a value of 0.0, the filter curve is applied to the track signal without any changes. At all other *Smoothing* settings, the filters are smoothed at a constant bandwidth. A value of 1.0, for example, means that all filters have a constant bandwidth of one semitone that is used to smooth the notch-like filters in the curve. A bandwidth of: four semitones (a value of 4.0—or a major third), an octave (a value of 12.0) and two octaves with the maximum setting (24.0).

Note: *Smoothing* does not affect any manual modifications of the filter curve.

The *Apply* slider exaggerates (101% to 200%), reduces (99% to 1%) or inverts the peaks/dips (–1% to –100%) the effect of the filter curve on the track signal. At a value of 100%, the signal is aligned to the curve without any changes.

The *Phase* toggle menu switches the operational principle of the filter curve.

- The *Linear* option prevents processing from altering the signal phase. At the same time, the latency of the plug-in will increase.
- The *Minimal* option alters the signal phase, but latency is reduced.

Manual Modifications

You can graphically edit the matched filter curve directly in the display. Just click at any point within the filter curve to create a new peak band. You can shift the peak frequency for this band (within the entire spectrum) by dragging the mouse horizontally. Vertically moving the mouse allows you to set the gain of this frequency band (range: –24 to +24 dB). The Q-factor of the filter is set by the vertical distance between the click point and the curve. By clicking on the curve, the maximum Q-value of 10 (for notch-like filters) is used. Clicking above or below the curve decreases the Q-value. The further you click from the curve, the smaller the value (down to the minimum of 0.3).

- The Q-factor can be changed continuously by pressing Shift and moving the mouse up/down while keeping the mouse button pressed.
- If Option is hold while releasing the mouse button, the modification is cancelled.

Note: The current values are shown in a window within the display while the left mouse button is held down.

The colors and modes of the dB scales on the left and right of the display are automatically adapted to the active function. If the analyzer is active, the left scale displays the average spectrum in the signal, while the right scale serves as a reference for the peak values of the analyzer. Basically, the analyzer visualizes a dynamic range of 60 dB. The displayed range can, however, be shifted between the extreme values of +20 dB and –100 dB by click-dragging on the scale.

If the resulting filter curve is displayed, the left scale—and the right, if the analyzer is inactive—shows the dB values for the filter curve in an appropriate color. By click-dragging on one of the scales, the overall gain of the filter curve is adjusted in the range from -30 to $+30$ dB.

Fat EQ



The high-quality Fat EQ offers up to 5 fully parametric bands—buttons 1 through 5 activate these individually; inactive bands do not drain your computer's resources.

The icons above the graphic display let you determine whether Band 1 acts like a highpass filter or a low shelving EQ. Similarly, Band 5 can be switched back and forth between use as a lowpass filter and a high shelving EQ. Bands 2 and 4 can be switched from their normal operating mode (as fully parametric bandpass filters) to low or high shelving EQs. The center band (number 3) always operates as a fully parametric bandpass filter. The shelving filter's slope characteristics for bands 2 and 4 are adjustable via the Q parameter.

The area directly below the graphic display (depicting the frequency response curve) is used to select the frequency for the individual bands. Simply click on the number, and change the value with your mouse. You'll be able to hear an individual frequency better if you turn it up by rotating the *Cut/Boost* knob located below it clockwise.

The same holds true for any frequency that you want to attenuate. Once you've located the frequency that you're hunting for, you can back off the *Cut/Boost* knob level, and set it to the desired value. Use the Q (bandwidth) parameter located in the lower display to determine the extent that the band influences neighboring frequencies.

At low Q values, the EQ influences a wider frequency range, and at high Q values, the effect of the EQ band is limited to a very narrow frequency range. Please bear in mind that your perception of an attenuated or boosted frequency depends heavily on the Q parameter: If you're working with a narrow frequency band, you'll generally need to cut or boost it more drastically to notice a difference.

Silver EQ

The Silver EQ contains one High Shelf, a Parametric and one Low Shelf filter with the corresponding parameters. More on each of these is found in the Individual EQ's section below.

DJ EQ

The DJ EQ combines Low and High Shelving Filters with a fixed frequency, and one Parametric EQ with its attendant parameters. More on each of these is found in the Individual EQ's section below.

The special feature of the DJ EQ is that it allows the gain of the filters to be reduced down to -30 dB.

Individual EQs

Parametric EQ

The Parametric EQ offers the following three parameters:

- Hz : Center frequency
- dB : Cut/Boost
- Q : Quality

A symmetrical frequency range on either side of the center frequency is boosted or cut. You can adjust the width of this frequency range with the Q control.

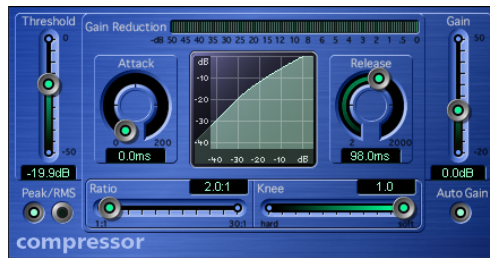
High Shelving EQ/Low Shelving EQ

- The Low Shelving Equalizer only affects the frequency range below the selected frequency.
- The High Shelving Equalizer only affects the frequency range above the selected frequency.

This chapter introduces Logic's Dynamic plug-ins.

This includes the Compressor, Silver Compressor, Expander, Noise Gate, Silver Gate, Enveloper, DeEsser, Limiter, Multipressor, and Adaptive Limiter plug-ins.

Compressor



A compressor tightens up the dynamics of a signal. This means that the difference in levels between loud and soft passages is reduced. This “evening out” of the loud and soft passages means that the peak level remains pretty constant, and the overall loudness—the perceived volume—of a track is increased. Next to an EQ, a compressor is your most valuable sound-shaping tool when mixing. A compressor is a universal effect, it has a virtually unlimited range of applications. You should definitely exploit it for vocal tracks, but a compressor can also often work wonders for entire mixes. When you use a compressor, be sure to route the entire signal through it, by inserting it directly into channels. It should only be used in a bus when you want to compress a group of tracks (a drum kit, for example) simultaneously, and by the same amount. Again, these tracks (individual drums in a kit, for example) should be routed to the bus in their entirety, as opposed to using Send knobs to route just parts of each signal to the bus. You do this by selecting the appropriate bus as the output destination for the tracks that you wish to compress.

Logic's Compressor was designed to emulate the response of the finest analog compressors. It follows the following principle: When a signal exceeds the defined *Threshold* level, the compressor actually alters the response, so that it is no longer linear. What happens is that all levels that exceed the Threshold are attenuated by the value set with the *Ratio* slider. A ratio of 4:1 means that an incoming level that is 4 dB louder than the Threshold level is dampened, so that it comes out the other end of the compressor with a level that is just 1 dB above the Threshold level. On the flip side, if you route in a signal that is loud enough to double the output level of the compressor (+6 dB), the input signal would need to have a level 24 dB greater than the Threshold level. This tells us that a compression ratio of 4:1 is a fairly drastic manipulation of the original signal's dynamics. Given that the compressor lowers levels, the volume of its output signal is normally lower than that of the input signal.

To compensate for this decrease in levels, the output of the compressor is equipped with a *Gain* slider. *Auto Gain* automatically sets the level of amplification to a value equivalent to the "sum of the threshold value minus the threshold value divided by the ratio" or put less confusingly $T - (T/R)$. This function ensures that a normalized input signal is amplified so that the output signal is also normalized, regardless of the values set for Threshold and Ratio—provided you are dealing with relatively static signals. Use the *Attack* and *Release* knobs to shape the dynamic response of the compressor. *Attack* determines the amount of time it takes for the compressor to react to signals that exceed the Threshold. At higher values, the compressor does not fully dampen a signal until it runs through its Attack phase. This type of setting ensures the original attack, for example the sound of a pick or finger striking a guitar string, remains intact or clearly audible. If, on the other hand, you want to maximize the level of a master signal, set the *Attack* knob to low values, ensuring that the compressor responds more swiftly. *Release* determines the amount of time it takes for the compressor to stop dampening louder passages, once the signal level falls below the Threshold level. If the compressor generates an ugly pumping sound, adjust the *Release* knob accordingly.

When you have configured a compressor so that it dampens the signal at or above the Threshold value by the predetermined Ratio, while the level just below the Threshold is routed through at a 1:1 Ratio, an audio engineer would term the compression as hard knee. In many cases, however, you'll come up with a better sounding track by using a more gradual transition from the 1:1 Ratio below the Threshold, to the Ratio that you entered for levels above the Threshold. In this scenario, the characteristic curve is not as radical—it rises gradually from the bottom left to the top right, as seen in the graphic display. This type of compression is called soft knee. The *Knee* slider lets you incrementally select anything from hard to soft knee. This wide range of options provides you with the tools you need to shape the sound as you like; whether you want to radically maximize loudness with absolutely no regard for the original dynamics (hard), or are going for the more musical compression that acoustic recordings typically require (soft). Keep in mind that *Knee* only controls the shape of the compression, not its intensity; use the *Threshold* and *Ratio* sliders for this purpose.

Incidentally, the *Gain Reduction Meter* indicates the intensity of compression used to tighten up the original signal. This feature is a great help, particularly if you're not experienced with using compression. Keep an eye on it to make sure that you're not overly compressing your tracks.

When the compressor has to decide whether or not the level exceeds the Threshold (or if the level is getting close to the Threshold, for soft knee compression), it can analyze either the peak or RMS level. The latter value is a better indication of how humans perceive loudness. When you use the compressor primarily as a limiter, select the *Peak* button. When you're compressing individual signals, use of the *RMS* button will often deliver better, more musical results.

If you activate *Auto Gain* and *RMS* simultaneously, the signal may be saturated. If you hear any distortion, switch *Auto Gain* off, and enter a suitable gain level manually.

The *Output Clip* parameter limits (clips) the output to 0 dB, via the *OFF/SOFT/HARD* settings. This setting is only available in the Controls view.

Note: Despite all of these handy tips for tweaking sounds, you should always keep one thing in mind—there are no hard and fast rules. Use your own taste and ears. If it sounds good, it is good.

Silver Compressor



The Silver Compressor is a simplified version of the Compressor. It is limited to *Threshold*, *Attack*, *Release*, and *Ratio* controls.

Expander



The Expander is similar to the Compressor, with one fundamental difference—it increases, rather than reduces, the dynamic range above the Threshold.

The *Ratio* slider features a value range of 1:1 to 0.5:1. This means that the Expander is a genuine upward expander (as opposed to a downward expander that increases the dynamic range below the Threshold). You can use this effect to emphasize the transients of highly compressed signals. This spices up the sonic image, making it sound livelier and fresher.

Please bear in mind the fact that you will perceive the signal as being softer, even when the peak level remains the same. In other words, the expander decreases loudness. If you manipulate the dynamics of a signal fairly radically (depending on the Threshold and Ratio values), you'll find that you'll need to reduce the level via the *Gain* slider to avoid distortion. In most cases, *Auto Gain* will take care of this for you.

Please check the Compressor section for details on the various parameters.

Noise Gate



Ordinarily, a noise gate suppresses unwanted noise that may become audible during a lull in the signal. You can, however, also use it as a creative sound-sculpting tool.

Here's the basic principle behind a noise gate: Signals that lie above the *Threshold* are allowed to pass unimpeded (open gate). Anything below the defined *Threshold* (background noise, crosstalk from other signal sources and so on) is fully muted (a closed gate). In other words, the *Threshold* slider determines the lowest level that a signal must be at, in order to open the gate—it separates the wanted or useful signal, from the unwanted or noise signal.

The *Reduction* slider allows you to control the intensity of noise suppression. As a rule, you should set it to the lowest possible value and leave it there, to ensure that the gate closes completely. If you prefer, you can select other values, thus reducing the noise signal less dramatically. As an alternative, you can actually boost the signal by up to 20 dB.

The three rotary knobs (at the top) influence the dynamic response of the noise gate. If you want the gate to open extremely quickly, say for percussive signals such as drums, set the *Attack* knob to the lowest value by turning it as far as it will go counter-clockwise. If the signal fades in a bit more softly, as is the case with string pads and the like, a noise gate that opens too quickly can wreak havoc with the signal, causing it to sound unnatural.

For this type of sonic scenario, set the *Attack* knob so that the gate emulates the attack of the original signal. Much the same holds true for the Release phase of signals. When you're working with signals that fade out gradually or have longer reverb tails, you should turn the *Release* knob up, allowing the signal to fade naturally.

The *Hold* knob determines the minimum amount of time that the gate stays open. This knob avoids the dreaded chattering effect caused by a rapidly opening and closing noise gate. The *Hysteresis* slider provides another option for avoiding chatter, without needing to define a minimum *Hold* time.

Let's back up a bit for a brief explanation: Noise gates often begin chattering when the level of a signal fluctuates slightly, but very rapidly, during the attack or release phase. Instead of clearly exceeding or falling short of the Threshold value, the signal level hovers around the Threshold. The Noise Gate then rapidly switches on and off to compensate, producing the undesirable chattering effect. If you were able to tell the Noise Gate to open at the determined Threshold level and remain open until the level drops below another, lower, predefined Threshold level, you'd be able to avoid chatter—as long as the sonic window formed by these two Threshold values is large enough to contain the fluctuating level of the incoming signal.

This is exactly what the *Hysteresis* feature enables you to do—the value determined by the *Hysteresis* slider is actually the difference between the *Threshold* values that open and close the gate. This value is always negative. Generally, -6 dB is a good place to start.

If you're dealing with audio material featuring extremely sensitive transients, or attack phases that are critical to the overall sound, you may find it beneficial to have the Noise Gate open up a tad before the useful signal fades in. This is what the *Lookahead* slider is designed for. The program analyzes the signal level ahead of time, and anticipates the point at which it can open the gate before the signal actually reaches the *Threshold* value. When you choose to use this feature, please make sure you set the *Attack*, *Hold* or *Hysteresis* controls to appropriate values.

When you're working with noise gates, you'll run across scenarios where the useful signal and the noise signal have levels that are near enough to be perceived as identical. A typical example is the crosstalk of a hi-hat—its signal tends to bleed into the snare drum track when you're recording a drum kit. If you're using a noise gate to isolate the snare, you'll find that the hi-hat will also open the gate in many cases. To avoid this effect, the Noise Gate offers Side Chain filters.

When you press and hold the *Monitor* button, you can audition the Side Chain signal. You can then set the filters to only allow frequencies that contain a particularly loud, useful signal to pass. For this example, we'll use the Noise Gate's *High Cut* filter—that only allows the bottom end and mids of the snare to pass, and cuts the higher frequencies of the hi-hat. When you switch Side Chain Monitoring off, it will be much easier to set a suitable Threshold level. This will be a value that is only exceeded by the level of the louder useful signal—the frequencies that make up the snare's fundamental tone, in our example. Put simply, the Noise Gate only allows the sound of the snare to pass. Should the need arise, you can follow much the same procedure to isolate a kick or snare drum within an entire mixdown.

Silver Gate



The Silver Gate is a cut-down version of the Noise Gate. It is limited to *Threshold*, *Lookahead*, *Attack*, *Hold*, and *Release* controls.

Enveloper



The Enveloper is an unusual tool that lets you shape transients—the attack and release phases of signals. No other type of dynamic effect (such as a compressor or expander) can achieve similar results—and these results can be quite impressive indeed.

The most important Enveloper controls are the two *Gain* sliders that govern *Attack* (left) and *Release* (right). In the center position, the signal remains unprocessed. If you turn the gain up, the attack or release phase is emphasized. If you turn it down, the corresponding phase is attenuated. As an example, boosting the attack lends a drum sound more snap, or amplifies the sound of a guitar string being plucked or picked. When you cut the attack, percussive signals fade in more softly. You can also mute the attack, making it virtually inaudible. This facilitates a range of interesting manipulations. Another handy application for this plug-in is maintaining friendships—it allows you to mask the poor timing of accompanying instruments, rather than tell your pals that they “groove like a bunch of accountants at the office Christmas party”.

Emphasizing the release also boosts the amount of any reverb on the affected track. Conversely, when you tone down the release phase, tracks originally drenched in reverb end up sounding drier. This effect is particularly useful when you're working with drumloops, but there are, of course, many other applications. Let your imagination be your guide.

When using the Enveloper, you should set the *Threshold* to the minimum value and leave it there. Only when you seriously crank the release phase, thus boosting the noise level of the original recording, should you turn the *Threshold* slider up a little. This limits the Enveloper to only influencing the useful signal.

Drastic boosting or cutting of the release or attack phase may change the overall level of the signal. The *Out Level* slider allows you to compensate for this effect.

The *Time* parameters for the attack and release phase (2 knobs below the graphic display) enable you to access the time-based intervals that the plug-in interprets as the attack and release phases. Generally, you'll find values of around 20 ms (attack) and 1500 ms (release) are fine to start with. Adjust them accordingly for the type of signal that you're processing.

Similar to its Noise Gate counterpart, the *Lookahead* slider allows you to define values that tell the Enveloper to anticipate what the signal will do in the very near future. Normally, you won't need to use this feature, except possibly for signals with extremely sensitive transients. If you do decide to use *Lookahead*, you may need to adjust the attack time accordingly.

To give you a better insight into the true nature of the Enveloper, here's a quick look at how it works: It is equipped with two internal envelope followers. One follows the amplitude of the input signal directly, whereas the other follows all changes generated by the variable delays (individually adjustable for attack and release). The difference between the two envelope followers is used to boost or cut the original signal by way of the corresponding *Gain* sliders (also individually adjustable for attack and release).

In contrast to a compressor or expander, the Enveloper operates independently of the absolute level of the input signal—provided the *Threshold* slider is set to the lowest possible value.

DeEsser



A DeEsser is a signal processor used for the rejection of hissing, or sibilant noises. This is why it is called a “DeEsser”, and occasionally an “S”-Suppressor. You can, of course, reject sizzling frequencies with an equalizer, but a DeEsser only rejects this high frequency band for as long as a threshold level is being exceeded in a specific frequency band. This “dynamic” ability is why the sound doesn’t get darker when no “sizzling” consonants are present in the signal. A DeEsser is a frequency-specific compressor, designed to only compress a particular frequency band within a complex full band signal. It features extremely fast attack and release times.

In the Logic DeEsser, the dynamic rejection does not necessarily need to take place in the same frequency range that’s being analyzed. Rather, the DeEsser performs a gain reduction in the frequency band displayed in the lower window for as long as the level exceeds a threshold (which falls within the frequency range) displayed in the upper window.

Note: Please don’t confuse a DeEsser with an effect known as a “Vocal Stressor”. The latter reduces the gain of the entire range when the level exceeds a threshold defined in a given frequency range. This type of processing can be achieved with any compressor with a high pass filter or EQ inserted in its side chain.

The Logic DeEsser does not make use of a frequency dividing network (a crossover, utilizing low and high pass filters). Rather, it is based on a subtraction of the isolated frequency band, leaving the phase-curve untouched.

The DeEsser is especially important in FM radio applications, because sharp S-type consonants can cause harsh intermodulation distortion noises. The need for this depends very much on the language spoken: English has fewer of these consonants than German or Spanish, for example.

Parameters of the DeEsser

Detector Frequency

This parameter defines the frequency band the DeEsser acts upon. It's not necessarily the same band that will be reduced.

Detector Sensitivity

This parameter defines the threshold level that needs to be exceeded (around the Detector Frequency), in order to reduce the level around the Suppressor Frequency.

Monitor

Activation of this switch allows you to monitor the Side Chain signal used by the DeEsser. If you want to reduce sizzling noises, listen to the input signal, and set the Detector Frequency in a way that makes the sizzling frequency range more prominent. (If you find that you like this filtered sound, combine a highpass and lowpass filter—in order to construct a bandpass—as this approach uses less processing power).

Suppressor Frequency

This parameter defines the frequency band that is reduced when the Detector Frequency Sensitivity threshold is exceeded.

Strength

Strength sets the amount of gain reduction around the Suppressor Frequency.

Smoothing

Smoothing controls the reaction speed of the gain reduction start and end phases. It's a combination of attack and release time parameters, as known from compressors.

Limiter



The Limiter is also a standard effect for processing a summed stereo signal. It is normally used for mastering.

You could say that a limiter is a compressor with an infinite compression ratio. The Limiter restricts dynamics to an absolute level. Any input level that exceeds the Limiter's threshold (*Gain*) will be output at this "limited" level, no matter how much higher the original signal level may have been. The fact that there is a level that the signal cannot exceed is the distinguishing characteristic of a limiter, when compared to a compressor.

Parameters of the Limiter

Gain

Most analog limiters would have a "Threshold" control (like that of the Multipressor), rather than a "Gain" control. This sets the level at which the Limiter will begin to work.

As the Limiter is digital, and is normally is applied as the very last mastering tool, we can presuppose that:

- the input signal sometimes reaches 0 dB, but does not exceed this value, and
- that the Limiter is being used to raise the signal's overall volume. This is the reason why you find a Gain control here—to set the desired level of gain for the signal.

The Limiter is designed in such a way that if set to 0 dB Gain and 0 dB Output Level, it doesn't work at all—on normalized regions. If there should be a region that clips (red gain line), the Limiter—using its basic settings—reduces the level before clipping can occur. (This does not apply to data that was clipped during recording).

Lookahead

Lookahead determines how far the processor looks into the future, in order to react earlier (thus better) to peak volumes. Unlike stand-alone processors, this function does not apply a general signal delay, as the Limiter is not limited to seeing the signal in real-time.

Set *Lookahead* to higher values, if you want the limiting effect to take place before the maximum level is reached.

Release

Here, you can set the time required by the Limiter (after limiting) to release the effect.

Output Level

This simple volume control sets the desired maximum level of the Limiter's output signal.

Softknee

Activate the *Softknee* button to produce a softer transition from no limiting to full limiting.

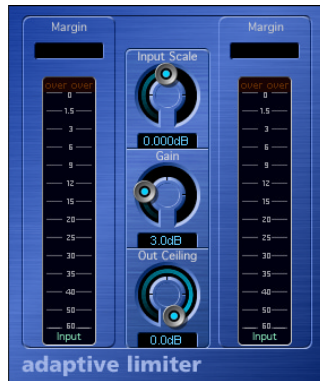
If switched off, the signal will be limited (following a linear curve) absolutely and exactly when a level of 0 dB is reached.

If switched on, the transition to full limiting is non-linear, meaning softer. The limiting of the signal will start before a level of 0 dB is reached. This will avoid distortion artefacts occurring when strong limiting is used without softknee.

Graphic Display

The graphic display shows the reduction of the level (starting from 0 dB downwards).

Adaptive Limiter



With its Compressor (see “Compressor” on page 39), Multipressor (see “Multipressor” on page 52) and Limiter (see “Limiter” on page 49) Logic features several extremely versatile options for increasing perceived volumes. A further tool which can be used to increase the perceived level of signals is the Adaptive Limiter plug-in. In the world of analog processors, it could be more closely compared to a clipper, rounding and smoothing only harsh level peaks, rather than to a VCA-type limiter. It allows you to achieve maximum gain, without having to fear exceeding 0 dBFS. The Adaptive Limiter may slightly color the sound. An effect most similar to an amplifier when driven hard.

Following the Adaptive Limiter process, tracks normalized to 0 dB appear to sound about 2 dB louder, depending on the source signal. As with other Logic dynamic processes, this plug-in also features a lookahead facility (the *Lookahead* parameter can be set in Controls view, allowing the Adaptive Limiter to look into the future. The Adaptive Limiter reacts to level peaks in signals streaming from the hard disk before they are played back, and delaying the monitored signal. The typical use of the Adaptive Limiter is in the summed mix. It is placed after the Multipressor and before Gain, in order to produce a CD of maximum loudness. As the Adaptive Limiter compresses signals, it can produce results which sound louder than those resulting from normalizing in the Sample Editor.

Start the process by adjusting the *Input Scale*, just as you would set a mixing desk's Gain parameter, or a digital recorder's recording level. The parameter behaves much like a Gain control, but its purpose is to adjust the input level, which must never exceed 0 dBFS. Adjust the *Gain* parameter to musically control the internal process of peak smoothing and gain increase.

Out Ceiling reduces the output level of the process in very fine steps within a range of only 2 dB. This is no threshold control, just a simple output gain.

The *Mode* menu in the Adaptive Limiter's Controls view allows you to choose between two different forms of peak smoothing. If you choose:

- *OptFit*, the signal will be limited by following a linear curve. This form of peak smoothing allows signal peaks that exceed 0 dB.
- *NoOver* ensures that the signal never surpasses 0 dB, avoiding distortion artefacts.

Remove DC (Controls view) activates a highpass filter that removes direct current from the signal. When using poorly constructed audio hardware, direct current (DC) can be undesirably layered over the audio signal.

The *Margin* display shows the maximum measured level. To reset, click the *Over* lamps.

Multipressor



The Multipressor (an abbreviation for multiband compressor) is the epitome of an audio mastering tool. It's a pretty complex tool; good sounding settings require quite a lot of listening experience.

Functional Principle of Multi-band Compressors

The multi-band compressor splits the incoming signal into two to four different frequency bands before applying compression. These frequency bands are then compressed independently. After compression, the frequency bands are mixed back together.

The aim of independent compression of different frequency bands is to reach high compression levels on the bands that need it, without the pumping effect that is normally heard at high compression levels.

Much higher Ratios, and therefore, a much higher average volume is possible before the unwanted artefacts of compression will be heard.

Downward Expansion

Strong multi-band compression allows you to raise the overall volume level—resulting in a dramatic increase of the existing noise floor. Downward expansion allows you to reduce or suppress this noise. Each frequency band features a downward expander. This works as the exact counterpart to the compressor: while the compressor compresses the dynamic range of the higher volume levels, the downward expander expands the dynamic range of the lower volume levels. With downward expansion, the signal will be reduced in level when it is lower than the defined *Threshold* level. The effect can be compared to a noise gate, but rather than simply cutting off the sound, it smoothly fades the volume using an adjustable Ratio.

Multipressor Parameters

Bands

This parameter (on the right side) determines the number of independently compressible frequency bands, and has a crucial impact on the amount of computing power needed for the effect. Classic multi-band compressors use three Bands.

Lookahead

Lookahead (just below the Bands parameter) determines how far the processor looks into the future, in order to react earlier (thus better) to peak volumes.

Set *Lookahead* to higher values, when the Peak/RMS control (see below) is set further towards RMS.

Peak/RMS

Adjusting the control between *Peak* (full left) and *RMS* (Root Meantime Square; full right) is dependent on the type of signal you would like to compress. An extremely short Peak detection setting is suitable for compression of short and high peaks of low power, which do not typically occur in music. The RMS detection method measures the power of the audio material over time and thus works much more musically. This is because human hearing is more responsive to the overall power of the signal than to single peaks. As a basic setting for most applications, the centered position is recommended.

Attack

Allows you to define the time (in milliseconds) required before compression is faded in.

Release

Sets the time required by the compressor (after compression) to release the effect. Just as with the other values, the best setting for this parameter depends greatly on the material to be compressed.

Multi-band Graphic

The graphic editor to the left of the Multipressor displays several settings, both graphically and numerically.

- Crossover Frequencies

The crossover frequencies (vertical borders) between the bands are variable. To change a band's crossover point, grab the borders directly within the graphic and move them to the left or to the right. The frequency is displayed numerically at the bottom of the graphic.

- Absolute Volume

The horizontal line in the middle of each band displays its current level (default: 0 dB). By grabbing the area below this line and moving it up and down, you can set the absolute volume level of the corresponding band. The level is displayed numerically at the bottom of the graphic. This ability allows the Multipressor to serve as a basic equalization tool, dependent on how the crossover frequencies are set.

- **Threshold Display**

The horizontal lines (up to three) in the lower area of the window represent the Threshold values for Compression (upper line), Expansion (middle line), and Reduction (bottom line). You can set these values by using the controls of the same name (see below).

Note: You can select the frequency band that you wish to edit by clicking in the lower section of each band.

Comp. Ratio

This, in conjunction with *Compression Threshold*, is the central parameter for compression. The *Comp. Ratio* determines the strength or rate of level reduction in the range you want to compress. In most cases, the most useful combinations of these two settings are either 1) low *Compression Threshold* and low *Comp. Ratio* or 2) high *Compression Threshold* and high *Comp. Ratio*.

Exp. Ratio

This, in conjunction with *Expansion Threshold*, is the central parameter for controlling downward expansion. It determines the strength of expansion applicable to the range that you wish to expand.

Graphic Curve

The graphic curve in the middle of the Multipressor shows the Ratio between input level (horizontal scale) and output level (vertical scale) of all bands. The colors correspond to the colors of the frequency bands in the left graphic area. Adjustments to the *Ratio* and *Threshold* controls allow you to change the curvature of the selected frequency band.

Thresholds

- **Compression**

Here, you can set the minimum level at which the compressor will begin to work. If the control is set all the way to the right (0 dB), the entire compressor section of the Multipressor is off-line. The further you move the control to the left, the lower the level above which the compressor will work.

- **Expansion**

Here, you can set the maximum level at which the expander should work. If the control is set all the way to the left (–50 dB), expansion will only occur on signals that fall below this level. (The *Exp. Ratio* can be set to a minimum of 1,2:1; below –50 dB the expansion always takes place at this low ratio.) The further the control is moved to the right, the higher the level below which the expander will work.

Reduction

Allows you to define the amount of noise level reduction (this is not a threshold value). If you move the control all the way to the left, the reduction will be at its maximum value (–50 dB). If the control is set all the way to the right, no reduction will occur.

Level Meter

In the Level Meter to the right, you may monitor either; the change of level caused by compression, or the output volume of each band, depending on whether you have selected *Gain change* or *Output* (see below). You can individually switch the bands on and off, in order to listen to single bands, by using the switches below the meters. If the switch below a band's meter is lit (light green), the band will be audible. If a switch is unlit, the band is muted.

Gain Change/Output

The Gain change and Output buttons can be used to switch the operating mode of the meters.

- If Gain change is selected, the meters indicate the strength of level reduction (to the audio material) by the compressor.
- If Output is selected, the meters show the absolute output level of the corresponding frequency band.

Master Gain

Allows you to reduce the overall gain increase (or increase the overall gain reduction), resulting from the Multipressor's settings.

This chapter introduces you to Logic's distortion effects.

This includes the Distortion, Overdrive, Bitcrusher, Clip Distortion, Phase Distortion, Distortion II, and Guitar Amp effect plug-ins.

Guitar Amp Pro



The Guitar Amp Pro plug-in simulates the sound of several famous guitar amplifiers and a number of cabinets/speakers. You can process guitar signals directly within Logic, allowing you to reproduce the sound of high-quality guitar amplification systems.

Guitar Amp Pro can also be used for experimental sound design and processing. You can freely use the plug-in on other instruments, as desired—applying the sonic character of a guitar amp to a trumpet or vocal part, for example!

Guitar Amp Pro offers a range of Amplifier, Speaker, and EQ models that can be combined in a number of ways. The EQ models are equipped with the Bass, Mid, and Treble controls typical of guitar amplifiers. Miking can be switched between two different microphone types and positions. To round out the complement of controls, Guitar Amp Pro also integrates two classic guitar effects, namely Vibrato and Tremolo.

The Guitar Amp Pro panel is divided into three areas. The upper part of the raised Y-shaped user interface contains the Amplifier parameters section. The bottom of this panel houses the Effect and Output section of the plug-in. The Speaker sections to the left and right provide access to the miking parameters of the virtual speaker.

Amp Section

In the upper area of the Amp section, you will find three pull-down menus. These allow you to set up your guitar amp as required, by selecting the appropriate Amp, Speaker, and EQ model(s).

Eleven different amplifier models can be accessed via the Amp menu to the left.

- UK Combo 30W—neutral sounding Amp model, well suited for clean or crunchy rhythm parts.
- UK Top 50W—quite aggressive in the high frequency range, well suited for classical rock sounds.
- US Combo 40W—clean sounding Amp model, well suited for funk and jazz sounds.
- US Hot Combo 40W—emphasizes the high mids of the frequency range, making this model ideal for solo sounds.
- US Hot Top 100W—this Amp model creates very fat sounds, even low *Master* settings result in broad sounds with a lot of “oomph”.
- Custom 50W—with the *Presence* parameter set to 0, this Amp model is well suited for smooth fusion lead sounds.
- British Clean—simulates the classic British Class A combos which have been continuously produced since the 1960s to the present, without any significant modification. This model is ideally suited for clean or crunchy rhythm parts.
- British Gain—reproduces the sound of a British tube head, and is synonymous with rocking, powerful rhythm parts and lead guitars with a rich sustain.
- American Clean—emulates the traditional full tube combos used for clean and crunchy sounds.
- American Gain—emulates a modern Hi-Gain head, making it suitable for distorted rhythm and lead parts.
- Clean Tube Amp—emulates a tube amp model with very low gain (distortion only when using very high input levels or Gain/Master settings).

The Speaker menu at the top right provides access to 15 speaker models.

- UK 1×12 open back—Classic open enclosure with one 12" speaker, neutral, well-balanced, multifunctional.
- UK 2×12 open back—Classic open enclosure with two 12" speaker, neutral, well-balanced, multifunctional.
- UK 2×12 closed—Loads of resonance in the low frequency range, therefore well suited for Combos: crunchy sounds are also possible with low Bass control settings.
- UK 4×12 closed slanted—when used in combination with off-center miking, you will get an interesting mid frequency range; therefore this model works well when combined with High Gain amps.

- US 1×10 open back—Not much resonance in the low frequency range. Suitable for use with (blues) harmonicas.
- US 1×12 open back 1—Open enclosure of an American lead combo with a single 12" speaker.
- US 1×12 open back 2—Open enclosure of an American clean/crunch combo with a single 12" speaker.
- US 1×12 open back 3—Open enclosure of another American clean/crunch combo with a single 12" speaker.
- US broad range—Cabinet simulation of a classic electric piano speaker.
- Analog simulation—Internal speaker simulation of a well-known British 19" tube preamplifier.
- UK 1×12—A British Class A tube open back with a single 12" speaker.
- UK 4×12—Classic closed enclosure with four 12" speakers (black series), suitable for Rock.
- US 1×12 open back—Open enclosure of an American lead combo with a single 12" speaker.
- US 1×12 bass reflex—Closed bass reflex cabinet with a single 12" speaker.
- DI Box—This option allows you to bypass the speaker simulation section.

The EQ models in the EQ menu refer to the simulated Amp models. Accordingly, the British 1, British 2, American, and Modern EQ models are available for the British Clean, British Gain, American Clean, and American Gain Amp models. You can, however, combine any Amp model with any EQ model, as required.

Directly below the EQ menu, you will find the Bass, Mid, and Treble controls. Use of these knobs allows you to adjust the frequency ranges of the EQ models as desired.

Presence is an additional high frequency control which exclusively affects the output stage (Master) of the Guitar Amp Pro plug-in.

The Link buttons between the menus link your menu selections. To explain, if the Link button between the EQ and Amp menus is active (yellow), selecting an Amp model will automatically load the corresponding EQ model. As mentioned earlier, you can, however, assign any other EQ model to the selected amp—via the EQ menu.

If the Link button between the Speaker and Amp menus is active, selecting an Amp model will automatically load the corresponding Speaker model. The cabinet assignment can be changed via the Speaker menu, as required.

To the far left of the Amp section, you will find the *Gain* knob, which controls the pre-amplification of the input signal. This control has different effects, dependent on the selected Amp model. As an example: A maximum Gain setting produces a powerful crunch sound when used in conjunction with the British Clean Amp model, but the same Gain setting results in a heavy distortion—suitable for lead sounds—with the British Gain or Modern Gain Amp models.

To the extreme right of the Guitar Amp Pro GUI, you will find the *Master* knob, which controls the output volume of the amplifier (to the speaker). Typically, in tube amplifiers, an increase in the Master control level produces a self-compressed and saturated sound, along with increased level, resulting in a more distorted and powerful amp signal. In the analog domain, this results in an extreme increase in loudness. In Guitar Amp, the Master control influences the sonic character, with the output level being set with the Output parameter (see below).

Speaker Section

Following the selection of a Speaker type, you make further adjustments to the miking parameters in the Speaker section.

The *Centered* and *Off-Center* buttons allow you to switch between two different microphone positions.

Centered aligns the microphone to the center of the speaker cone. This position is also called On Axis because the microphone capsule is approximately on the same axis (aligned) with the center of the speaker. In this position, the speaker sounds more full and powerful, making this setting suitable for Blues or Jazz guitar tones.

Off-Center aligns the microphone to the edge of the diaphragm. This placement is called Close Edge or Off Axis. The end result is an amplifier signal that is much brighter and sharper, but a little thinner. This position is more suitable for cutting rock or typical rhythm 'n' blues guitar tones.

The microphone Type, in conjunction with microphone placement, is equally essential for designing the required speaker sound.

When the *Condenser* button is active, a studio condenser microphone emulation is used for miking. The sound of condenser microphones is fine, transparent, and well balanced.

The *Dynamic* button switches to a dynamic cardioid microphone emulation. This microphone type sounds brighter and more cutting, in comparison to the Condenser model. At the same time, the lower Mids become less distinctive, making the Dynamic model more suitable for miking rock guitar tones.

Note: In practice, combining both microphone types can sound very interesting. Duplicate the guitar track, and insert Guitar Amp Pro as an insert effect on both tracks. Select different microphones in both Guitar Amp Pro instances, while retaining identical settings for all other parameters, and mix the track signal levels. You can, of course, choose to vary any other parameters, as desired.

Effect Section

The Effect section of Guitar Amp Pro contains the Tremolo and Vibrato effects—essentials in any guitar rig, and the Reverb.

Note: The Effect section is placed before the Master control in the signal flow, and therefore receives the preamplified signal (pre-Master).

In order to configure the Effect section, you must activate it via the *On/Off* buttons, found to the lower left of the *FX* and *Reverb* panels. When the respective effect section is active, the border of the On/Off button is highlighted.

In the upper middle portion of the FX section, you'll find the Effect menu, which allows you to select between the *Tremolo* and *Vibrato* effects. Tremolo modulates the amplitude (and therefore the volume), while Vibrato modulates the frequency (and therefore the pitch) of the signal. The intensity of the modulation is determined by the *Depth* parameter. *Speed* controls the modulation speed in Hz. Lower settings will produce a smooth and floating sound, with higher settings leading to a rotor-like effect. You can perfectly synchronize the modulation speed to the song tempo, if desired. To do so, simply press the *Sync* button, found beside the *Speed* control. Once synchronization mode is activated, the control range of the Speed control will display various musical values. Set the *Speed* control to the desired value, and your Guitar Amp Pro modulation will be perfectly synchronized to the song tempo.

The Reverb portion of the Effect section contains two controls. *Level* determines the amount of reverb signal applied to the pre-amplified signal. The pull-down menu to the right allows you to select one of three different Spring reverb models.

Output

The Output knob serves as a final level control for Guitar Amp's output.

The Output parameter can be viewed as a volume control “behind the cabinet,” and is used to set the level that is fed into ensuing plug-in slots on the channel, or into the channel output.

Note: This parameter is very distinct from the Master control, which serves a dual purpose—for sound design, as well as controlling the level of the Amp section.

Distortion



This distortion effect simulates the lo-fi dirt generated by a bipolar transistor.

Move the *Drive* slider up to increasingly saturate the transistor. Generally, the distortion created by the plug-in tends to increase the signal level, an effect that you can compensate for with the *Output* slider. The *Tone* knob filters the harmonics-laden distortion signal, delivering a somewhat less grating, softer tone.

The Distortion Eye is watching—it visually represents the *Drive* and *Tone* parameter settings.

Overdrive



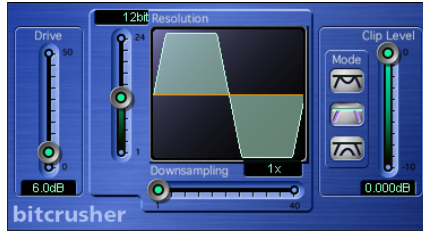
The Overdrive effect emulates the distortion of a field-effect transistor (FET). When saturated, FETs generate warmer sounding distortion than bipolar transistors.

The *Drive* slider pushes the transistor over the edge and into overdrive. Generally, the distortion created by the plug-in tends to increase signal levels, an effect that you can compensate for with the *Output* slider.

The *Tone* knob lets you filter the harmonics-laden distortion signal, which delivers an even warmer sound.

The Distortion effect's Eye visually represents the settings of the *Drive* and *Tone* parameters.

Bitcrusher



Bitcrusher is the ultimate digital distortion box. You can do all kinds of wild stuff with it, such as recreate the 8-bit sound of the pioneering days of digital audio, create artificial aliasing by dividing the sample rate, or distort signals so radically that they are rendered unrecognizable.

Warning: The Bitcrusher can damage your hearing (and speakers) when operated at high volumes.

The *Drive* slider boosts the level at the input of the Bitcrusher. Please note that this tends to excite the clipping stage located at the output of the Bitcrusher as well.

The *Resolution* knob allows you to reduce the resolution from 24 bits down to 1 bit.

The number of bits is always an exponent of two. The range of available values is equivalent to the exponents of two that a given sample rate can handle. As an example, 65,536 different values are possible for 16 bits, whereas at 8 bits, you're left with just 256. The sonic image becomes ever more ragged as the values decrease because the number of sampling errors increases, thus generating more distortion. At extremely low bit resolutions, the amount of distortion can be greater than the level of the usable signal.

The *Downsampling* slider lowers the sample rate. As an example, at a value of 2 (halved), the original 44.1 kHz signal is sampled at a rate of just 22.05 kHz. At a factor of 10, the rate is knocked all the way down to 4.41 kHz.

The *Clip Level* slider lets you define the point below the normal threshold that you want the signal to start clipping. The *Mode* buttons are used to determine whether the signal peaks that exceed the clip level are Folded, Cut, or Displaced (check out the graphics on the buttons and the resulting waveform in the display). The kind of clipping that occurs in standard digital systems is usually closest to that of the center mode (Cut). Internal distortion may generate clipping similar to the types generated by the other two modes.

Clip Distortion



The Clip Distortion plug-in is a non-linear distortion effect that produces unpredictable spectra. Beyond drastic distortions, it's well suited for the simulation of warm tube overdrive sounds.

The best way to learn what effect the various parameters have is to experiment with them on different signal sources. As a starting point, the following describes what each control basically does:

The signal is first amplified by the *Drive* value, which is a simple gain control. The signal then passes through a highpass filter. The filter's cutoff frequency is determined by the *Tone* control. The actual non-linear distortion process is controlled by the *Symmetry* parameter.

Once the signal has been distorted asymmetrically, the signal passes through a lowpass filter. This filter's cutoff frequency is determined by the *Filter* fader. The *Mix* parameter combines the effected signal with the dry signal. This mixed signal then passes through yet another lowpass filter, where the cutoff frequency is controlled by the *Sum Filter* parameter. All filters have a slope of 6 dB/Oct.

The last stage of signal processing is a tunable shelving filter. If you set its *Frequency* to about 12 kHz, it will behave like a normal treble control, as found in any mixer's channel strip or stereo hi-fi amplifier. Unlike such treble controls, this filter allows for boosts or cuts of up to ± 30 dB (Gain parameter). This somewhat unorthodox combination of serially connected filters allows for gaps in the frequency spectra that can sound quite good with this sort of non-linear distortion. The clip circuit graphic visually depicts every parameter, with the exception of the shelving filter controls.

There are two more parameters in the Controls view: Input Gain and Output Gain. These can be used to raise/lower the input and output signal levels by up to 30 dB.

Phase Distortion



The Phase Distortion plug-in is based on a modulated delay line, much like the well-known chorus or flanger effects. As opposed to these, the delay time is not modulated by a low frequency oscillator (LFO), but rather by a lowpass-filtered version of the audio input signal itself. This is how the signal modulates its own phase position.

In the signal flow of this effect, the parameters do the following:

The input signal only passes the delay line and is not affected by any other process. *Mix* blends the effected signal with the original signal. The delay time is modulated by a Side Chain signal—namely, the input signal. The input signal passes through a resonant lowpass filter, the *Cutoff* frequency and *Resonance* of which can be set with dedicated controls. You also can listen to the filtered Side Chain (instead of the Mix signal), if you engage *Monitor*. The maximum delay time is set with *Max Modulation*. The amount of modulation itself is controlled with *Intensity*.

In the Controls view, there is one more parameter which can't be seen in the Plug-in window. It is only valid for the stereophonic version of the effect. Normally, a positive input value results in a longer delay time. If you engage *Phase Reverse* (On), positive input values result in a reduction of the delay time on the right channel only.

Distortion II



The Distortion II plug-in is based on the EVB3's distortion effect. More information about its parameters can be found in the EVB3's “Distortion” section on page 473.

The Distortion II plug-in offers one additional parameter: the PreGain knob. This allows you to raise the input signal by up to *20 dB* or lower it by as much as *10 dB*, in order to provide a broader range of distortion colors.

This chapter covers Logic's filter effects.

The filter effects include the AutoFilter, Fuzz-Wah, EVOC 20 FB, EVOC 20 TO, Low/High Pass Filter and Low/High Cut plug-ins. The EVOC 20 TO is based on a vocoder. Further information about vocoders can be found in the chapter “Vocoder—Basics” on page 167.

AutoFilter



The AutoFilter is an extremely versatile, resonance-capable lowpass filter, that offers a couple of truly unique features. The most important parameters are located to the right side of the Plug-in window: The *Cutoff Freq.* knob determines the point where the filter kicks in. Higher frequencies are attenuated, lower frequencies are allowed to pass through.

The *Resonance* knob emphasizes the frequency range surrounding the cutoff frequency. When you turn the Resonance up sufficiently, the filter itself begins oscillating (at the cutoff frequency). Self-oscillation is initiated before you max out the Resonance parameter, just like the filters on the legendary Minimoog. When working with Resonance, the manner in which the lowpass filter allows frequencies to pass changes: higher Resonance values cause the filter to cut the bottom end, making the signal sound thinner. The *Fatness* parameter compensates for this audio artefact. When you turn Fatness up to its maximum value, the Resonance setting has no effect on the response of the frequencies below the cutoff frequency.

The *Slope* buttons determine the steepness of the lowpass filter: frequencies above the cutoff frequency are dampened by 6, 12, 18, or 24 dB per octave (in audio jargon, these are called filters of the 1st, 2nd, 3rd, and 4th order). Even though the 24 dB filter is largely the component of choice for synthesizer designers, be sure to experiment with the other options, as they can also deliver pretty hip results. The *Distortion Input* and *Output* parameters allow you to individually control each of the two distortion units—one pre-input and the other post-output. Although the two distortion modules are identical, their respective positions in the signal chain—before and after the filter, respectively—enable them to generate remarkably different sounds.

All other AutoFilter parameters are used to dynamically modulate the cutoff frequency. These fall into two sections: *Envelope* (ADSR, Envelope Generator) and *LFO* (Low Frequency Oscillator, Modulation Generator).

The *Threshold* parameter applies to both sections, and analyzes the level of the input signal. If the input signal level exceeds that of the variable Threshold level, the envelope and LFO are retriggered. The *Modulation* slider of each section determines the intensity of the control signal's effect on the cutoff frequency.

Envelope: when the Threshold level is exceeded, the control signal is triggered at the minimum value. Following a variable interval, the length of which is determined by the *Attack* parameter, the signal reaches its maximum value. It drops in level during the interval defined by the *Decay* value, and ends up at the *Sustain* value. Once the signal level drops below the Threshold value, it falls all the way to its minimum value over the time determined by the *Release* parameter. If the input signal falls below the *Threshold* level before the control signal has reached the *Sustain* level, the Release phase is triggered. The *Dynamic Modulation* parameter lets you modulate the peak value of the Envelope section, by using the level of the input signal.

LFO: the wave shape used for LFO oscillation is determined by the Waveform buttons. The choices are: descending sawtooth (saw down), ascending sawtooth (saw up), triangle, pulse wave, or random (random values, Sample & Hold). Once you've selected a waveform, you can shape the curve with the *Pulsewidth* knob. Use the *Frequency* knobs to define the desired LFO frequency: *Coarse* sets a value between 0.1 and 10,000 Hz, *Fine* lets you adjust it in smaller increments. The *Speed Mod.* (Speed Modulation) knob is used to modulate the LFO frequency independently of the input signal level. If the input signal exceeds the Threshold level, the modulation width of the LFO increases from 0 to the value specified for *Modulation*. You can also define the amount of time this process takes, by entering the desired value with the *Delay* knob. If the *Sync* button is activated, the waveform is started at 0° as soon as the Threshold is exceeded.

Whenever you use the AutoFilter as a stereo plug-in, you can determine the phase relationships of the LFO modulations on the two stereo sides, with the *Stereo Phase* knob.

There are five additional parameters in the Controls view of the Autofilter.

The *Volume* parameter can lower the Volume by as much as -50 dB, allowing you to compensate for higher levels when using *Distortion*, for example. If you switch *Beat Sync* to On, the LFO is synchronized to the sequencer's tempo. The speed values include bar values, triplet values and more. These are determined by the *Rate* slider directly below Beat Sync. Use *Sync Phase* to shift the phase relationship between the LFO and the sequencer. *Dry Signal* sets the level ratio/portion of the non-effected (dry) signal.

Fuzz-Wah



The Fuzz-Wah effect is the standalone plug-in version of the EVD6's Wah effect. It incorporates additional compressor and distortion (Fuzz) facilities, and features some additional parameters over the integrated EVD6 Wah. These are outlined below.

Parameters of the Fuzz-Wah

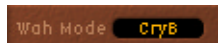
FX Order



This parameter allows to you select the order in which the Fuzz/Wah effects are placed. Choices are: *Fuzz –Wah* or *Wah–Fuzz*.

Wah Mode

There are simulations of several classic wah effects, as well as some basic filter types available. Available models are: *off*, *ResoLP*, *ResoHP*, *Peak*, *CryB*, *Mor11*, *Mor12*.



Wah Level

Can be used to adjust the level of the wah-filtered signal, relative to the original level. Also see the *Auto Gain* section below.

Auto Gain

While sweeping through the main formants of the input signal, the output level of the Wah may vary wildly, which is not always desirable. Activating the *Auto Gain* parameter will automatically compensate for this side-effect. Range: *on/off*



To hear the difference *Auto Gain* can make:

- Switch Auto Gain to *on*.
- Raise the effect level to a value just below the mixer's clipping limit.
- Make a sweep with a high *relative Q* setting.
- Now switch Auto Gain to *off*, and repeat the sweep.

Warning: Please take care while doing this, or your ears and speaker system may be damaged.

Relative Q

The quality of the main filter peak can be increased/decreased, relative to the model setting, thereby obtaining a sharper/softer wah sweep. When set to a value of 0, the original setting of the model is active. Range: -1.00 to $+1.00$ (0.00 is the default)



Pedal Range

Common MIDI foot pedals have a much larger mechanical range than most classic Wah pedals.



The exact sweep range of the wah filter effected by the MIDI foot pedal is set with the Pedal Range parameters. The highest and lowest possible value reached by the pedal is graphically represented by a gray bracket around the *Pedal Position* fader (see below). The left and right limit is set by clicking and moving it with the mouse. Additionally both values can be moved simultaneously by clicking in the center of the bracket and moving it to the left or right.

Pedal Position

This parameter represents the current position of the Wah pedal.

To control and automate the Pedal Position via an external MIDI controller for example a MIDI pedal, your Logic environment has to be prepared accordingly. For more information please read “Controlling the EVD6 via MIDI” on page 502.

AutoWah Depth

In addition to using MIDI foot pedals (see above), the wah effect can be controlled using the Auto Wah facility. The sensitivity of the Auto Wah can be set with the *Depth* parameter. Range: 0.00 to 100. (See also the “Envelope (Depth)” section, from page 500 onwards.)



AutoWah Attack/Release

These parameters allow you to define how much time it takes for the Wah filter to open and close. Range (in milliseconds): 10 to 10,000



Comp Ratio

The Comp Ratio of the integrated compressor can be adjusted between 1:1 (no compression) and 30:1. The Compressor is tied to the Fuzz effect, and always precedes it. As such, the *FX Order* parameter is very important for placement of the Compressor in the effects chain.



Fuzz Gain

Controls the level of Fuzz (distortion). Range: 0 dB to 20 dB.

Fuzz Tone

The integrated Fuzz effect can be adjusted, tonally, with this parameter. Range: 2000 Hz to 20,000 Hz

EVOC 20 Filterbank

The EVOC 20 FB consists of two formant filter banks, which are also used in Logic's EVOC 20 PS vocoder. More information on the filter banks are found in "How Does a Filter Bank Work?" on page 168.

The input signal runs through both filter banks in parallel. Each bank features independent volume faders for each band, allowing levels to be set freely—ranging from unchanged through to silence. The latter completely suppresses the selected formants in the overall sound spectrum. Use of the *Formant Stretch* and *Formant Shift* parameters provide total control over the position and width of the filter bands. In addition, you can also crossfade between the two filter banks.

Parameters of the EVOC 20 FB

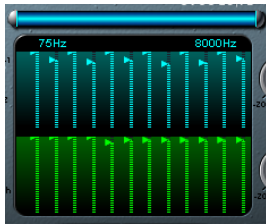


The EVOC 20 FB interface is divided into three main sections. These are the *Formant Filter*, *Modulation*, and *Output* areas.

The Formant Filter Area

The Formant Filter Window

The Formant Filter window is divided into two sections by a horizontal line. The upper half applies to the Filter Bank A, and the lower half to the Filter Bank B.



The individual vertical bars in each bank of settings are faders which represent the level of a particular frequency band/formant. To adjust each fader, simply click-hold on the desired bar and drag up or down.

Complex bar curves are easily created by “painting” them in: Click and hold the mouse button next to a bar on the blue or green portion of the background, and drag left or right over the bars within the editing field. The length of the bars will be adjusted in accordance with the mouse movement. This method makes editing multiple frequency band levels quick and convenient.

Bands

The *Bands* parameter determines the number of frequency bands used by the EVOC 20 FB. It ranges from 5 to 20.



Note: Increasing the number of bands also increases the processor overhead.

High/Low Frequency

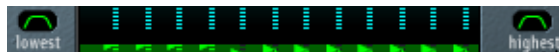
The blue bar shown just beneath the EVOC label is a multi-part control which is used to determine the lowest and highest frequencies allowed to pass by the filter. Frequencies which fall outside these boundaries will be cut. All filter bands are distributed evenly across the range defined by the *High/Low Frequency* values.

- To adjust the Low Frequency value, simply click-hold on the silver slider to the left of the blue bar, and drag to the right (or left). The value range is 75–750 Hz.
- To adjust the High Frequency value, simply click-hold on the silver slider to the right of the blue bar, and drag to the left (or right). The value range is 800–8000 Hz.
- To adjust both sliders simultaneously, click on the area between the slider halves (directly on the blue bar) and drag to the left or right.
- You can make changes to the *High/Low Frequency* values directly by using your mouse as a slider on the numerical entries—80 and 8000 Hz in the diagram.

Lowest/Highest

These parameters can be found in the two small switches on either side of the Formant Filter window. These switches determine whether the lowest and the highest filter bands are bandpass filters (just like all the bands between them), or whether they act as lowpass/highpass filters, respectively. Click once on them to switch between the two curve shapes available.

- In the *Bandpass* setting, the frequencies below/above the lowest/highest bands are ignored.
- In the *Highpass* or *Lowpass* setting, all frequencies below the lowest (or above the highest) bands will also be treated.



Slope

The pull-down menu *Slope* determines the amount of filter slope applied to all filters of both filter banks. Choices are 1 (filter attenuation of 6 dB/Oct.) and 2 (filter attenuation of 12 dB/Oct.): 1 sounds softer, 2 sounds tighter.

Boost A/B Controls

The *Boost A* and *Boost B* knobs allow an increase or cut in the overall gain of the A and B filter banks. Their range is ± 20 dB. To adjust, click-hold and drag up or down with the mouse.

Note: You will need these controls, as the filter bank achieves its sounds by turning down the level of one or more filter bands. To make up for the resulting energy loss, use *Boost*.

Note: *Boost* is also quite handy to adjust the levels of both filter banks to each other, so that using *Fade A/B* (see below) leads only to a sound color change, but not to a level change.

Fade AB Control

The *Fade AB* crossfades between the A and B filter bank. At its extreme top or bottom position, you will only hear one of the filter banks.

Formant Shift



Moves the position of all bands in both filter banks up and down the frequency range.

Note: You can jump directly to the values -0.5 , -1 , 0 , $+0.5$ and $+1.0$ by clicking on their numbers.

Resonance



Resonance is responsible for the basic sonic character of both filter banks: low settings give it a soft character, high settings will lead to a more snarling, sharp character. Increasing the value for *Resonance* emphasizes the middle frequency of each frequency band.

Modulation Parameters



The Modulation (LFO) area controls the *Formant Shift* and *Fade A/B* parameters of the EVOC 20 FB. It allows synchronous/non-synchronous modulation in bar, beat (triplet) or free values.

- *LFO Shift*, on the left-hand side, controls *Formant Shift* modulation of the filter bands.
- *LFO Fade* controls the *Fade AB* parameter.

Waveform

The Waveform switches allow the selection of the waveform type used by *LFO Shift* and *LFO Fade*. A selection of Triangle, falling and rising Sawtooth, Square up and down around zero (bipolar), Square up from zero (unipolar), a random stepped waveform (S&H), and a smoothed random waveform is available for each LFO.



Intensity

The *Intensity* sliders control the amount of *Formant Shift* and *Fade A/B* modulation by the respective LFOs.

Rate

These knobs determine the speed of the modulation. Values to the left of the center positions are synchronized with the sequencer's tempo and include bar values, triplet values and more. Values to the right of the center positions are non-synchronous and displayed in Hertz.

Note: The *Formant Shift* and *Fade* LFO modulations are the keys to the most extraordinary sounds of the EVOC 20 FB: Make sure to set up totally different or complementary filter curves in both filter banks. Use rhythmic material like a drumloop as an input signal. Set up tempo-synchronized modulations—with different *Rates* for each LFO—for *Formant Shift* and *Fade A/B*. And then try a tempo-synchronized Tape Delay after the EVOC 20 FB. You will end up with unique rhythms.

Output Section



Overdrive

This switch enables/disables the *Overdrive* circuit of the EVOC 20 FB.

Note: To actually hear the Overdrive effect, you may need to boost the level of one or both filter banks.

Level

The *Level* slider controls the level of the EVOC 20 FB's output signal.

Stereo Mode

This pull-down menu determines the input/output mode of the EVOC 20 FB. Choices are: *m/s*—mono input to *stereo* output and *s/s*—*stereo* input to *stereo* output.

- The *Stereo Mode* should be set to *m/s* if the signal going into the EVOC 20 FB is monophonic, for example a mono audio track.
- Stereo/stereo (*s/s*) is the preferred setting for stereo input signals. In this case, the stereo signal is processed by separate filter banks for the left and right channels. When using the *m/s* Mode on stereo input signals, the stereo signal is first summed to mono before it is passed on to the filter banks.

Stereo Width

Stereo Width distributes the output signals of the filter bands in the stereo field.

- In the left position, the output of all bands are centered.
- In the center position, the output of all bands ascends from left to right.
- In the right position, the bands are output evenly on the left and the right channel.

The stereo/stereo mode (*s/s*) uses one A/B filter bank per channel. The positioning of the frequency bands correspond to that described above, but the bands of each filter bank ascend in opposing directions, from left to right.

MIDI Controllers Received

The following tables show the CC numbers used when the following MIDI preference is active: *Options > Settings > MIDI Options > (Version 4.x behavior)*.

Filter Bands	Band Level	Bank A	Bank B
	Band 1	CC #64	CC #96
	Band 2	CC #65	CC #97
	Band 3	CC #66	CC #98
	Band 4	CC #67	CC #99
	Band 5	CC #68	CC #100
	Band 6	CC #69	CC #101
	Band 7	CC #70	CC #102
	Band 8	CC #71	CC #103
	Band 9	CC #72	CC #104
	Band 10	CC #73	CC #105
	Band 11	CC #74	CC #106
	Band 12	CC #75	CC #107
	Band 13	CC #76	CC #108
	Band 14	CC #77	CC #109
	Band 15	CC #78	CC #110
	Band 16	CC #79	CC #111
	Band 17	CC #80	CC #112
	Band 18	CC #81	CC #113
	Band 19	CC #82	CC #114
	Band 20	CC #83	CC #115
Boost	Bank A	CC #84	
	Bank B	CC #116	
Fade A/B		CC #117	
Formant Filter	Bands	CC #85	
	FF Low Freq	CC #88	
	FF Hi Freq	CC #89	
	Formant Shift	CC #90	
	FF Resonance	CC #91	
	Slope	CC #92	
	FF Low/Bandpass Select	CC #119	
	FF High/Bandpass Select	CC #120	

Filter Bands	Band Level	Bank A	Bank B
LFO 1 (Shift)	LFO 1 Rate	CC #93	
	LFO 1 Waveform Select	CC #94	
	LFO 1 Intensity	CC #95	
LFO 2 (Fade)	LFO 2 Rate	CC #121	
	LFO 2 Waveform Select	CC #122	
	LFO 2 Intensity	CC #123	
Output	Level	CC #124	
	Mono/Stereo Select	CC #87	
	Stereo Width	CC #86	

EVOC 20 TO

The EVOC 20 TO is a vocoder equipped with a monophonic pitch *tracking oscillator*, hence the TO in its name (more information about vocoders can be found in the chapter “Vocoder—Basics” on page 167). Non-technically, this allows the EVOC 20 TO to follow (track) the pitch of a monophonic incoming audio signal. If the audio signal is a vocal melody, for example, the individual pitches of the sung notes will be tracked and mirrored by the *Synthesis* engine.

The EVOC 20 TO derives its synthesis signal from its monophonic tracking oscillator. Alternatively to the tracking oscillator, the EVOC 20 TO can use a freely selectable audio signal as the synthesis signal.

Note: For good pitch tracking, it is essential that the signal is monophonic (one pitch at a time) and as dry as possible. Absolutely avoid background noises. As an example, using a voice already processed with even a slight reverb will deliver pretty strange results. The results will be even stranger when signals with no audible pitch are used—such as drumloops. The resulting artefacts might, however, be exactly what you’re after in some situations.

It should be noted that the EVOC 20 TO uses a Side Chain, allowing the use of another track as the analysis and/or synthesis signal. In the gray area at the top of the Plug-in window, click-hold on the *Side Chain* pull-down menu, and select the desired Audio track. In the Mixer, adjust the volume levels of the EVOC 20 TO and the audio track used for the *Side Chain* to taste.

The EVOC 20 TO is not limited to pitch tracking effects. It can vocode a signal by itself, making it very useful for unusual filter effects. Try this with different *Resonance*, *Formant Shift* and *Formant Stretch* settings.

As both analysis and synthesis input signals are freely selectable, you can even vocode an orchestra with train noises, for example.

The EVOC 20 TO can be used in the insert slots of Audio, Audio Input, Bus, Master, and Audio Instrument channels.

The signal path of the EVOC 20 TO is shown in the block diagram on page 190.

Parameters of the EVOC 20 TO



The EVOC 20 TO interface is divided into five main sections. From left to right, these are the *Analysis/Synthesis*, *Formant/Filter*, *Modulation*, *Unvoiced/Voiced (U/V) Detection* and *Output* areas.

Analysis In Parameters

Attack

The *Attack* parameter determines how quickly each *envelope follower* coupled to each *Analysis* filter band reacts to rising signals. Longer *Attack* times result in a slower tracking response to transients of the *Analysis* input signal.

Note: A long *Attack* time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulated vocoder effect. Set *Attack* as low as possible to achieve precise articulation.

Release

The *Release* parameter determines how quickly each *envelope follower* coupled to each *Analysis* filter band reacts to falling signals. Longer *Release* times make transients of the *Analysis* input signal sound for a longer period of time at the Vocoder's output.



Note: A long *Release* time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulated vocoder effect. *Release* times that are too short result in rough, grainy vocoder sounds. *Release* values of around 8 to 10 ms have proven to be useful starting points.

Freeze

The *Freeze* button holds the current analysis sound spectrum indefinitely.

The “frozen” *Analysis* signal can capture a particular characteristic of the source signal, which is then imposed as a complex sustained filter shape on the *Synthesis* section.

Using a spoken word pattern as a source, for example, the *Freeze* parameter could capture the attack or tail phase of an individual word within the pattern—the vowel *a*, for example.

With *Freeze* engaged, the *Analysis* filter bank ignores the input source until it is disengaged.

Another use of the *Freeze* parameter (which can be automated) could be to compensate for people’s inability to sustain sung notes for a long period without taking a breath. If the *Synthesis* signal needs to be sustained, when the *Analysis* source signal (a vocal part) isn’t, *Freeze* can be used to lock the current formant levels (of a sung note), even during gaps in the vocal part—between words in a vocal phrase.

Note: When the *Freeze* parameter is used, the *Attack* and *Release* parameters have no effect.

Analysis In (pull-down menu)

This pull-down menu determines the *Analysis* signal source—*Track* or *Side Chain*. To switch between them, use the mouse as a slider and drag vertically.



- *Track*—allows you to use the audio track, into which the EVOC 20 TO is inserted, as the *Analysis* signal.
- *Side Chain*—allows you to use another audio track as the *Analysis* signal. The selection of the actual *Side Chain* source track is achieved by click-holding on the *Side Chain* pull-down menu in the gray area at the top of the Plug-in window.

Note: If no *Side Chain* track is assigned in the *Side Chain* pull-down menu, the track’s signal will be used.

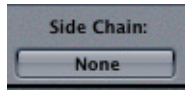
Synthesis In Parameter

Synthesis In (pull-down menu)

This pull-down menu determines the *Synthesis* signal source—*Osc(illator)*, *Track* or *Side Chain*. To switch between them, use the mouse as a slider and drag vertically.

- *Oscillator*—allows you to use the built-in monophonic tracking oscillator. The oscillator tracks the pitch of the *Analysis* input signal. Selection of the *Oscillator* activates the other parameters in the *Synthesis* section. If *Osc* is not selected, the *FM Ratio*, *FM Int* and other parameters in this area have no effect.

- *Track*—allows you to use the audio track, into which the EVOC 20 TO is inserted, as the *Synthesis* source signal.
- *Side Chain*—allows you to use another audio track as the source material for the *Synthesis* section. Selection of the Side Chain track is achieved by click-holding on the *Side Chain* pull-down menu in the gray area at the top of the Plug-in window.



Note: If no Side Chain track is assigned in the Side Chain pull-down menu, the track's signal will be used.

Bands

The *Bands* window determines the number of frequency bands used by the EVOC 20 TO. It ranges from 5 to 20. Adjustments are made by using the mouse as a slider. The greater the number of bands, the more precisely the sound can be reshaped.



Note: Increasing the number of bands also increases the processor overhead.

The Tone Generator of the Tracking Oscillator

Depending on the position of the *FM Int* control, the tracking oscillators delivers either; a sawtooth wave or the signal of an FM tone generator.

The FM tone generator consists of two oscillators, each generating a sine wave. The frequency of Oscillator 1 is linearly modulated by Oscillator 2. This deforms the sine wave of Oscillator 1 to a waveform with rich harmonic structure. Its harmonic structure depends on the modulation intensity and the frequency ratio of both oscillators.

Tune

Coarse Tune offsets the pitch of the oscillator in semitones by up to ± 2 octaves.



Fine Tune: The default value is concert pitch A = 440 Hz. The range is from 425.00 to 455.00 Hz.

FM Int



This knob selects the basic waveform and controls the intensity of FM modulation.

- If set to 0, the FM tone generator is disabled, and a sawtooth wave is generated instead.
- If set to values higher than 0, the FM tone generator is activated. Higher values result in a more complex and brighter sound.

FM Ratio

The *FM Ratio* (value range 0.5 to 3.5) knob defines the ratio between the Carrier and Modulator frequencies—the frequencies of Oscillators 1 and 2. This setting defines the basic character of the sound.

With even-numbered values or their multiples, harmonic sounds are produced. With odd-numbered values or their multiples, inharmonic sounds are produced, which we perceive as being metallic sounding.

- An *FM Ratio* of 1.000 produces results resembling a sawtooth waveform.
- An *FM Ratio* of 2.000 produces results resembling a square wave with a pulse width of 50%.
- An *FM Ratio* of 3.000 produces results resembling a square wave with a pulse width of 33%.

FM Ratio is only relevant if *FM Int* is not set to 0.

Pitch Quantize

The *Pitch Quantize*, *Root/Scale* and *Max Track* controls, in conjunction with the piano keys of the onscreen keyboard, control the automatic pitch correction facility (Pitch Quantize) of the tracking oscillator. *Pitch Quantize*, in conjunction with the *Root/Scale* and *Max Track* parameters, can be used to constrain the pitch of the tracking oscillator to a scale or chord. This allows subtle or savage pitch corrections, and can be used creatively on unpitched material with high harmonic content, such as cymbals and high-hats. To use pitch quantization, the *Strength* parameter must be set above a value of zero, and at least one of the onscreen keyboard keys needs to be activated.



- *Strength*—determines how pronounced the automatic pitch correction is.
- *Glide*—determines the amount of time the pitch correction takes, allowing sliding transitions to the quantized pitches.

Root/Scale

The *Root* and *Scale* parameters, in combination with the onscreen keyboard, define the pitch(es) that the tracking oscillator is quantized to.



- If you click on the value shown below the word *Scale*, a pull-down menu opens. Here you can select a scale or chord. See the listing of preset scales and chords shown alongside.
- *Root* selects the root key of the respective scale or chord. The *Root* parameter is not displayed when *chromatic* or *user* is selected.
- Any combination of keys can be activated by clicking on the notes of the onscreen keyboard. Activated keys are illuminated. To disable any active notes on the keyboard, simply click on the note a second time. The *Scale* display will change to *user* as soon as any key is edited.
- The previously displayed scale or chord is used as the starting point when creating a user scale. This allows you to select a preset scale or chord, and then modify it by clicking on the notes of the onscreen keyboard.
- The last edit will be remembered. You can select a new preset scale or chord, and as long as you don't make any changes you can always jump back to the previously set *user* scale.

As with all Logic plug-ins, the *Root* and *Scale* parameters, and the keys of the onscreen keyboard can be automated.

Max Track

This parameter cuts the high frequencies of the analysis signal, making the pitch detection more robust. Should the pitch detection produce unstable results, reduce the *Max Track* parameter value to the lowest possible setting.

Formant Filter

The Formant Filter Window

The Formant Filter window is divided into two sections by a horizontal line. The upper half applies to the *Analysis* section and the lower half to the *Synthesis* section. Changes made to the *High/Low* frequency parameters, the *Bands* parameter or the *Formant Stretch* and *Formant Shift* parameters will result in visual changes to the Formant Filter window. This provides you with invaluable feedback on what is happening to the signal as it is routed through the two formant filter banks.



High/Low Frequency

The blue bar shown just beneath the EVOC 20 TO logo is a multi-part control which is used to determine the lowest and highest frequencies allowed to pass by the filter section. The length of the blue bar represents the frequency range for analysis and synthesis. Frequencies of any audio input which fall outside these boundaries will be cut. All filter bands are distributed evenly across the range defined by the *High/Low Frequency* values.

- To adjust the low frequency value, simply click-hold on the silver slider to the left of the blue bar, and drag to the right (or left). The value range is 75–750 Hz.
- To adjust the high frequency value, simply click-hold on the silver slider to the right of the blue bar, and drag to the left (or right). The value range is 800–8000 Hz.
- To adjust both sliders simultaneously, click on the area between the slider halves (directly on the blue bar) and drag to the left or right.
- You can make changes to the High/Low Frequency values directly by using your mouse as a slider on the numerical entries—80 and 8000 Hz in the diagram.

Lowest/Highest

These parameters can be found in the two small fields on either side of the Formant Filter window. These switches determine whether the lowest and highest filter bands are bandpass filters (just like all the bands between them), or whether they act as lowpass/highpass filters, respectively. Click once on them to switch between the two curve shapes available.

- In the *bandpass* setting, the frequencies below/above the lowest/highest bands are ignored on analysis and synthesis.
- In the *highpass* (or *lowpass*) setting, all frequencies below the lowest (or above the highest) bands will be considered for analysis and synthesis.

Formant Stretch

This parameter alters the width and distribution of all bands in the *Synthesis* filter bank, extending or narrowing the frequency range defined by the blue bar (*Low/High Frequency* parameters) for the *Synthesis* filter bank.

If *Formant Stretch* is set to 0, the width and distribution of the bands in the *Synthesis* filter bank is equal to the width of the bands in the *Analysis* filter bank. Low values narrow the width of each band, while high values widen it. The control range is from 0.5 to 2 (as a ratio of the overall bandwidth).

Note: You can jump directly to a value of 1 by clicking on its number.

Formant Shift



Formant Shift moves the position of all bands in the *Synthesis* filter bank up and down. With *Formant Shift* set to 0, the position of the bands in the *Synthesis* filter bank is equal to the position of the bands in the *Analysis* filter bank. Positive values will move the bands up in frequency, while negative values will move them down in respect to the *Analysis* filter bank.

Note: You can jump directly to the values -0.5 , -1 , 0 , $+0.5$ and $+1$ by clicking on their numbers.

Note: When combined, *Formant Stretch* and *Formant Shift* alter the formant structure of the resulting vocoder sound, and can result in some interesting timbre changes. As an example, using speech signals and tuning *Formant Shift* up results in Mickey Mouse effects.

Note: *Formant Stretch* and *Formant Shift* are especially useful if the frequency spectrum of the *Synthesis* signal does not complement the frequency spectrum of the *Analysis* signal. You could create a *Synthesis* signal in the high frequency range from an *Analysis* signal which mainly modulates the sound in a lower frequency range, for example.

Resonance



Resonance is responsible for the basic sonic character of the Vocoder: low settings give it a soft character, high settings will lead to a more snarling, sharp character. Increasing the value for *Resonance* emphasizes the middle frequency of each frequency band.

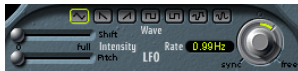
Note: The use of either, or both, of the *Formant Stretch* and *Formant Shift* parameters can result in the generation of unusual resonant frequencies when high *Resonance* settings are used.

Modulation Parameters

The *LFO* can modulate ...

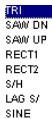
- the *frequency (Pitch)* of the tracking oscillator (vibrato) or
- the *Formant Shift (Shift)* parameter of the *Synthesis* filter bank.

It allows synchronous/non-synchronous modulation in bar, beat (triplet) or free values.



Wave

The *Wave* switches allow the selection of a waveform type to be used by the *LFO*. A selection of Triangle, falling and rising Sawtooth, Square up and down around zero (bipolar, good for trills), Square up from zero (unipolar, good for changing between two definable pitches), a random stepped waveform (S&H), and a smoothed random waveform is available. Simply click on the appropriate button to select a waveform type.



Intensity

The *Intensity* sliders control the amount of *Formant Shift* and *Pitch* modulation (Vibrato) by the *LFO*.

Rate

This knob determines the speed of the modulation. Values to the left of the center positions are synchronized with the sequencer's tempo and include bar values, triplet values and more. Values to the right of the center positions are non-synchronous and displayed in Hertz (cycles per second).

Note: The ability to use synchronous bar values could be used to perform a formant shift every four bars on a one bar percussion part, which is being cycled. Alternately, you could perform the same formant shift on every eighth note triplet within the same part. Either method can generate interesting results, and can lead to new ideas, or a new lease of life on old audio material.

U/V Detection

Please refer to “Analyzing Speech Signals,” from page 169 onwards, for an explanation of the *U/V Detection* principle.



Speech intelligibility is highly dependent on high frequency content, as human hearing is reliant on these upper-end frequencies to determine syllables within words. Bear this fact in mind when using the EVOC 20 TO, and take care with filter frequency settings in the *Synthesis* and *Formant Filter* sections.

To aid intelligibility, it may be worthwhile using equalization to boost particular frequencies in the mid to high frequency range, before processing the signal with the EVOC 20 TO. Please see the “Tips for Better Speech Intelligibility” on page 169 for further information.

Sensitivity

This parameter determines how responsive the U/V detection is. By turning this knob to the right, more of the individual *unvoiced* portions of the input signal are recognized.

When high settings are used, the increased sensitivity to unvoiced signals can lead to the U/V source—determined by the *Mode* parameter—being used on the majority of the input signal, including *voiced* signals. Sonically, this results in a sound that resembles a radio signal which is breaking up and contains a lot of static or noise.

Mode

Here, you select the sound source(s) which can be used to replace the *unvoiced* content of the input signal. Possible settings are *Off*, *Noise*, *Noise + Synth*, or *Blend*.



- *Noise*—uses noise alone for the unvoiced portions of the sound.
- *Noise + Synth*—uses noise and the synthesizer for the unvoiced portions of the sound,
- *Blend*—uses the Analysis signal after it has passed through a highpass filter, for the unvoiced portions of the sound. This filtered analysis signal is then mixed with the EVOC 20 TO output signal. The *Sensitivity* parameter has no effect on this setting.

Level

Level controls the amount of the signal (*Noise*, *Noise + Synth*, or *Blend*) used to replace the *unvoiced* content of the input signal.



Warning: Care should be taken with this control, particularly when a high *Sensitivity* value is used, to avoid internally overloading the EVOC 20 TO.

Output Parameters

Signal



This pull-down menu offers the choice of *Voc(oder)*, *Syn(thesis)* and *Ana(lysis)*. Selection of one of these determines the signal that you wish to send to the EVOC 20 TO's main outputs. To hear the vocoder effect, *Signal* must be set to *Voc*. The other two settings are useful for monitoring purposes.

Level

Level controls the volume of the EVOC 20 TO's output signal.

Stereo Mode



This pull-down menu determines the input/output mode of the *Synthesis* filter bank. Choices are: *m/s*—*mono* input to *stereo* output and *s/s*—*stereo* input to *stereo* output.

Note: The *Stereo Mode* should be set to *m/s* if the signal going into the *Synthesis* filter bank is monophonic or *Synthesis In* is set to *Osc*.

Note: Stereo/stereo (*s/s*) is the preferred setting for stereo *Synthesis* input signals. In this case, the stereo signal is processed by a separate filter bank for the left and right channels. When using the *m/s* mode on stereo *Synthesis* input signals, the stereo signal is first summed to mono before it is passed on to the *Synthesis* filter bank.

Stereo Width

Stereo Width distributes the output signals of the *Synthesis* section's filter bands in the stereo field.

- In the left position, the output of all bands are centered.
- In the center position, the output of all bands ascends from left to right.
- In the right position, the bands are output—alternately—on the left and right channels.

Note: The stereo/stereo mode (*s/s*) uses one filter bank per channel. The positioning of the frequency bands correspond to that described above, but the bands of each filter bank ascend in opposing directions, from left to right.

MIDI Controllers Received

The following tables show the CC numbers used when the following MIDI preference is active: *Options > Settings > MIDI Options > (Version 4.x behavior)*.

Sidechain Analysis	Attack	CC #79
	Release	CC #80
	Freeze	CC #78
Synthesis	Input Mode	CC #65
	Bands	CC #66
	FM Ratio	CC #86
	FM Intensity	CC #87
	Coarse Tune	CC #85
	Fine Tune	CC #105
Pitch Quantize	Strength	CC #90
	Root Scale Key	CC #103
	Root Scale Presets	CC #104
	Max Track	CC #88
	Glide Time	CC #89
Formant Filter	FF Low Freq	CC #67
	FF Hi Freq	CC #68
	Formant Shift	CC #69
	Formant Stretch	CC #74
	FF Resonance	CC #75
	FF Low/Bandpass Select	CC #76
	FF High/Bandpass Select	CC #77
LFO	LFO Rate	CC #70
	LFO Waveform Select	CC #71
	Shift Intensity	CC #72
	Pitch Intensity	CC #73
U/V Detection	Sensitivity	CC # 83
	Mode	CC # 82
	Level	CC # 84
Output	Signal Out Select	CC #106
	Level	CC #107
	Mono/Stereo Select	CC #108
	Stereo Width	CC #81

Root/Scale kyb.	C	CC #91
	C#	CC #92
	D	CC #93
	D#	CC #94
	E	CC #95
	F	CC #96
	F#	CC #97
	G	CC #98
	G#	CC #99
	A	CC #100
	A#	CC #101
	B	CC #102

High Cut/Low Cut

- The Low Cut filter attenuates the frequency range below the selected frequency.
- The High Cut filter attenuates the frequency range above the selected frequency.

High Pass/Low Pass Filter

- The High Pass Filter affects the frequency range below the set frequency. Higher frequencies pass through the filter. You can use the High Pass Filter to completely get rid of the bass range below a selectable frequency.
- The Low Pass Filter affects the frequency range above the selected frequency. Lower frequencies pass through the filter. You can use the Low Pass Filter to completely get rid of the treble range above a selectable frequency.

This chapter describes Logic's delay effects.

This includes the Sample Delay, Tape Delay and Stereo Delay plug-ins.

Sample Delay

This plug-in allows the simple delaying of a channel by single sample values. The stereo version of the plug-in provides separate controls for each channel. This plug-in (when used in conjunction with the phase inversion capabilities of the Gain plug-in) is particularly suited to the correction of run-time problems that may occur with multi-channel microphones.

Every sample (at a frequency of 44.1 kHz) is equivalent to the time taken for a sound wave to travel 7.76 millimeters. Looked at differently: If you delay one channel of a stereo microphone by 13 samples, this will emulate an acoustic (microphone) separation of 10 centimeters.

Tape Delay



The Tape Delay simulates a vintage tape echo device, although with some very useful features that such old devices never offered. The first of these is that its delay settings are variable in musical increments. It is equipped with a highpass and lowpass filter in the feedback circuit, as well as a circuit that simulates tape saturation effects. This plug-in is ideal for the dub delays invented by Jamaican toast masters, and used in many styles of music today.

Switching the *Sync* button on forces the plug-in to use the internal tempo of the sequencer. Tempo information is updated in the plug-in window when you open it, and every time you subsequently execute a mouse operation. The plug-in can even handle tempo changes. The Tempo parameter field serves solely to display the current bpm value—you can't use it to change the tempo of the sequencer.

When you want to create dotted note values, move the *Groove* slider all the way to the right to 75%; for triplets, select the 33.33% setting. Note that all intermediate values are possible. You can view the current delay value in the *Delay* parameter field.

Disengage *Sync* if you would like to adjust the delay time independently of the song tempo (or change the song tempo without changing the delay time). In this mode, the bpm or ms values can be altered freely by clicking in the *Tempo* parameter field, while dragging up or down with the mouse. Note when changing the ms values using the left portion of the Delay parameter field, the ms values will increment in large steps, while using the right portion of the field will increment the ms values in small steps.

As you might expect, the *Feedback* slider determines feedback intensity; in other words, the amount of delayed and filtered signal that is routed back to the input of the Tape Delay. When you set it to the lowest possible value, the Tape Delay generates a single echo. If Feedback is turned all the way up, the echoes are repeated ad infinitum. Keep in mind that the levels of the original signal and its taps (echo repeats) tend to add up, and may cause distortion. This is where the internal tape saturation circuit comes to the rescue—it can be used to ensure that these overdriven signals sound good.

The *Freeze* parameter captures the current delay repeats and sustains them until the *Freeze* parameter is released.

You can shape the sound of the echoes, using the on-board highpass and lowpass filters. Although these filters are fairly flat, they're not located post-output. They are located in the feedback circuit, meaning that the effect achieved by these filters increases in intensity with each repeat. If you're in the mood for an increasingly muddy tone, move the High Cut filter slider towards the left. For ever thinner echoes, move the Low Cut filter slider towards the right.

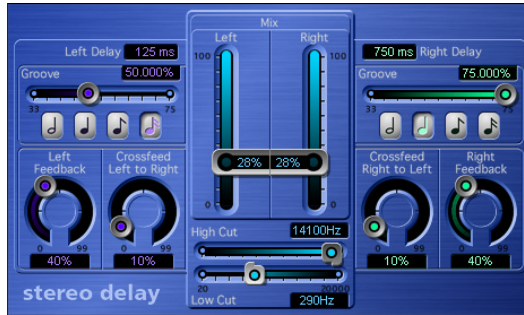
The *Mix* slider determines the balance between the original (dry) signal and effects (wet) signals. If you've inserted the Tape Delay in an individual track, you'll generally find that settings of up to 50% are desirable. If the Tape Delay is patched to the insert of a Bus channel, and you're routing the signals of a track to the plug-in with the *Send* controls, you should set the *Mix* slider to 100%, and leave it there.

If you're unable to hear the effect, even though you've set up a suitable configuration, be sure to check out not only the *Mix* knob, but also the filter settings: Move the *High Cut filter* slider to the far right, and the *Low Cut filter* slider to the far left.

The Tape Delay includes an LFO for delay time modulation. Use it to produce very pleasant and special chorus effects, even on long delays. The LFO produces a triangular wave, with adjustable speed and modulation intensity, that can be evened out with the *Smooth* parameter. This also smoothes the Flutter. *Flutter* simulates the irregularities of tape transport speeds used in analog tape delay units, and is also adjustable in speed and intensity.

There are three further parameters in the Tape Delay's Controls view. The *Dry* and *Wet* sliders can be used to control the original and effect signal amounts individually, independently of the *Mix* parameter. *Distortion Level* can lower the distorted signal (tape saturation) level by up to 20 dB.

Stereo Delay



The Stereo Delay works much like the Tape Delay, which is why we'll skip the general info, and take a closer look at the differences between the two. There is just one Stereo Delay (s/s), hence the stereo input and output. You are free to use the Stereo Delay for monaural tracks or busses, when you want to create independent delays for the two stereo sides. Please bear in mind that if you use this option, the track or bus has two channels from the point of insertion forward. Unlike the Tape Delay, the Stereo Delay does not feature a circuit that replicates tape saturation.

You can set the *Delay* (using Note buttons and Groove sliders), *Feedback*, and *Mix* values separately for the two sides. The *High Cut* and *Low Cut* sliders, however, apply equally to both sides. In addition, the plug-in features a *Crossfeed* knob for each stereo side. It determines the feedback intensity—or the level at which each signal is routed to the opposite stereo side.

There are ten additional parameters in the Stereo Delay's Controls view.

If you would like to adjust the delay time independently of the song tempo, select *ms* in the *Delay Unit* pull-down menu. You can use the *Left Delay* and *Right Delay* sliders just above the Delay Unit pull-down menu to set the delay time in milliseconds. *Left Input* and *Right Input* determine the input signal for the two stereo sides. You can choose between Off, Left, Right, L+R, L–R.

Selecting the *Inv* option in the *Phase Left FB* and *Phase Right FB* pull-down menus allows you to invert the phase of the corresponding channel's feedback signal. The *inv* option is also available in the *Phase L→R FB* and *Phase R→L FB* pull-down menus, where it can be used to transfer the inverted feedback signal of the left/right channel to the right/left channel. The *Tempo Freeze* parameter captures the current delay time and sustains it until the *Freeze* parameter is released.

This chapter introduces Logic's modulation effects.

This includes the Modulation Delay, Chorus, Flanger, Phaser, Ring Modulator, Tremolo, Ensemble, Rotor Cabinet, Scanner Vibrato, and Spreader plug-ins.

Modulation Delay



As its name implies, the Modulation Delay generates effects such as flanging or chorus, based on modulated short delays. It can also be used—without modulation—to create resonator or doubling effects.

The modulation section consists of two LFOs, with variable frequencies (0 to 20 Hz). The balance between these two is determined by the *LFO Mix* slider. Use the *Width* slider to enter the desired modulation width. When the *Width* slider is set to the far right position, delay modulation is switched off completely. The *Vol. Mod.* (Volume Modulation) slider determines the intensity of amplitude modulation (Tremolo). The *Constant Mod.* (*Constant Modulation*) button lets you do just that—ensure that the modulation width remains constant, regardless of the modulation rate. When this feature is switched off, higher modulation frequencies reduce the modulation width. In simple delay circuits, a delay modulation would normally also modulate the pitch of the signal. Use the *Anti Pitch* button to ensure that the pitch of the modulated signal remains constant. This is exactly how high-end chorus and flanger effects work.

Set the basic delay time with the *Flanger-Chorus* knob. Set to the far left position, the Modulation Delay puts on its flanger cap. As you move towards the center position, it thinks it's a chorus. As you move the knob closer to the far right position, you will hear clearly audible delay taps. This latter type of setting is generally used without modulation (Width = 0), for doubling effects.

The *Stereo Phase* knob defines the phase of the modulation between the left and right stereo sides. At 0°, the extreme values of the modulation are achieved simultaneously on both sides, at 180°, the extreme values opposite each other are reached simultaneously.

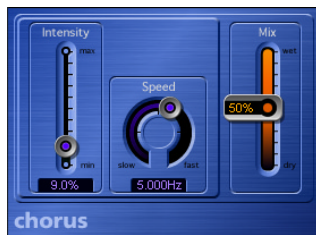
The *Feedback* slider determines the intensity at which the effect's signal feedback is routed to the input. If you're going for radical flanging effects, enter a high *Feedback* value. If simple doubling is what you're after, you won't want any feedback at all. The *Mix* slider determines the balance between dry and wet signals.

The *Controls* view offers six further parameters:

If you set *True Analog* to on, an additional all-pass filter is switched into the signal path. An all-pass filter shifts a signal's phase angle, influencing its stereo image. Use *Analog Left* and *Analog Right* to control the way that the allpass filter affects each of the stereo channels.

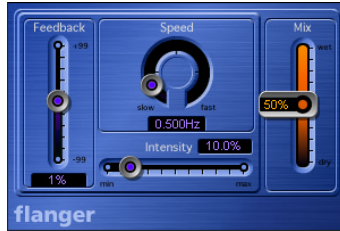
The *Speed LFO 1 R* and *Speed LFO 2 R* sliders allow independent modulation rate settings for LFO1 and 2 (for the right stereo channel). These parameters only work if the *Free* option is chosen in the *Stereo* pull-down menu. With *Stereo* set to *Link*, the modulation rates of the left and right stereo channels are tied to each other, and rates are set by the LFO controls in the Plug-in window. In this situation, the *Speed LFO 1 R* and *Speed LFO 2 R* parameters are non-functional.

Chorus



The Chorus effect is based on a delay line. It's output is mixed with the original, dry signal. While the chorus effects delay time is set internally, you can define its modulation width (*Intensity* parameter) and modulation frequency (*Speed* parameter). The *Mix* slider determines the balance of dry and wet signals.

Flanger



The Flanger works in a similar fashion to the Chorus, but with a shorter delay time, and the output signal being fed back into the input of the delay line. Use the *Intensity* slider to determine the Flanger's modulation width. *Speed* sets the frequency of the modulation. *Feedback* determines the amount of the delayed signal that is routed back into the input. Negative values invert the phase of the routed signal. The *Mix* slider determines the balance of dry and wet signals.

Phaser



The Phaser emulates the effect of analog phaser circuits with four to twelve orders (as in 4th order, 5th order and so on) Use the *Order* slider to set the desired number of orders. As a rule, the more orders a phaser has, the heavier the effect. The 4, 6, 8, 10, and 12 settings put five different phaser algorithms at your fingertips, all of which replicate the analog circuits that they are modeled on, each designed for a specific application.

Note: You are free to select odd numbered settings (5, 7, 9, 11), which, strictly speaking, don't generate actual phasing. The more subtle comb filtering effects produced by odd numbered settings can, however, come in handy on occasion.

The modulation section offers two LFOs, featuring individually variable frequencies, and freely variable mix options (*LFO Mix*). Additionally, the frequency of LFO 1 can be modulated by the level of the input signal. Use the *Envelope Modulation* slider to set the desired modulation intensity. By staking out the limits of the modulation with its highest and lowest values, you can determine the modulation width and range. These high/low limits are controlled by the *Sweep Ceiling* and *Sweep Floor* sliders—you can enter values for them directly in the form of the desired frequency. This value also determines the maximum intensity of the comb filtering created by the phasing effect.

The *Stereo Phase* knob is used to define the phase for the left and right channels of a stereo phaser (s/s). When you're using a monaural phaser, this parameter is, of course, meaningless and can't be set. As the icing on the phasing cake, you can tweak the *Color* slider to add just that to the effect. Technically, the comb filtering effect is amplified via feedback.

There are six additional parameters in the Phaser's Controls view.

The *Mix* slider determines the balance of dry and wet signals. Negative values result in a phase inverted mix of effect and direct signal. The Phaser's built-in envelope follower tracks any volume changes in the input signal, generating a dynamic control signal. This control signal can be used as a modulation source. *Dir.-Env-Mod* sets the desired modulation intensity for the envelope control signal. *Warmth* switches on an additional distortion effect, which allows the creation of warm overdrive effects. *FB Filter* can be used to activate an additional filter section, which processes the feedback signal of the Pitch Shifter. This filter section consists of a highpass and lowpass filter, where cutoff frequency can be set with *LP Cutoff* and *HP Cutoff*.

RingShifter—Ring Modulator/Frequency Shifter



Logic's RingShifter plug-in combines a ring modulator with a frequency shifter effect. These two related effects are based on modulation of the signal amplitude. Both effects were popular during the 1970's, and are currently experiencing something of a renaissance. The ring modulator, for example, was extensively used on jazz rock and fusion records in the early 70's. The frequency shifter was, and still is, found as part of many modular synthesizer systems. Due to the intricate nature of its hardware, the frequency shifter was (and remains) relatively expensive to produce, and was therefore never as widespread as the simpler ring modulator.

Technical Background

The ring modulator modulates the amplitude of the audio input signal using either; the internal oscillator or a second audio signal. The frequency spectrum of the resulting effect signal equals the sum and difference of the frequency content of the two original signals. Its sound is often described as metallic or clangorous.

An elaborate array of allpass filters enables the frequency shifter to separate the sum and difference signals into two separate audio signals—one carries the audio signal with its frequency spectrum shifted up, the other with its spectrum shifted down. The amount of frequency shift is set via the frequency of the internal sine wave oscillator.

Frequency shifting should not be confused with pitch shifting. Pitch shifting transposes the original signal, leaving its harmonic frequency relationship intact. The frequency shifter shifts the frequency content by a fixed amount and, in doing so, alters the frequency relationship of the original harmonics. The resulting sounds range between sweet and spacious phasing effects to strange robotic timbres.

Modes

The four *Mode* buttons determine whether the plug-in either operates as a frequency shifter or as a ring modulator. The frequency shifter offers the *Single* and *Dual* settings. The ring modulator provides the *OSC* and *Side Chain* settings.

- *Single (Frequency Shifter)*: The frequency shifter generates a single shifted effect signal. The position of the large *Frequency* rotary control determines whether the signal is shifted up (positive value) or down (negative value).
- *Dual (Frequency Shifter)*: The frequency shifting process produces one shifted effect signal for each stereo channel—one is shifted up, the other is shifted down. The position of the large *Frequency* rotary control (in relation to the 0 point) determines the shift direction in the left versus the right channel.
- *OSC (Ring Modulator)*: The Ring Modulator uses the internal sine wave oscillator to modulate the input signal.
- *Side Chain (Ring Modulator)*: The Ring Modulator modulates the amplitude of the input signal with the audio signal assigned via the side chain of the plug-in.

Note: The internal sine wave oscillator has no effect in the Side Chain mode, and for this reason, the oscillator frequency controls are not accessible.

The Oscillator

In both the Ring Modulator OSC mode and the Frequency Shifter modes, the internal sine wave oscillator serves as an amplitude modulator of the input signal. The large *Frequency* rotary control is used to set the frequency of the sine oscillator. It can be set between 0 and $\pm 5,000$ Hz in extremely fine increments.

- In the Frequency Shifter modes, this parameter controls the amount of frequency shifting (up and/or down) applied to the input signal.
- In the OSC mode of the ring modulator, this parameter controls the frequency content (timbre) of the resulting effect. This timbre can range from subtle tremolo effects to clangorous metallic sounds.

To optimize adjustment, the scaling of the *Frequency* rotary control can be switched via the *Lin(ear)* and *Exp(ponential)* buttons. The exponential scaling offers extremely small increments around the 0 point, which is useful for programming slow moving phasing and tremolo effects. In the *Lin(ear)* mode, the resolution of the scale is even over the entire control range.

Further to these options, the oscillator frequency can be modulated with an envelope follower and LFO (see later). The oscillator is capable of frequency sweeps through the 0 Hz point. The modulation depth for the envelope follower and LFO is set independently, by using the corresponding bipolar slider.

Delay

The effect signal is routed through a delay, following the oscillator. The *Level* control sets the level of the delay added to the ring modulated or frequency shifted signal.

Note: A *Level* value of 0 passes the effect signal directly to the output (bypass).

The *Time* control sets the delay value from 0 to 2,000 milliseconds. Activation of the *Sync* button synchronizes the delay to your Logic song tempo, in musical note values.

Output

The RingShifter offers a feedback loop, which operates independently of the delay section, by routing the output of the RingShifter back into its input. Feedback gives the RingShifter sound an additional edge, and is useful for a variety of special effects. In combination with a slow oscillator sweep, it produces a rich phasing sound. Comb filtering effects are created using high *Feedback* settings with a short delay time (< 10 ms). Using longer delay times in combination with Feedback creates spiralling and continuously rising and falling frequency shift effects.

The *Stereo Width* rotary control determines the breadth of the effect signal in the stereo field.

Note: *Stereo Width* only affects the effect signal of the RingShifter, not the dry input signal.

The *Dry/Wet* rotary control sets the mix ratio of the dry input signal and the wet effect signal. If required, the *Dry/Wet* mix can also be modulated with an envelope follower and LFO. The modulation depth for the envelope follower and LFO is set independently with their bipolar sliders

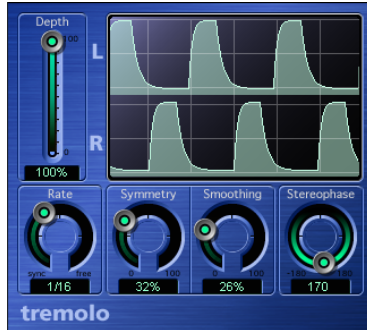
Modulation Sources

The *Oscillator Frequency* and *Dry/Wet* mix ratio parameters can be modulated via the internal Envelope Follower and LFO. The *Oscillator Frequency* even allows modulation through the 0 Hz point, thus changing the oscillation direction.

The Envelope Follower analyzes the amplitude (volume) of the input signal, and uses this to create a continuously changing control signal—a dynamic volume envelope of the input signal. This “control signal” can be used for modulation purposes. The *Power* button turns the Envelope Follower on or off. The *Sensitivity* slider determines how responsive the Envelope Follower is to the input signal. At lower settings, the Envelope Follower will only react to the most dominant signal peaks. At higher settings, the Envelope Follower will track the signal more closely, but may react less dynamically. Try to find a suitable median (compromise) setting for the input signal. The response time of the Envelope Follower is set with the *Attack* slider. *Decay* controls the time it takes the Envelope Follower to return from a higher to a lower value.

The LFO is the second modulation source. It is activated/deactivated via its own *Power* button. The LFO produces continuously cycled control signals. The LFO waveform can be shaped as required via the *Symmetry* and *Smooth* sliders. The LFO waveform display provides visual feedback of the resulting waveform. The *Rate* rotary control sets the cycle speed of the LFO. Press the *Sync* button if you want to synchronize the LFO cycles with the Logic song tempo (using musical note values).

Tremolo



The tremolo effect is a cyclic modulation of the amplitude, resulting in periodic volume changes of. As opposed to the vibrato effect which can be achieved with the Modulation Delay plug-in, the amplitude (not the frequency) is the modulated parameter. You'll recognize this effect from vintage guitar combo amps (where it is sometimes incorrectly referred to as vibrato).

The intensity of modulation is set with *Depth*. *Rate* defines the speed (frequency) of the modulation. If *Symmetry* is set to 50% and *Smoothing* to 0%, the modulation has a rectangular shape. This means that the timing of the full volume signal is equal to that of the low volume signal, and that switching between both states occurs abruptly. You can define the loud/quiet time ratio with *Symmetry*, and make it fade gently in or out with *Smoothing*. *Stereophase* defines whether the modulation takes place in phase or out of phase, when in stereo mode. It can be set to any phase angle. When set to out of phase (-180°) the balance wanders from left to right. When set to 180° , left and right channels are altered in volume simultaneously (in phase).

The graphic display is self-explanatory: All parameters, except modulation speed (*Rate*), are displayed.

Ensemble



The Ensemble is like a pitch shifter on steroids—it consists of eight internal, modulatable pitch shifters. Two standard LFOs and one random LFO enable you to come up with fairly complex pitch modulations, which—much like a natural chorus effect—conjure up the impression of an instrumental or vocal ensemble. The Ensemble’s graphic visually represents the number of voices, and their modulations.

Use the *Voices* slider to determine how many voices (1 to 8) are generated, in addition to the original. Please note that the plug-in’s appetite for computer resources increases proportionally to the number of voices: When you activate eight voices, the Ensemble requires roughly eight times the CPU power of a pitch shifter.

The two conventional LFOs and the random LFO (which generates random modulations), each feature a *Rate* knob that controls frequency, and an *Intensity* slider to determine the modulation width.

The *Phase* knob controls the phase relationship between the modulations of the individual voices. The value that you select here depends on the number of voices, which is why it is indicated in percentages rather than degrees. The value 100 (or –100) is equal to the greatest possible distance between the modulation phase of all voices. Here, the voices are distributed an equal distance apart over the full 360°.

The *Stereo Base* slider serves to distribute the voices across the stereo field. When you set a value of 100%, the stereo base is expanded artificially. Please note that monaural compatibility may suffer.

In addition to the familiar *Mix* slider that determines the balance of dry and wet signals, the Ensemble also features an *Effect Volume* knob. This lets you determine the level of the effects signal separately. This feature allows you to compensate for changes in volume caused by manipulating the *Voices* parameter.

Rotor Cabinet

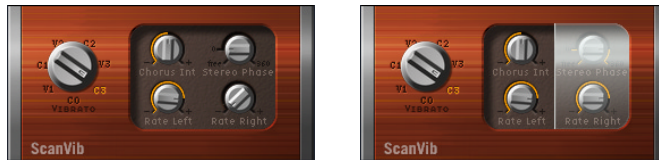


The Rotor Cabinet plug-in is based on the EVB3's Rotor Cabinet effect section. A detailed description of its parameters can be found in the EVB3's "Rotor Cabinet" section on page 474.

Please note: There is no *Speed Control* parameter on the Rotor Cabinet plug-in. You can switch rotor speeds manually.

Scanner Vibrato

In its mono-version, the stereo parameters of the scanner vibrato are hidden behind a transparent cover (right):



The Scanner Vibrato plug-in is based on the EVB3's Scanner Vibrato effect section. More information about its parameters can be found in the EVB3's "Scanner Vibrato" section on page 462.

The stereo version of the Scanner Vibrato effect features two additional parameters: *Stereo Phase* and *Rate Right*.

If *Stereo Phase* is set to *free*, the modulation speed can be set independently for the left (*Rate Left*) and right (*Rate Right*) channels. This allows for quite wild effects, as left and right modulation are not synchronized with each other.

If *Stereo Phase* is set to 0 to 360 degrees, the modulation speed for both the left and right channels is set with *Rate Left*. *Stereo Phase* determines the phase relationship between left and right channel modulations, thus enabling synchronized stereo effects. *Rate Right* has no function when in this mode.

Spreader

The Spreader plug-in widens the stereo spectrum with an effect that is quite similar to the Chorus effect. The frequency range of the original signal is periodically shifted in a non-linear way. In comparison to the Stereo Spread effect, the perceived pitch changes.

Use the *LFO Intensity* parameter to set the modulation width of the Spreader. *LFO Speed* controls the modulation frequency. *Channel Delay* determines the delay time in Samples. *Mix* sets the balance of dry and wet signals.

This chapter describes Logic's reverb effects.

This includes AVerb, SilverVerb, GoldVerb, PlatinumVerb, Enverb, and Space Designer. Space Designer is Logic's only convolution reverb, and is described separately in the "Convolution Reverb: Space Designer" chapter, from page 117.

AVerb



Although the AVerb is based on a simple reverb algorithm, it delivers remarkably good results.

The actual reverb algorithm is controlled by just four parameters:

- As its name implies, *Reflectivity* defines how reflective the imaginary walls, ceiling, and floor will be.
- *Room Size* challenges your architectural skills—use it to define the dimensions of simulated rooms.
- *Density/Time* determines both the density and duration of the reverb.
- *Pre Delay* determines the delay between the original signal and the reverb tail.

The *Mix* parameter determines the balance between the effected (wet) and direct (dry) signals.

Where high Pre Delay settings tend to generate something similar to an echo, low values often muddy the original signal. Ideally, you should go for as high a setting as possible before the plug-in begins generating something that sounds like a tap delay. With appropriate Pre Delay settings, you can apply relatively generous amounts of reverb to percussive parts, while retaining definition on the attack portions of the sounds.

SilverVerb



The SilverVerb algorithm is controlled by just three parameters: As its name implies, *Reflectivity* defines how reflective the imaginary walls, ceiling, and floor will be. *Room Size* challenges your architectural skills—use it to define the dimensions of simulated rooms. The graphic display visually represents these parameter settings.

PreDelay determines the delay between the original signal and the reverb tail.

Whereas high PreDelay settings tend to generate something similar to an echo, low values often muddy the original signal. Ideally, you should go for as high a setting as possible before the plug-in begins generating something that sounds like a delay tap. With appropriate PreDelay settings, you can apply relatively generous amounts of reverb to percussive parts, while allowing the attacks to remain intelligible.

Low Cut and *High Cut* let you filter bass and treble frequencies out of the reverb tail.

In most cases this will open up your mix. The reason for this is that a long reverb with a great deal of bottom end generally makes for a flabby mix, and high frequencies in the reverb usually sound somewhat unpleasant, hamper speech intelligibility, or mask the overtones of the original signals.

There are four further parameters that are available in the Extra Controls view.

Density/Time determines both the density and duration of the reverb. Small value settings tend to generate something similar to an echo. High values result in a reverb-like effect.

The *Modulation Rate*, *Modulation Int* and *Modulation Phase* parameters control an additional modulation delay. It consists of two LFOs with variable frequencies (set with *Modulation Rate*). The desired modulation width is set with the *Modulation Int* slider. When this slider is set to the far right position, delay modulation is switched off completely. The *Modulation Phase* knob defines the phase of the modulation between the left and right stereo sides. At 0°, the extreme values of the modulation are achieved simultaneously on both sides, at 180°, the extreme values opposite each other are reached simultaneously.

GoldVerb



The GoldVerb consists of two sections: *Early Reflections* and *Reverb* (diffuse reverberations). The balance between these two sections is controlled by the *Balance ER/Reverb* slider, located above the graphic. When you set this *Balance* slider to either of its extreme positions, the unused section is deactivated, maximizing performance.

Early Reflections

This section emulates the original signal's first reflections as they bounce off the walls, ceiling, and floor of a natural room. These early reflections are essential to how we perceive a room. All information about the size and shape of a room capable of being discerned by the human ear is contained in these early reflections.

Predelay

Predelay is the amount of time that elapses between the original signal, and the arrival of the early reflections. In any given room size and shape, *Predelay* determines the distance between the listener and the walls, ceiling, and floor. When used with artificially generated reverb, it has proven advantageous to allow this parameter to be manipulated separately from, and over a greater range than, what is considered natural for *Predelay*. In practice, too short a *Predelay* tends to make it difficult to pinpoint the position of the signal. It can also color the sound of the original signal. On the other hand, too long a *Predelay* can be perceived as an unnatural echo. It can also divorce the original signal from its early reflections, which leaves an audible gap. The ideal *Predelay* setting depends on the properties or, more accurately, the envelope of the original signal. Percussive signals generally require shorter *Predelays* than signals where the attack fades in gradually. A good practice is to use the longest *Predelay* possible before undesirable side effects, such as an audible echo, begin materializing.

Room Shape

Use this slider to define the geometric form of the room. The numeric value (3 to 7) represents the number of corners it has.

Room Size

Unsurprisingly, *Room Size* determines the dimensions of the room. The numeric value indicates the length of its walls—the distance between two corners.

Stereo Base

The *Stereo Base* parameter enables you to define the distance between the two virtual microphones that you are using to audition the simulated room. Spacing the microphones slightly further apart than the distance between two human ears generally delivers the best results. Of course, more realistic results can be obtained if you choose to use the distance between two ears located on opposite sides of the same head.

Reverb

This section generates diffuse reverberation.

Initial Delay

This is the delay between the original signal and the diffuse reverb tail. If you're going for a natural-sounding, harmonic reverb, the transition between the early reflections and the reverb tail should be as smooth and seamless as possible. Basically, what we said about the *Predelay* holds true for this parameter:

Set the *Initial Delay* so that it is as long as possible without a perceptible gap between the early reflections and the reverb tail.

Density

This parameter controls the density of the diffuse reverb. Ordinarily, you want the signal to be as dense as possible. However, less Density means the plug-in eats up less computing power. Moreover, in rare instances, too great a Density can color the sound, which you can fix simply by reducing the *Density* knob value. Conversely, if you select a Density value that is too low, the reverb tail will sound grainy.

Diffusion (Controls View)

Diffusion sets the diffusion of the reverb tail. Sometimes, the terms “diffusion” and “density” are confused. The density is the average number of reflections in a given period of time. The diffusion is the amount of irregularity of the density. High values of diffusion represent a regular density, with few alterations in level, times, and panorama position. At low *diffusion* values, the reflection’s density becomes more irregular and grainy. The stereo spectrum changes, too.

Reverbtime

Reverbtime is commonly considered as the amount of time it takes for the level of a reverb signal to drop by 60 dB. This is why the reverb time is often indicated as RT60. Most natural rooms have a reverb time somewhere in the range of one to three seconds, a value which absorbent surfaces and furniture reduces. Large empty halls or churches have reverb times of up to eight seconds, some cavernous or cathedral-like venues even beyond that.

High Cut

Uneven or absorbent surfaces (wallpaper, wood paneling, carpets, and so on) tend to reflect lower frequencies better than higher frequencies. The *High Cut* filter replicates this effect. If you set the *High Cut* filter so that it is wide open, the reverb will sound as if it is reflecting off stone or glass.

Spread

This parameter controls the stereo image of the reverb. At 0%, the plug-in generates a monaural reverb, at 100%, the stereo base is artificially expanded—which, of course, makes the reverb sound monumental, but collapses in monaural playback.

PlatinumVerb



The difference between the PlatinumVerb and the GoldVerb is the former's enhanced Reverb section. The *Early Reflections* sections of the two plug-ins are identical. For more information, please read the "GoldVerb" section, on page 111. We'll focus on the additional features offered by the PlatinumVerb in this section.

The *Reverb* section of the PlatinumVerb is based on a genuine dual-band concept. This is to say that the on-board frequency crossover splits the incoming signal into two bands, which are then processed with reverb in two separate modules.

Parameters of the PlatinumVerb

Crossover

This is the frequency that the two frequency bands are split at, for separate processing.

Low Ratio

This parameter determines the reverb time of the bass band. The *Reverbtime* parameter applies to the high band. At 100%, the reverb times for the two bands are identical. At lower values, the reverb time of the frequencies below the crossover frequency is shorter. At values greater than 100%, the reverb time for low frequencies is longer.

Both of these phenomena occur in nature. In most mixes, a shorter reverb time for bass frequencies is preferable. As an example, if you're using the PlatinumVerb to put reverb on a drumloop featuring kick and snare drums, a short reverb time for the kick drum allows you to set a substantially higher wet signal.

Low Level

This knob determines the level of the bass reverb. At 0 dB, the volume of the two bands is equal. The bass reverb level can be boosted by up to 12 dB and attenuated by up to 100 dB.

In the vast majority of mixes, your best bet is to set a lower level for the low frequency reverb signal. This enables you to turn up the level of the bass instrument—making it sound punchier. This also helps to counter bottom-end masking effects.

The Controls view offers four additional parameters.

ER Scale allows you to scale the early reflections along the time axis, enabling the Room Shape, Room Size and Stereo Base parameters to be influenced simultaneously. *Dry* and *Wet* can be used to control the amounts of the original and effect signal individually, and independently of the Mix parameter. The *Diffusion* slider is also available in the GoldVerb plug-in. A detailed description of its function can be found on page 113.

EnVerb



Logic's EnVerb is based on a rather unusual and innovative reverb algorithm. It has a unique feature—you can adjust the envelope of the diffuse reverb tail freely. This provides options that far exceed those of a conventional gated reverb.

The EnVerb algorithm requires a reasonable amount of computing power.

Time Parameters

With a concept as sophisticated as that of the EnVerb, you can well imagine that a single parameter for reverb time just won't do the trick.

Original Delay

This parameter enables you to delay the original signal. This delay is only noticeable when the *Mix* parameter is set to a value other than 100%. The starting point of the diffuse reverb tail is not influenced in any way.

A delayed original signal is particularly handy when you want to generate reverse reverb: Set all envelope parameters to 0 with the exception of *Attack* and *Original Delay*, which you should set to approximately the same value that you want to pre-delay the given region or track.

Predelay

This is the delay between the (undelayed) original signal, and the starting point of the reverb attack phase.

Attack

This is the amount of time it takes for the reverb to climb to its peak level.

Decay

This is the amount of time it takes for the level of the reverb to drop from its peak to the sustain level.

Sustain

This is the level of the reverb that remains constant throughout the sustain phase.

Hold

This is the duration (time) of the sustain phase.

Release

This is the amount of time that the reverb takes to fade out completely, after it has run through its sustain phase.

Sound Parameters

The following parameters shape the sound of the reverb. (For more information on these parameters, check out the in-depth descriptions of the GoldVerb or PlatinumVerb).

Density

Here, you can set the reverb's density. Higher values generally sound better.

Spread

Here, you can set the stereo base of the reverb.

High Cut

Here, you can set the high-frequency attenuation for the reverb.

Crossover

Here, you can set the crossover frequency for the Low Level parameter.

Low Level

Relative reverb level of frequencies below the crossover frequency. Although you are free to turn the level of these frequencies up, in most cases, you'll get better-sounding results when you set negative values for this parameter.

This chapter introduces you to Logic's Space Designer Reverb effect.

Space Designer is a convolution reverb plug-in. Reverberation is generated by means of a real-time convolution process, using any loaded impulse response (IR) recording (reverb sample). Put another way, an IR recording of an actual real-world room, a vintage plate or spring reverb unit, for example. The result is an exceptionally realistic reverb/room sound.

An impulse response can be viewed as the total echoes (reflections) in a given room, following an initial signal spike. The waveform display of an IR is also known as a reflectogram. The IR file is simply an audio file.

Space Designer is able to modify existing impulse responses, providing unprecedented control over dynamics, timbre, and length via a comprehensive set of parameters such as envelopes and filters. The graphically-editable envelopes are optimized for reverb tasks, allowing the creation of ultra-smooth envelope curve shapes.

In addition, Space Designer includes an innovative on-board IR synthesis facility that incorporates the editing flexibility of the envelopes and filters. Reverbs based on synthesized IRs provide an incredibly rich and smooth sound.

The sound of the Space Designer reverb never exhibits the shattering or resonances typical of conventional reverb units—just smooth and rich reverb tails—even when using extreme settings on the most complex of input material.

Using Space Designer

Inserting Space Designer

As per all effect plug-ins in Logic, Space Designer can be inserted into any Audio Object's insert slot. In doing so, you should take the following information into consideration:

- Set the *Direct Output* parameter to a value of 0 (mute), if Space Designer is inserted into a bus channel. This happens by default in Logic, but should you change it (accidentally or by design), and things don't sound right, this is the first thing to look at.
- When inserted on an Audio Instrument, Audio channel and so on, the *Direct Output* level is set to 100%.
- The default *Reverb Level* is set at -10 dB , regardless of the channel type that Space Designer is inserted into.
- Note that the Project Manager sees all IRs as simple audio files.

Automation in Logic

As with all Logic plug-ins, a Space Designer effect used on a Logic mixer channel is fully programmable.

Important: Space Designer can not be fully automated as per most other Logic plug-ins. This is because Space Designer needs to reload the Impulse Response (and recalculate the convolution) before audio can be routed through it.

You can, however, record, edit, and play back any movement of the following Space Designer parameters via Logic's track-based automation system.

- *Stereo Crossfeed*
- *Direct Output*
- *Reverb Output*

Operation of Space Designer's Parameters

Space Designer's envelopes can be adjusted graphically, by click-holding on the nodes (shown as hollow or filled dots) or envelope curves, and dragging to the desired position in the Envelope window of the Space Designer's GUI. The envelopes can also be adjusted via click-hold and vertically dragging on the various parameter values in the long horizontal panel below the graphical Envelope window.

There are also some clickable key parameter positions in the GUI—with the slider or dial jumping to that position immediately. Clickable parameter positions are:

- The three base positions of the reverb input slider—*Stereo*, *Mono*, *Xstereo*.
- The zero mark of the Low Shelving EQ Gain control.
- The *HP*, *BP*, *12 dB* and *6 dB* filter modes.

Note that parameter changes occur after the release of the mouse button, and after an additional grace period has elapsed, which is indicated by a blue bar. The calculation itself requires a certain amount of processing time—depending on the speed of your Macintosh CPU(s). During this calculation time, no other parameters can be adjusted. A (red) progress bar below the *Length* parameter panel will advise you of the calculation status.

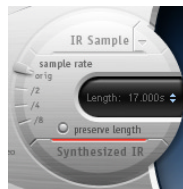
- Prior to the appearance of the progress bar, a blue grace period bar will appear for some parameters. This represents the time before any calculation starts. The bar counts down from right to left, during which time you can adjust the parameter.

Space Designer's Parameters

The following chapter discusses the parameters of the Space Designer.



Impulse Response Parameters



The Impulse Response section allows you to select or create Impulse Responses.

The Space Designer offers two modes of operation—*IR Sample* and *Synthesized IR*.

IR Sample

When the *IR Sample* button is initially clicked, an operating system File Selector dialog will be launched, allowing you to select the desired Impulse Response file from a folder on your hard disks or CD.

If you have already loaded an IR file, this button is simply used to switch back from *Synthesized IR* to *IR Sample mode*. To change the Impulse Response click the downward pointing arrow to the right of the button.

The default IR folder is found in the following path: *Local:/Library/Application Support/Apple/Impulse Responses*

Simply browse to the desired file, and click OK. Any mono, stereo or split stereo AIFF, SDII or WAV file can be used.



The downwards pointing arrow to the right of the button accesses the following menu functions:

- *Load IR*—loads an IR sample without changing the envelopes.
- *Load IR & Init*—loads an IR sample and initializes the envelopes.

The name of the loaded IR file and its length are displayed in the Envelope window.

Synthesized IR

Clicking on this button will activate the Impulse Response synthesizer. A synthesized IR—determined by the values of the *Length*, *Envelope*, *Filter*, and *Spread* parameters—is generated.



Subsequent button pushes will *randomly* generate new IRs with slightly different reflection patterns. The current IR state will always be saved with a setting, allowing for an accurate reproduction of the reverberation sound when next loaded.

You may freely switch between a loaded IR sample and a synthesized IR without losing the settings of the other.

Sample Rate

This parameter determines the sample rate of the Impulse Response. By default the current Logic song sample rate is used by Space Designer as well (if the Logic song is running at 96 kHz, Space Designer uses the same rate). When loading an Impulse Response, Space Designer automatically converts the sample rate of the IR to match the current Logic song sample rate—should it be necessary. As an example, this allows you to load a 44.1 kHz Impulse Response into a Logic song running at 96 kHz, and vice versa.



Three other options are also available. These are half-divisions of the preceding value—one-half, one-quarter, one-eighth. As examples:

- If the top sample rate is 96 kHz, the options will be 48 kHz, 24 kHz and 12 kHz.
- If 44.1 kHz is the selected sample rate, the options will be 22.05 kHz, 11.025 kHz and 5512 Hz.

Changing the *Sample Rate* increases (up) or reduces (down) the frequency response (and length, see below) of the Impulse Response, and therefore the overall sound quality of the Space Designer.

By selecting half the Sample Rate, the IR becomes twice as long. The highest frequency that can be reverberated will be halved. This facility results in a behavior that is much like doubling every dimension of a virtual room (multiplying a room's volume by eight). This can sound great!

Another benefit is that the process requires significantly less processing power, making half Sample Rate settings the ideal solution for wide, open spaces. Check it out!

Don't worry too much if the maximum bandwidth of the reverb tail is reduced to 11.025 kHz when you select a sample rate of 22.05 kHz (half of 44.1 kHz). Natural room surfaces (concrete and tiles, excluded) barely reflect such high frequencies.

The lower sample rates can also be used for interesting tempo/pitch and retro-digital sounding effects.

Note: If running Space Designer in a song at 96 kHz (utilizing an Impulse Response originally recorded at 44.1 kHz), you may want to reduce the IR Sample Rate to 1/2. To do so, use the *Sample Rate* Slider in Space Designer. Make sure the *Preserve Length* function is enabled. This cuts CPU power consumption by about 50%, without compromising reverb quality. There is no loss in reverb quality, because the Impulse Response—originally recorded at 44.1 kHz—will not benefit from the higher song sample rate of 96 kHz.

Note: Similar adjustments can be made while running in Synthesized IR mode. Most typical reverb sounds don't feature an excessive amount of high frequency content. If you were running at 96 kHz, you would need to make use of some deep lowpass filtering to obtain the mellow frequency response characteristics of many reverb sounds. As a different approach, you are better served to first reduce the high frequencies by 1/2 or even 1/4 using the *Sample Rate* slider in Space Designer, and then apply the lowpass filter. This conserves a considerable amount of CPU power.

Length

This parameter determines the length of the Impulse Response (sampled or synthesized).



All envelopes are calculated as a percentage of the overall *Length* automatically, which means that if this parameter is altered, your envelope curves will stretch or shrink to fit, saving you time and effort.

Note that the *Length* parameter value can not exceed the actual length of an IR *sample*.

Also note that longer IRs (sampled or synthesized) place a higher strain on the CPU.

Preserve Length

Activating this button preserves the length of the Impulse Response when the *Sample Rate* is changed. If the *Sample Rate* is halved, the Impulse Response will be doubled in length. To avoid this, make use of the *Preserve Length* function. Having said this, feel free to manipulate these two parameters as you see fit, as it can lead to a number of interesting results. In any case a lower sample rate will save CPU power.



Global Parameters

Input (Crossfeed)

This parameter allows a stereo input signal to be:



- processed on both channels, retaining the stereo balance of the original signal—top of slider,
- to be processed in mono—middle of slider,
- to be inverted, with processing for the right channel occurring on the left and vice versa—bottom of slider.
- a mixture of stereo to mono cross feeds (in-between positions)

Note: This slider is not available when Space Designer is used as a mono plug-in. When Space Designer is inserted as a mono to stereo plug-in this parameter does not have a function.

Direct Output

This parameter sets the direct signal output level—the level of the non-effected (dry) signal.

Note: Set this to a value of 0 (mute) if the Space Designer is inserted in a bus channel, or when using modelling IRs such as speaker simulations.



Reverb Output

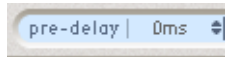
This parameter sets the output level of the effected (wet) signal.

Rev Vol Compensation



Rev Vol Compensation (Reverb Volume Compensation) attempts to match the perceived (not actual) volume differences of Impulse Response files. It is set to *on* by default, and should generally be left in this mode, although you may find that it isn't successful with all types of Impulse Responses. In such situations, switch it off and adjust the *Input* and *Output* levels accordingly.

Pre-Delay



Pre-delay is the amount of time that elapses between the original signal, and the arrival of the early reflections. Put another way, it represents the time the effected signal is delayed in relation to the unprocessed direct signal.

In any given room size and shape, pre-delay determines the distance between the listener and the walls, ceiling, and floor. When used with artificially generated reverb, it has proven advantageous to allow this parameter to be manipulated separately from, and over a greater range than what is considered natural for pre-delay. In practice, too short a Pre-Delay tends to make it difficult to pinpoint the position of the signal. It can also color the sound of the original signal.

On the other hand, too long a Pre-Delay can be perceived as an unnatural echo. It can also divorce the original signal from its early reflections, leaving an audible gap between the signals. The ideal Pre-Delay setting depends on the properties or, more accurately, the envelope of the original signal. Percussive signals generally require shorter Pre-Delays than signals where the attack fades in gradually. A good rule of thumb is to use the longest Pre-Delay possible before undesirable side effects, such as an audible echo, begin materializing. More information on the general principles of reverberation can be found in "So, Just What Is Reverberation?" on page 139.

IR Start

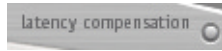


This parameter enables you to shift the playback point into the IR, which will effectively cut off the beginning of the IR.

The IR Start parameter can for example be used to eliminate any peaks at the beginning of the IR sample. It also offers a number of creative options, such as its use when combined with the *Reverse* function.

Note: The *IR Start* parameter is not available in Synthesized IR mode. In the Synthesized IR mode this parameter is not required as, by design, the *Length* parameter provides identical functionality.

Latency Compensation



When activated, this parameter delays the direct signal (in the Output section of the plug-in) to match the processing delay of the effect signal—this compensation occurs *within* the plug-in.

Processing latency is 128 samples at 44.1 kHz, and is *higher* at lower sample rates.

Envelope Window Parameters



The Envelope window of the Space Designer facilitates a new envelope design based on Bezier curves, with two curve segments (attack and decay) being used to form a complete envelope. Note the transparent view of each of the envelope curves. Also note the small IR Overview. It helps to orientate yourself when zoom is active.

The parameters discussed below are global and will affect the currently selected envelope, dependent on the envelope mode—*Volume*, *Filter* or *Density*.

Please note that use of the word mode here is designed to simplify your understanding. Obviously, the Space Designer displays, and uses, up to three discrete envelopes simultaneously.

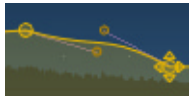
Envelope Window Node Handling

Before we get to the buttons, we'd like to briefly touch on the Envelope window of the Space Designer GUI.

When first launched, a default synthesized Impulse Response and set of Envelopes are automatically created.

You will see a few nodes placed around and within the Volume Envelope that is displayed. These nodes are indicators of several parameter positions/values.

If you look carefully, you will see that some nodes are actually a little larger than others, that some are filled and others are hollow. To explain:



- The filled parameters are active/selected. Hollow ones are not selected/active.
- The small nodes attached to a line are used for finer adjustments to Envelope curves. We refer to these as the Curve Form nodes. More information on their use can be found in the “The Volume Attack and Decay Curve Form Nodes” section, on page 130.
- The large nodes are value indicators of the parameters that appear in the horizontal box below—*Init Level*, *Attack Time*, *Decay Time*, and so on.

If you click-hold on any numerical value, say the *Init Level*, and drag your mouse up/down, the corresponding node will move in the Envelope window. Try this with each numerical parameter to establish which node is which.

Now, move your mouse cursor over the edge of the node that controls the *Init Level*, and you'll see a pair of arrows. If you move your mouse cursor into the *center* of the node, the arrows (and node) are filled. The arrows simply indicate the possible directions that the node can be moved.

Experiment with each node/parameter in each of the Envelope window modes (see just below) to get a hang of this. You'll find that it's very intuitive to use.

Reset

Here you can reset the currently selected envelope—*Volume*, *Filter* or *Density*—to default values.



All

Here you can resets all envelopes to default values.

Envelope Mode Buttons



Clicking on these buttons will switch to the selected envelope mode. The Envelope window display will adjust accordingly, with the selected envelope being topped in the superimposed display. The other envelope curves are shown as transparencies in the background.

- The Volume Envelope is shown in red.
- The Filter Envelope is shown in yellow (the Filter needs to be switched on).
- The Density Envelope is shown in light blue (*Synthesized IR* only).

Note: You can also switch between envelope modes via the pull-down menu to the left of the horizontal parameter panel below the Envelope window.

Please note that use of the word mode here is designed to simplify your understanding. Obviously, the Space Designer displays, and uses, up to three discrete envelopes simultaneously.

Reverse

A single click on this button reverses the Impulse Response together with its envelopes, allowing easy creation of reversed reverb effects, and also some unusual sound processing options. A second click undoes the reverse.



When you reverse the Impulse Response, you are effectively using the tail rather than the front end of the sample. As such, you may need to use lower or even negative *Pre-Delay* values when reversing.

Zoom to Fit/A and D Buttons

The parameter name effectively describes what it does. It zooms in on the Impulse Response waveform to make use of the entire Envelope window, in doing so the display will follow any envelope length changes.



Zoom is directly tied to the current envelope mode, so the Zoom to Fit appearance will be different dependent on the currently active envelope viewing mode.

The *A* and *D* buttons alongside the Zoom to Fit parameter are for the Attack and Decay portions of the (currently selected) envelope. The *A* and *D* buttons are only available to the Volume and Filter Envelopes. Simply click on the appropriate button(s) to activate the desired viewing mode.

The small Overview display indicates which portion of the IR file is currently visible in the Envelope window.



Uncheck all buttons to return to the standard, non-zoomed view.

Zoom to Fit will only result in a visible change when the selected envelope is actually smaller than the overall Envelope window display.

Volume Envelope



Init Level

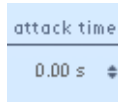
This parameter sets the initial volume level (the start level) of the Impulse Response/ Reverb attack phase. It is expressed as a percentage of the full scale volume of the Impulse Response file. The attack phase is (generally) the loudest point of the Impulse Response.



The *Init Level* should be set to 100% to ensure maximum volume for the early reflections, but obviously use your ears and level meters to ensure that signal levels and audio quality are optimal.

Adjusting this up or down can be done graphically in the Envelope window (the large node on the left), or by click-holding and dragging the mouse cursor up/down on the numerical entry.

Attack Time



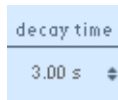
This parameter alters the level of the Impulse Response attack phase over time. It is expressed in second values, with the maximum possible *Attack Time* mirroring the value set by the *Length* parameter (see “Length” on page 122).

The *Attack Time* parameter determines the length of time before the decay phase of the Volume Envelope begins.

Adjusting this left or right can be done graphically in the Envelope window (the large node at the bottom—on the center line), or by click-holding and dragging the mouse cursor up/down on the numerical entry.

Note that the combined total of the *Volume Attack* and *Decay Time* parameters is equal to the total length of the Impulse Response file (determined by the *Length* parameter, see “Length” on page 122), unless the Decay is shortened.

Decay Time

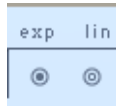


This parameter alters the level of the Impulse Response decay phase over time. It is expressed in second values, with the maximum possible *Decay Time* mirroring the value set by the *Length* parameter (see “Length” on page 122).

Adjusting this left or right can be done graphically in the Envelope window (the large node at the bottom right), or by click-holding and dragging the mouse cursor up/down on the numerical entry.

Volume Decay Mode

You have two options—*Exp* (Exponential) and *Lin* (Linear).



The output of the envelope generator is shaped by an exponential function during the decay phase. If the decay phase is set as a straight line, the result is an exponential function that describes a natural decay.

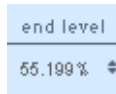
If set *Lin*, no shaping is performed.

An *Exp* curve will result in a much more natural sounding reverb tail.

Click on the desired button to activate the required mode.

End Level

This parameter is used to set the end volume level. It is expressed as a percentage of the overall Volume Envelope—the default *End Level* value is *0.000%*. At this value, the reverb tail cuts off abruptly, which is great for gated reverb effects.



Adjusting this up or down can be done graphically in the Envelope window (the large node at the bottom right), or by click-holding and dragging the mouse cursor up/down on the numerical entry.

The Volume Attack and Decay Curve Form Nodes

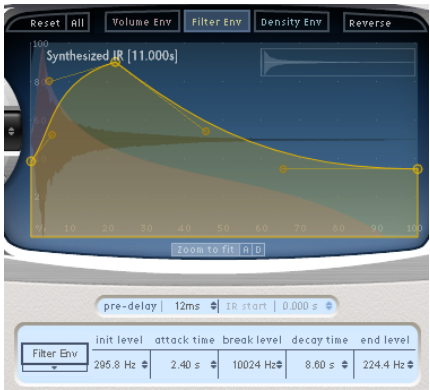
On the curve displayed in the Envelope window, you will find two small nodes in the attack portion of the overall Volume Envelope curve. A vertical line separates the attack and decay portions of the envelope. There are also two small nodes in the decay portion of the envelope—to the right of the vertical line. These are the *Volume Attack/Decay Curve Form node* parameters.



The Curve Form nodes are tied to the envelope curve itself, so you can view them as envelope handles if you wish. Moving the nodes vertically/horizontally will change the shape of the envelope curve. To change the Curve Form node positions, simply click-hold and drag them.

The curve shape can also be changed by clicking and dragging on the envelope curve directly.

Filter Envelope

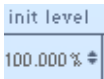


The entire Filter Envelope graphic, including nodes plus the Filter Envelope parameters below. The central large node indicates/controls the Attack endpoint (and Decay startpoint) and Break Level parameters simultaneously. The large node on the right-hand edge controls the Decay endpoint and End Level parameters simultaneously. Also note the smaller nodes in the attack and decay portions of the envelope. These are the *Curve Form node* parameters that are used to create curves.

The *Filter On* switch must be active, or the Filter Envelope controls discussed below will have no effect. The Filter Envelope only is available if the Filter is switched on.

Init Level

This parameter sets the initial cutoff frequency of the Filter Envelope in Hertz.

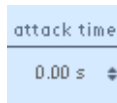


Obviously, you should use your ears and level meters to ensure that the filter (and signal) levels, and desired tonal quality, are appropriate.

Adjusting this up or down can be done graphically in the Envelope window (the large node on the left), or by click-holding and dragging the mouse cursor up/down on the numerical entry.

Attack Time

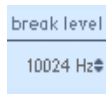
This parameter determines the time required to reach the *Break Level* (see below) value. You may use the *Filter Attack Curve Form node* parameters to alter the shape of the Filter Attack curve.



The combined total of the *Filter Attack* and *Decay Time* parameters is equal to the total length of the (synthesized or sampled) Impulse Response (determined by the *Length* parameter, see “Length” on page 122), unless the Decay time is reduced.

Break Level

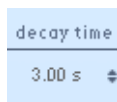
This parameter determines the maximum filter cutoff frequency, the envelope reaches. It also acts as the break point between the attack and decay phases of the overall filter envelope. In other words, when this level has been reached after the Attack phase, the Decay phase will begin.



Note: You can create interesting filter sweeps by setting the *Break Level* to a value lower than that of the *Init Level*.

Decay Time

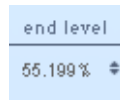
Determines the time required (after the *Break Level* point) to reach the *End Level* value. You may use the *Filter Decay Curve Form node* parameters to alter the shape of the Filter Decay curve.



The combined total of the *Filter Attack* and *Decay Time* parameters is equal to the total length of the Impulse Response file (determined by the *Length* parameter, see “Length” on page 122), unless the Decay time is reduced.

End Level

This parameter is used to set the filter end cutoff frequency. It is expressed as a percentage of the overall Filter envelope scale. If set to a value of 0, the filter stays “open” for the remainder of the reverb signal (provided that the filter is configured as “Low Pass” 6 dB or 12 dB).

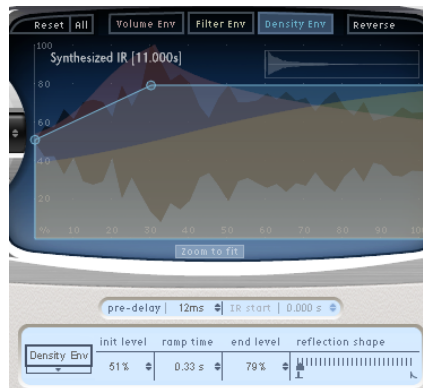


The Filter Attack and Decay Curve Form Nodes

On the Attack curve displayed in the Envelope window, you will see two nodes in the attack portion of the overall Filter envelope curve. The *Break Level* (a larger node) separates the attack and decay portions of the envelope. There are also two nodes in the decay portion of the envelope—to the right of the *Break Level* node. These are the *Filter Attack/Decay Curve Form node* parameters.

Their use is identical to that of the Volume Envelope Curve Form nodes, but obviously they are used to control the Filter Envelope. Please see the “The Volume Attack and Decay Curve Form Nodes” section, on page 130, for further information.

Density Envelope



When synthesizing artificial Impulse Responses, Space Designer offers a facility that approximates the sound of early reflection clusters as part of the overall reverb sound. This is achieved through use of a ramp envelope to dynamically change the density of the early reflection simulation.

The Density parameters can be used in conjunction with the Filter to fine-tune the timbre of your reverbs.

Please note that the Density Envelope is only available when in the Synthesized IR mode.

Init Level

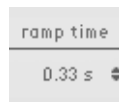
This parameter controls the density (the average number of reflections in a given period of time) of the diffuse reverb. Both the *Initial* and *End Density* (*Init* and *End Level*) can be controlled over time through use of the Density Envelope. Lowering the density levels will result in audible reflections patterns and discreet echoes.



Adjusting this up or down can be done graphically in the Envelope window, or by click-holding and dragging the mouse cursor up/down on the numerical entry.

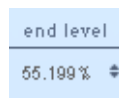
Ramp Time

This parameter adjusts the length of time elapsed between the *Initial* and *End Density* levels. Adjusting this up or down can be done graphically in the Envelope window, or by click-holding and dragging the mouse cursor up/down on the numerical entry.



End Level

As above, but controls the reverb tail. If you select an *End Level* (End Density) value that is too low, the reverb tail will sound grainy. You may also find that the stereo spectrum is affected by lower values. Adjusting this up or down can be done graphically in the Envelope window, or by click-holding and dragging the mouse cursor up/down on the numerical entry.



Reflection Shape

The range from 0 to 100 determines the steepness (shape) of the early reflection clusters as they bounce off the walls, ceiling, and furnishings of a virtual space. A value of 0 results in clusters with a sharp contour and a value of 100 results in an exponential slope, and a smoother sound.



This is handy when recreating rooms constructed of different materials. Use of this parameter, in conjunction with suitable settings for the *Envelopes*, *Density*, and *Early Reflection* will assist you in creating rooms of almost any shape and material.

Click-hold on the triangle, and slide left/right to adjust.

Filter Parameters

The filter section of the Space Designer provides control over the timbre of the reverb. Not only can you select from several filter types, but you also have envelope control over the filter cutoff, independent from the Volume Envelope. Changes to the filter settings result in a recalculation of the IR, rather than a straight change to the sound as it plays through the reverb.



Filter On/Off

Switches the filter section on and off. This must be switched *on* if you wish to make use of *any* of the Filter and Filter Envelope controls.

Filter Mode

Switches between four modes. Click on the desired *LP* (lowpass) *6 dB* and *12 dB*, *BP* (bandpass) or *HP* (highpass) value. To explain:

- *6 dB* (LP)—Bright, good general purpose filter mode. It can be used to retain the top end of most material, while still providing some filtering/control.
- *12 dB* (LP)—Useful where you want a warmer sound, without drastic filter effects. It is handy for smoothing out bright reverbs.
- *BP*—6 dB per octave design. Reduces the amount of signal that surrounds the mids of the input material, leaving the frequencies around the cutoff frequency intact.
- *HP*—12 dB per octave/two-pole design. This filter reduces the level of frequencies that fall below the cutoff frequency.

Reso

The *Reso* (resonance) parameter emphasizes frequencies above, around or below the cutoff frequency—determined by the selected Filter mode. As you increase the *Reso* value, the sound will lose bass and become thinner.

Note: Use the *Reso* control to add a basic, general color to your reverb, or for drastic Filter effect sounds.

Effect Parameters

Low Shelving EQ

As the name suggests, this EQ has a specific frequency that, once set, only allows frequencies that fall below it to be affected. The *Gain* control determines the amount of cut or boost of frequencies below the value set with the *Freq* parameter (expressed in Hertz). To adjust these values, click-hold on the applicable parameter field or knob and drag the mouse up and down.



Stereo Spread

This facility applies additional stereo information to a *Synthesized IR*.



It has *no* impact when using an IR sample or when using the Space Designer as a mono plug-in.

Spread extends the stereo base to frequencies that fall below the frequency determined by the *Xover* (crossover) parameter.

At a *Spread* value of *0*, no stereo information is added (although the inherent stereo information of a stereo signal and reverb will be retained). At a value of *100* the left and right channel divergence is at its maximum.

The *Xover* parameter is set in Hertz. Any synthesized IR that falls below this threshold value will be processed by adjustments over *0* for the *Spread* parameter.

The effect enhances the perceived width of the signal, without losing the directional information of the input signal, normally found in the higher frequency range. Low frequencies are spread to the sides, reducing the amount of low frequency content in the center—allowing the reverb to nicely wrap around the mix.

Creating Impulse Responses

This section discusses briefly the different methods for creating your own Impulse Response files for use with Space Designer.

About Impulse Responses

Impulse Responses are recordings (stored as AIFF, SDII or WAV files) made in acoustic spaces. To create an impulse response the sound of a starter pistol, digital spike or sine sweep is recorded inside the desired room together with the resulting reflections.

Starter Pistols

Starter pistol shots are quite loud allowing a good signal to noise ratio to be recorded. However, a perfect, undistorted recording of a starter pistol shot is difficult to produce. In addition starter pistol shots contain very little high or bass frequency information.

Digital Spikes

A digital spike contains a wide frequency spectrum. You can create a spike by using the *Pencil* tool in Logic's Sample Edit window to draw a single sample in a silent sound file (a recording of silence). The catch with this method is that, although fast, the output level is often too low to be usable. You can, however, boost the signal in Logic.

Sweeps

The downside with starter pistols or digital spikes is that they release their (high level) energy in a very short time period. As an alternative, you can playback, and record a broadband audio sweep with optimal recording levels. For this a sine sweep covering the whole audible frequency range could be used.

The recording of a sine sweep that is played back in an acoustic environment is known as a sine sweep response, rather than an impulse response.

By means of a deconvolution process the sine sweep is removed from the sine sweep response recording, leaving you with an impulse response file that can be loaded into Space Designer. This method of creating impulse responses often results in a superior reverb sound quality.

Using the Deconvolution Facility

Now that you know about sine sweep responses you'd probably like to know how to use it to roll your own. It couldn't be easier.

Accessing the Deconvolution Facility



The Deconvolution facility is necessary only in combination with a sweep response recording.

To access the Deconvolution facility:

- 1 Bypass the plug-in by clicking on the *Bypass* button at the top of the of the Plug-in window. This conserves CPU resources.
- 2 Click on the *Deconvolution* button towards the top right of the Space Designer GUI. This will launch a File Selection box.
- 3 The File Selector menu bar will show *Load Coded Impulse Response for Deconvolution*. Click *Choose* and select the wet, reverberated sweep response recording.
- 4 The File Selector menu bar will change to display *Load Testsignal*. Click *Choose*. Now open the same dry sweep audio file you used as a source for the wet response recording.
- 5 You will be prompted to save the decoded audio file. Select the desired path and file name. Now the new IR will be calculated.
- 6 Any silence at the end and beginning of the new IR file should be removed using a sample editor.
- 7 This file can now be loaded into the Space Designer.

About Convolution

This section looks at the technicalities behind convolution (in a reverb sense).

The use of reverberation on dry sounds is commonplace in computer music production. Reverbs (and delays) can be used to simulate room and other acoustic spaces or entirely new sounds, which may (or may not) be related to an actual physical space.

There are several methods used to simulate and model different reverbs and physical environments. One technique involves the actual recording of the ambience of a room, and overlaying this recording (the impulse) onto another sound. This technique is called convolution.

Before we get into convolution, a few basics need to be covered on reverberation itself.

So, Just What Is Reverberation?

Put simply, reverberation is basically a delay of the source signal \times number of times (by a very small time value), which is then fed back onto itself, simulating the way sound bounces around a room.

Prior to the invention of digital systems, engineers used a variety of techniques to create reverb-type effects. As examples:

Reverberant chambers—sending sound via a loudspeaker into a tiled room and re-recording the reverberant sound—ever recorded a vocal in the toilet?

Reverb units with springs and metal plates hung from the walls—with the latter being close and distant-miked then re-recorded alongside the original signal.

There are some essential elements for creating a reverberant sound, with the key thing to remember being that reverb is simply a number of (delayed) copies of the original signal—which emulates the multiple versions of a sound bouncing around a room.

In a software-based reverb plug-in there may be one copy of the signal dedicated to making the first reflection: the very first reflected sound we hear when a signal is introduced into a reverberant space. Other algorithms within the plug-in may be dedicated to early reflections—the first sounds we hear after the initial reflection. The (delayed) reflections are fed back, and added, to the original signal at a lower level. This creates an effect known as comb filtering (a short feedback delay that emphasizes specific harmonics).

Following on from the delayed signals (first and early reflections) you'll generally find that filters are used to make the reverb tails sound as if they are far away. Filtration of higher frequencies—say 5 kHz and up—makes a signal sound farther away. This is because high frequencies have very little energy, and don't travel far. Filtration may also be used to soften any harmonics introduced by the comb filtering mentioned above. Needless to say, emphasis of particular harmonics can make a reverb sound very artificial, which may be just what you're after ... but not if you're simulating a room!

Note that not all reverb units/plug-ins offer this sort of filtration. Another approach is to both feed back and feed-forward the same signal simultaneously, phase inverted, to create an all-pass filter. With this type of filter/delay the comb filtering effect doesn't occur (because of the phase inversion of the feed-forward signal—this fills in the missing frequencies created by the comb filter of the feedback signal). This creates a blended reverb.

Typical reverb algorithms have parameters for:

- room size (church, club, closet, bathroom, and so on),
- brightness (hard walls, soft walls or curtains and so on) and
- feedback or absorption coefficient (are there people, carpets, and so on in the room?—how quickly does the sound die?).
- Upmarket reverbs may also contain several filter parameters.

When trying to simulate rooms, take the following into consideration:

- How large is the room? How long will it take for the first reflection to get back to your ears?
- What are the surfaces like in the room? What material are they made of? Not just the walls, ceiling, and floor, but are there objects in the room that can reflect sound waves? As examples: plastic chairs, tables, and so on.
- Are there surfaces in the room that can absorb sound? As examples: curtains, soft furnishings such as lounges. Even people in the room can also remove energy from reflected sounds!

Convolver Reverb

The Space Designer is, as you know, a convolver reverb. So, just how does convolution work? You've already learnt how to use the effect parameters, and how to record (or create) an impulse, and use the impulse response, but now we're going to look at the maths behind the process.

Convolution is a fairly complicated software process. It takes each sample in the impulse response file (of a given room) and multiplies that sample by each sample in the sound file that we want to place in the room. So each sample input sound file, such as a vocal sound file that you want to add reverb to, is multiplied by each sample in the impulse response file.

As an example, imagine a three second impulse file, and a one minute sound file that you want to add the reverberant characteristics of some space to. At 44.1 kHz, that's:

$$\begin{aligned} & 3 \text{ (sec. — IR)} \times 44,100 \text{ (Samples)} \times 60 \text{ (sec.—audio)} \times 44,100 \text{ (Samples)} \\ &= 180 \times 680,683,500,000 \\ &= 122,523,403,030,000 \text{ (Wow!)} \end{aligned}$$

This multiplication of each point in each function by every other point in the other function (in the time domain)—called a cross multiply—produces what we refer to as the convolution in the frequency domain.

As we're sure that few of you have applied mathematics degrees, we'll just say that there's a whole lot of computation going on! This computationally intensive (and remember that we only looked at a 60 second sound!) has not been widely adopted for the simple reason that the garden variety computer just simply wasn't fast enough to do it—until now, that is.

The reason why we can actually perform the necessary calculations in real-time these days is due to a mathematical operation known as the FFT (Fast Fourier Transform). For filters (and we use the term filters here as this is what is effectively happening—the input sound is being filtered by the impulse response) with lots of nonzero values, it is easier to compute the convolution in the spectral domain. Here's how:

If we want to convolve our music function (the sound you wish to process) against our filter function (the impulse response), this results in another sound (the convolved results). This convolved sound has a spectrum that is equal to the product of the spectrum of the music function and the filter function.

Put another way, the Fourier coefficients of the convolution can be computed by simply multiplying together each of the Fourier coefficients of the music and filter functions.

The End Result?

Knowing the ins and outs (mathematics) behind convolution doesn't really matter too much. The important thing is to have a large library of great impulse responses—the best sounding cathedrals, recording studios, concert halls, railway tunnels, electronic reverb units or even resonance bodies of instruments ... you name it!—and you can simulate any space for any sound.

Thankfully, we've created and included a number of Impulse Responses to get you started, and you may find that they're all you'll ever need.

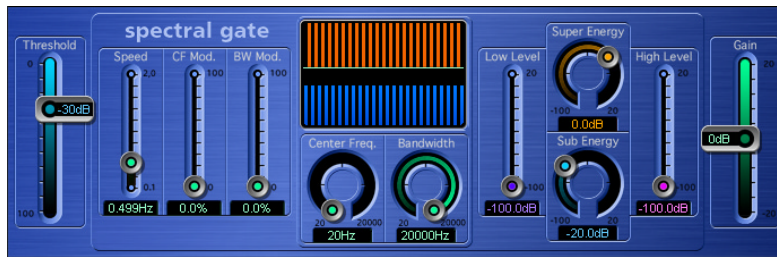
If you wish to create your own, the Internet is a great place to share them with other Space Designer users from around the globe.

Whatever way you decide to go, either using the factory IRs, downloading another user's efforts or rolling your own, Space Designer makes it easy to get that perfect reverb sound.

This chapter introduces Logic's special plug-ins.

This includes the Spectral Gate, Pitch Shifter II, SubBass, Denoiser, Exciter, and Stereo Spread plug-ins, amongst others.

Spectral Gate



The Spectral Gate separates the signals above and below the *Threshold* level into frequency ranges that can be independently modulated. It does this via a Fast Fourier Transformation (FFT) of the entire signal. Technical jargon aside, the Spectral Gate is a tool that lets you come up with some pretty wacky filtering effects.

The frequency range that you wish to process is defined by the *Center Freq.* and *Bandwidth* knobs. This frequency band is separated by steep slopes. Within this band, you can use the *Threshold* slider to determine a level that separates the frequencies above and below it. The frequencies above the Threshold are made audible with the *Super Energy* knob, the frequencies below it with *Sub Energy*. The original signal, outside the defined frequency band, can also be added to the mix: *Low Level* blends in the frequencies that lie below the frequency band (bass frequencies,) and *High Level*, the higher frequencies that lie above the defined frequency band.

The actual frequency band can be modulated by three parameters: *Speed* determines the modulation frequency, *CF Mod.* (Center Frequency Modulation) defines the intensity of the center frequency modulation, and *BW Mod.* (Band Width Modulation) controls the bandwidth modulation.

The *Gain* slider lets you adjust the level of the generated effects signal.

We suggest you use a drumloop when you begin experimenting with this plug-in. Set *Center Freq.* to its minimum, and *Bandwidth* to its maximum, values (the entire frequency range is processed at these settings). Turn up the *Super Energy* or *Sub Energy* knobs—one at a time—and experiment with the *Threshold* knob. You'll soon get a feel for how different Threshold levels affect the sound of *Super Energy* and *Sub Energy*. When you've come across a sound that you consider particularly nifty (or even just useful), you can narrow the Bandwidth drastically, gradually increase the Center Freq., and use the *Low Level* and *High Level* sliders to mix in some treble and bass from the original signal. At lower Speed settings, turn up the *CF Mod.* or *BW Mod.* knobs.

Pitch Shifter II



The Pitch Shifter II takes a minimalist approach—with just a few parameters available in the Editor view.

Semi Tones is used to set the transposition value—in semi-tone increments, within a range of one octave upwards or downwards. *Cents* controls detuning in increments equivalent to 1/100th of a semi-tone step. Use the *Mix* slider to control the desired balance between the original and processed signals.

The *Drums*, *Speech*, and *Vocals* buttons allow you to choose between three presets that optimize the Pitch Shifter II to deliver the best results for different audio material.

- When you select *Drums*, the groove of the original track remains intact.
- With *Vocals*, the intonation of the original is retained unaltered. Hence *Vocals* is well-suited for any signals that are inherently harmonic or melodious, such as string pads.
- *Speech* mode is a compromise between the two—the program attempts to retain both the rhythmic and harmonic aspects of the signal, which is desirable for complex signals, such as spoken-word recordings or rap music. Speech is thus also suitable for other hybrid signals, such as rhythm guitar.

Note: When in doubt, *Speech* is a good place to start. A/B the options to compare them, and use the one that suits a given recording best. When auditioning and judging settings for quality, it's a good idea to temporarily turn the *Mix* knob up to 100%. Keep in mind that Pitch Shifter II artefacts are a lot harder to hear when you mix a smaller percentage of a transposed audio to the overall signal.

In the Pitch Shifter II's Controls View you can create your own presets, using the *Delay* and *Crossfade* parameters. These parameters are only effective when you select the *Manual* option in the *Timing* menu. You can also select the *Auto* option here—the Pitch Shifter will then automatically create presets by analyzing the incoming signal. The *Stereo Link* parameter allows you to invert the stereo channel's signals, with processing for the right channel occurring on the left and vice versa.

Vocal Transformer



The *Vocal Transformer* plug-in allows you to manipulate vocal tracks in many different ways. If you want to transpose the pitch of a vocal line, to augment or diminish the range of the melody, or even reduce it to a single note to mirror the pitches of a melody—the *Vocal Transformer* will be your plug-in of choice. No matter how you change the pitches of the melody, formants remain the same. You can shift the formants independently, which means that you can turn a vocal track into a “Mickey Mouse” voice, while maintaining the original pitch.

The *Vocal Transformer* is well suited to extreme vocal effects. The best results are achieved with monophonic signals, including monophonic instrument tracks. The plug-in is not designed for polyphonic voices (a choir on a single track, for example) or other “chordal” tracks.

The *Vocal Transformer* operates in two modes: *Robotize on* or *off*.

The *Robotize* function is used for augmenting, diminishing or mirroring the melody. Let's first have a look at how the *Vocal Transformer* operates when *Robotize* is switched *off*.

Pitch, Formant, and Mix

Mix defines the level ratio between the original (dry) and effect signals.

Note: In order to get an idea of what the plug-in does, it may be helpful to listen to the original dry signal in the background. A *Mix* setting of about 75% should suffice.

The *Pitch* parameter transposes the pitch of the signal (up to) two octaves upwards or downwards. Adjustments are made in semitone steps. Incoming pitches are indicated by a vertical line.

Note: Transpositions of a fifth upward (*Pitch* = +7), a fourth downward (*Pitch* = -5) or by an octave (*Pitch* = ±12) are the most useful, harmonically.

As you alter the *Pitch* parameter, you might notice that the formants don't change.

Note: Formants are characteristic emphases of certain frequency ranges. They are static, and do not change with pitch. Formants are responsible for the specific timbre of a given human voice.

The *Pitch* parameter is expressly used to change the pitch of a voice, not its character. If you set negative *Pitch* values for a female soprano voice, you can turn it into an alto voice, without changing the specific character of the singer's voice.

The *Formant* parameter shifts the formants, while maintaining the pitch or while independently altering the pitch. If you set this parameter to positive values, the singer sounds like Mickey Mouse. By altering the parameter downwards, you can achieve sound effects reminiscent of Darth Vader from Star Wars.

Note: If you set *Pitch* to 0 semitones, *Mix* to 50% and *Formant* to +1 (while *Robotize* remains switched off), you can effectively place a singer (with a smaller head) next to the original singer. Both will sing with the same voice—in a choir of two. This choir effect is quite effective, and is easily controlled with the *Mix* parameter.

Robotize

If you switch *Robotize* on, the Vocal Transformer can augment or diminish the melody. You can control the intensity of this distortion with the *Tracking* parameter.

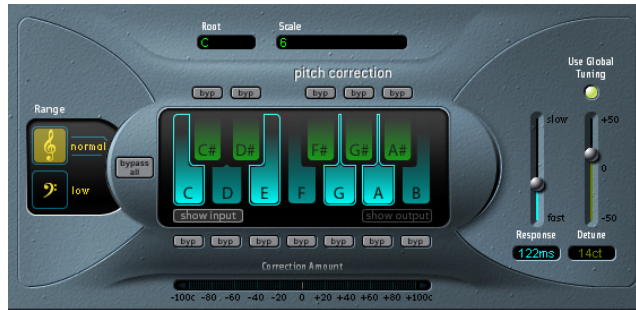
Note: The four -1, 0, 1, and 2 switches set the *Tracking* slider to values of -100%, 0%, 100% and 200%. These switches aren't individual parameters. They are merely additional controls that make faster to set the *Tracking* parameter to its most useful settings.

- At a value of 100% (switch 1), the range of the melody is maintained. Higher values augment, and lower values diminish the melody.
- At a setting of 200% (switch 2) the intervals are doubled.
- The reduction to 0% (switch 0) delivers interesting results, with every syllable of the vocal track being sung at the same pitch. Low values turn sung lines into spoken language.

- At a setting of -100% (switch -1), all intervals are mirrored.

The *Pitch Base* parameter is used to transpose the note that the Tracking parameter is following. As an example: the note which is spoken, if Tracking is set to 0% .

Pitch Correction



The Pitch Correction plug-in enables you to correct the pitch of audio tracks. Improper intonation is a common problem with vocal tracks, and this can be easily fixed with the Pitch Correction plug-in. The sonic artefacts of the process are minimal and can barely be heard, as long as the corrections are moderate. The natural articulation of the performance is preserved. Should large transposition intervals be used, you can achieve a static pitch, thus creating an interesting effect, known as the “Cher” effect. Any scale can be defined as a pitch quantization grid. Improperly intonated notes will be corrected to this grid. The keyboard at the center of the Plug-in window is used to define the scale that will serve as the pitch quantization grid.

Note: Polyphonic Recordings (choirs on one track) and highly percussive signals, with prominent noisy portions, can’t be corrected to a specific pitch. Despite this, feel free to try the plug-in on drum tracks!

Function Principle

The Pitch Correction algorithm is designed for corrections of smaller pitch shifting intervals. The correction works in a similar fashion to an audio tape, with the playback speed accelerated and slowed down in a way that ensures the singing voice always matches the correct note pitch. If you force the algorithm to correct larger intervals, you can create special effects.

Range

The *Normal/Low* parameter determines the pitch range that you wish to search for notes that need correction. Simply select *Normal* or *Low* for the track. *Normal* is the default range, and works for most audio material. *Low* should only be used for audio material that contains extremely low frequencies (below 100 Hz) which may cause the pitch detection to work incorrectly. The parameter does not affect the sound. It is designed to optimize tracking in the target pitch range.

Keyboard

The central element of the plug-in is the 12-note *Keyboard*. The *Keyboard* is used to exclude notes from the pitch quantization grid. When you first open the plug-in, all notes of the chromatic scale are selected. This means that every incoming note will be altered to fit the next semitone step of the chromatic scale. If the intonation of the singer is poor, this might lead to notes being incorrectly identified, and corrected. As an example, the singer might have intended to sing an *e*, but the note is actually closer to a *d#*. If you don't want the *d#* in the song, the *d#* key can be disabled on the *Keyboard*. Given that the original pitch was sung closer to an *e* than to a *d*, it will be corrected so that it becomes an *e*.

Note: The settings are valid for all octave ranges. Individual settings for different octaves aren't provided.

Show Input/Output

These two buttons below the *Keyboard* will display the pitch of the input or output signal, respectively, on the notes of the keyboard. This provides a real time, at-a-glance overview of either the original or corrected signal.

Programmable Scales (Root and Scale)

Clicking the *Scale* field allows you to select different pitch quantization grids. The scale that is set manually (with the *Keyboard*) is called the *User Scale*. The default setting is the *Chromatic* scale. The other scale names are self-explanatory. If you're unsure of the intervals used in any given scale, simply select it in the *Scale* field and check out the values shown in the *Keyboard* grid. You can alter any note in the scale by clicking on the *Keyboard* keys. Any such adjustments will overwrite the existing *User Scale* settings.

Note: There is only one *User Scale* per song. You can, however, create multiple *User Scales*, and save them as Pitch Correction plug-in *Settings* files.

The *Drone* scale uses a fifth as a quantization grid, the *Single* scale defines a single note. Both of these scales aren't meant to result in realistic singing voices, so if you're after interesting effects, you should give both of them a try.

Root allows you to select the root note of the scale (with the exception of the *User* and *Chromatic Scale*, where there is no root note (*none*)). You may freely transpose the major and minor scales as well as the scales named after chords.

Bypass

Use of the small bypass buttons (*byp*) above the green (black) and below the blue (white) keys excludes notes from correction. This is useful for blue notes. Blue notes are notes that slide between pitches, making the major and minor status of the keys difficult to identify. As you may know, one of the major differences between *C minor* and *C major* is the *e flat* and the *b flat*, instead of the *e* and the *b*. Blues singers glide between these notes, creating an uncertainty or tension between the scales. Use of the bypass switches allows you to exclude particular keys from changes, leaving them as they were.

Bypass All

With *Bypass All* active, the input signal is passed through unprocessed and uncorrected. The *Bypass All* function is useful for spot corrections to pitch through use of Logic's automation system. *Bypass All* is optimized for seamless bypass switching in all situations.

Note: You'll often find that it's best to only correct notes with the most harmonic gravity. As an example, select a *Scale* of *sus 4* and the *Root* note of the song's key. This will limit correction to the root note, the fourth and the fifth of the key scale. Switch all other notes to *Bypass*, and only the most important and sensitive notes will be corrected, while all other singing remains untouched.

Reference Tuning

The *File > Song Settings > Tuning* settings determine the tuning reference for all software instruments. If you engage *Use Global Tuning* in the Pitch Correction Plug-in window, the *Global Tuning* settings will be used for the pitch correction process. If this parameter is switched off, you can use this section to freely set the desired reference tuning in cents.

Note: Tunings that differ from software instrument tuning can be interesting, when you want to individually correct the notes of singers in a choir. If all voices were individually and perfectly corrected to the same pitch, the choir effect would be partially lost. You can avoid this by (de)tuning the pitch corrections individually.

Response

Response determines how quickly the voice reaches the corrected destination pitch. Singers use portamenti and other gliding techniques. If you choose a *Response* that's too high, seamless portamenti turn into semitone-stepped glissandi, but the intonation will be perfect. If the *Response* is too slow, the pitch of the output signal won't change quickly enough. The *Response* of pitch changes is indicated in milliseconds. The optimum setting for this parameter depends on the singing style, tempo, vibrato, and quality of the original performance.

Correction Amount

The amount of pitch change is indicated in the horizontal bar display. If you keep a close eye on this display, you can use it for two important tasks: To better understand the inner workings of the algorithm works, and adjust the Response accordingly. You can also use the display when discussing (and optimizing) the vocal intonation with a singer during a recording session.

Automation

As with almost all Logic plug-ins, *Pitch Correction* can be fully automated. This means that you can automate the *Scale* and *Root* parameters to follow the harmonies of the song. Simply select the *Touch* or *Write* automation modes for the track, and change the *Scale* and *Root* parameters by key during playback. Depending on the quality of the original intonation, setting the key *Scale* might suffice. Weaker intonations might need more significant changes to the *Scale* and *Root* parameters.

Note: The keys of the scale can't be switched by MIDI notes.

SubBass



This plug-in generates frequencies below those in the original signal—in other words an artificial bass.

Warning: This process is known as a loudspeaker killer! Choose moderate monitoring levels, and never try to play back sub-bass frequencies with loudspeakers which aren't capable of doing so. Never try to force a loudspeaker to output these frequency bands with an EQ.

The simplest application of the SubBass plug-in is as an octave divider, similar to “Octaver” effects pedals for electric bass guitar. A simple frequency division circuit in such pedals requires a monophonic input sound source, with a clearly defined pitch. This type of device is only capable of producing an output signal which is one or two octaves lower than the input signal. An octave is a frequency division by two. A ratio of four means two octaves, and a ratio of eight equals three octaves.

As opposed to a pitch shifter, the waveform of the signal synthesized by the SubBass plug-in has nothing to do with the waveform of the input signal. It’s shape is sinusoidal—but a pure sine wave is rarely achievable in complex arrangements. It is mixed together with the original signal. The mix ratio is defined by the *Mix* parameter.

SubBass creates two bass signals, derived from two freely definable portions of the incoming audio signal. Therefore, you are not limited to monophonic signals with a defined pitch, but complex summed signals may be processed as well. *Center High* and *Center Low* define the center frequency of the band that the transposed signal will be derived from. *Bandwidth* sets the bandwidth of the frequency band.

Within these frequency bands, the filtered signal should have a reasonably stable pitch, in order to be analyzed correctly. The graphic shows the frequency bands of a typical boom box, which transposes two frequency bands, with the width of a fifth, by one octave each. If you set the analyzed frequencies a little higher, the SubBass plug-in plays these frequencies, much like a bass guitar player doubling the lower notes of a guitar player.

In real life, narrow bandwidths lead to the best results as they avoid unwanted intermodulations. Set *Center High* a fifth higher than *Center Low*, which means a factor of 1.5 for the *Center Frequency*. Derive the sub-bass to be synthesized from the existing bass portion of the signal, and transpose by only one octave in both bands (*Ratio* = 2). Do not overdrive the process, or you will introduce distortion. If you recognize frequency gaps, move the *Center Frequencies*, or widen the *Bandwidth* a little.

Be prudent when using the SubBass plug-in, and compare the extreme low frequency content of your mixes with other productions.

Denoiser



The Denoiser eliminates or reduces almost any kind of noise floor.

Denoiser Parameters

Threshold

The value of this parameter determines how high you think the noise floor of the material is.

Tip: Find a passage where only noise can be heard in isolation, and set the Threshold value so that only signals of this volume will be filtered out.

Reduce

Reduce determines the level that the noise floor should be reduced to. A CD theoretically has a maximum signal to noise ratio of 96 dB. Each 6 dB reduction is equivalent to halving the volume level (a 6 dB increase equals a doubling of the volume level).

If the noise floor of your recording is very high (on recordings from cassette—more than -68 dB), you should be content with reductions of 83 to 78 dB, provided that there aren't any plainly audible side effects. After all, you have reduced the noise by more than 10 dB, which is less than half of the original volume.

Noise Type

This value effectively states what type of noise you think the material contains:

- A value of 0 means white noise (equal frequency distribution);
- Positive values change the noise type to pink noise (harmonic noise; greater bass response),
- Negative values change the noise type to blue noise (Hiss—tape noise).

Smoothing

The Denoiser uses FFT (Fast Fourier Transform) analysis to recognize frequency bands of a lower volume and less complex harmonic structure, and then reduces them to the desired dB value. In principle, this method is never exact, as neighboring frequencies will also be affected.

If you use the Denoiser too aggressively, the algorithm will produce artefacts, such as “glass noise” which—in most cases—are less desirable than the existing noise.

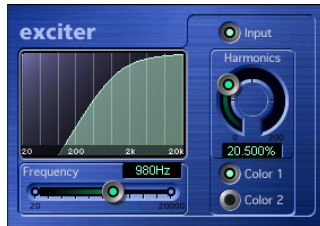
Therefore, there are three parameters for reducing this effect in all three dimensions of sound:

- Time Smoothing
This is the simplest form of smoothing. This parameter sets the time required by the Denoiser to reach (or release) maximum reduction.
- Frequency Smoothing
This parameter sets a factor for smoother transition of the denoising to neighboring frequencies. More precisely: If the Denoiser recognizes that only noise is present in a certain frequency band, the higher the Frequency Smoothing parameter is set, the more it will also change the neighboring frequency bands to avoid “glass noise.”
- Transition Smoothing
This parameter sets a factor for smoother transition of the denoising to neighboring volume levels. More precisely, if the Denoiser recognizes that only noise is present in a certain volume range, the higher the Transition Smoothing parameter is set, the more it will also change similar level values to avoid “glass noise.”

The Graphic Display

The graphic displays how the lowest volume levels of your audio material (which ideally is only noise) will be reduced.

Exciter



An Exciter generates high frequency components that are not part of the input signal. It's a nonlinear distortion process, that resembles overdrive and distortion effects. As opposed to these processes, however, the distorted signal is mixed with the dry signal. In addition, the harmonics generator is fed by a highpass filtered version of the input signal. This means that the artificial harmonics have frequencies at least one octave above the lowest frequency being allowed through by the highpass filter. Human hearing can not easily distinguish artificial harmonics in very high frequency ranges.

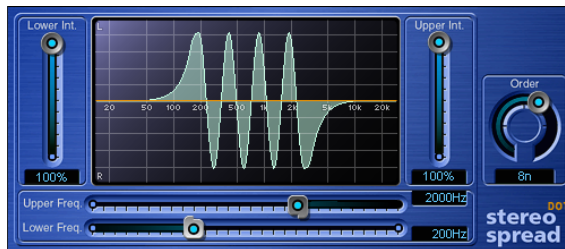
The cutoff frequency parameter of the high pass filter is called *Frequency*. The graphic displays the frequency range that is used as the source signal for the process. *Harmonics* controls the level of the effect signal mixed to the original signal. If you disable *Input*, the original signal won't be fed through. This should be your approach when using the Exciter plug-in on a Bus channel, being fed by Sends from several channel strips simultaneously, or should you wish to listen to the soloed effect signal.

The lower you tune the filter, the more harmonics the effect will produce, the lower the frequencies of the artificial harmonics will be, and the less natural they will sound.

You can choose between two sound characteristics for the distortion. *Color 1* is less dense in spectrum than *Color 2*. The effect of *Color 2* is more intense, but it also introduces more unwanted intermodulation distortion.

Basically, higher settings for *Frequency* and *Harmonics* are preferable, because we cannot easily distinguish between artificial and original high frequencies. Exciters add life to digital recordings. They are especially well suited to audio tracks where the treble is weak. Exciters are also popular for guitar tracks.

Stereo Spread



The Stereo Spread plug-in is a standard effect for mastering. There are several ways to extend the stereo base, including complex algorithms of the Ghetto blaster type (“wide” or “space”), or tampering with the signal’s phase. They all sound great, but most of them could be very dangerous to your mix, ruining the transient response, for example.

The Stereo Spread follows a sure and simple method: it extends the stereo base by alternately distributing a (selectable) number of frequency bands from the middle frequency range to the left and right channels. This greatly increases the perception of stereo width, especially when applying a stereo effect to monaural recordings—without making them sound totally alien.

Parameters of the Stereo Spread

Order

The *Order* knob determines the number of frequency bands that the signal is to be divided into.

Upper Int.

This parameter controls the intensity of the base extension of the upper frequency bands.

Lower Int.

This parameter controls the intensity of the base extension of the lower frequency bands.

Note that

- the stereo effect shows up mainly in the middle and higher frequencies, and
- if very low frequencies are distributed between the left and right speakers, the energy distribution for both speakers will be significantly reduced.

Therefore, it is always best to select a lower *Intensity* setting for the lower frequency bands, and to avoid setting the *Lower Freq.* below 300 Hz.

Upper Freq.

This sets the upper limit of the highest frequency band to be distributed in the stereo image.

Lower Freq.

This sets the lower limit of the lowest frequency band to be distributed in the stereo image (see the note in the Lower Int. section).

The Graphic Display

The graphic display shows the selected order (how many bands the signal is divided into), and how strong the Stereo Spread effect will be in the upper and lower frequency bands.

The upper section represents the left channel, the lower section the right channel, and the frequency scale shows the frequencies in ascending order from left to right.

This chapter introduces you to Logic's Helper plug-ins.

This includes Test Oscillator, Tuner, Gain, I/O, Direction Mixer, MultiMeter, Levelmeter, and Correlation Meter.

Test Oscillator



The Test Oscillator generates a static frequency or a sine sweep. The latter is a user-defined frequency spectrum tone sweep that can be used for the creation of Impulse Responses, for use with Logic's Space Designer (see "Convolution Reverb: Space Designer", on page 117). Here's a quick run-down on the parameters:

The *Waveform* radio buttons select the type of waveform to be used for test tone generation. Selection of any button will generate a fixed oscillation with the selected waveform. Some further information about the *Waveform* section:

- the *square* and *needle* waveforms are available as either aliased or anti aliased versions, when pressed in conjunction with the *anti aliased* button between them.
- *needle* is a single needle impulse waveform.
- if the *Sine Sweep* button is active, the fixed oscillator settings in the *Waveform* section above are disabled.

Frequency determines the frequency of the oscillator. *Level* determines the overall output level of the Test Oscillator.

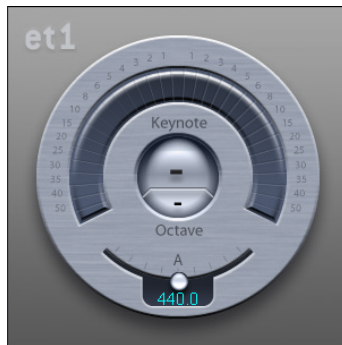
The *Sine Sweep* section generates a user-defined frequency spectrum sine wave sweep. The *Time* field determines the duration of the sweep. The *Start* and *End Freq(ueency)* parameters define the oscillator frequency at the beginning and end of the sine sweep. The *Trigger* button behavior can be switched via the edit field below:

- single: pressing the *x1* symbol triggers the sweep once.
- continuous: pressing the *Infinity* symbol triggers the sweep indefinitely.

Tuner

The ET1 Tuner plug-in can be used to tune acoustic instruments. This ensures that software instruments, existing samples or recordings are perfectly tuned to any new acoustic recordings you may make.

You would normally insert the ET1 Tuner into an *Input* fader channel.



Use couldn't be simpler. There is a single tuning control at the bottom of the ET1 Tuner interface. To adjust, simply click and drag it to the desired pitch for A. By default, the ET1 Tuner is set to concert pitch A = 440 Hz.

The *Keynote* and *Octave* panels display the incoming note pitch, and the octave that the incoming note belongs to. This matches the MIDI octave range, with the C above middle C displayed as C4, and middle C displayed as C3.

The numeric semicircle around the top of the ET1 interface displays the amount of variance—in cents—from the original pitch. The range is displayed in single semitone steps ± 6 cents, and then in larger increments to a maximum of ± 50 cents.

If the incoming note is slightly flat, the indicator segments to the left will be illuminated. If the incoming note is slightly sharp, the indicator segments to the right will be illuminated.

When the pitch is perfect, the center segment is lit.

Adjust the pitch of your instrument—using the tuning nuts on your guitar, for example—until the center segment (at the very top of the ET1) is illuminated. This indicates that the incoming note/string pitch is perfectly tuned.

Gain



This plug-in allows a constant amplification or reduction, by a specific decibel amount, of an Audio Track or Bus Object. It is ideal for use in situations where you're working with automated tracks during post-processing, and you want to quickly adjust master levels. This could be the case when you've inserted an additional plug-in that doesn't feature a dedicated gain control, or you want to change the basic level of a track for a remix version.

Note: The Gain plug-in replaces the previous Volume and Gainer plug-ins.

- The old Gainer plug-in will remain in Logic to retain compatibility with older songs, but it is no longer directly accessible from the plug-in menu.
- Automation data is upwardly compatible for any existing Gainer instances.
- Any Volume plug-in instances in older songs will automatically be replaced by the Gain plug-in. Any existing Volume plug-in automation data will be understood and used by the Gain plug-in.
- Similarly, the Settings files used by the Volume and Gainer plug-ins can be read by the Gain plug-in. If such Settings are used in a loaded song file, the Gain plug-in will replace the Volume plug-in, and equivalent parameters will be set.

Parameters

The following parameters are available in the Gain plug-in:

Gain

This control adjusts levels from -96 to +24 dB, in steps of 0.1 dB. Press Shift while dragging on the Gain parameter to adjust in fine increments.

Phase Invert

These buttons invert the phase of the left and right channels. This allows you to combat time alignment problems, particularly those caused by running multiple microphones at the same time. When you invert the phase of a signal, it sounds identical to the original. Only when the signal is heard in conjunction with other signals does phase inversion have an audible effect. As an example, if you mic a snare drum from the bottom as well as from the top, you should invert the phase of the bottom microphone's signal so that it is in phase with the top mic signal.

Stereo Balance

The Stereo Balance control distributes the incoming signal between the left and right channels.

Swap Left/Right

This button swaps the left and right output channels. It is placed after the Stereo Balance in the signal path.

Mono

Activation of the Mono button outputs the summed mono signal on both the left and right channels.

Note: The Gain plug-in is available in $m \rightarrow m$, $m \rightarrow s$ and $s \rightarrow s$ configurations. In $m \rightarrow m$ and $m \rightarrow s$ modes, only one *Phase Invert* button is available. In the $m \rightarrow m$ version, the *Stereo Balance*, *Swap Left/Right* and *Mono* parameters are disabled.

I/O

The I/O insert plug-in allows you to insert external effect processors into the Logic mixer.

This only makes sense with audio cards that have more than two outputs, thus providing discrete inputs and outputs (analog or digital) for sending signals to/from the external effect processor.

The I/O plug-in offers assignment pull-down menus by which it can access the input/outputs provided by your audio hardware. In and output levels can be adjusted with the respective volume sliders.

- Output—assigns the output (or output pair) of the plug-in.
- Output Volume—adjusts the volume of the output signal.
- Input—assigns the input (or input pair) of the plug-in.
- Input Volume—adjusts the volume of the input signal.

Direction Mixer



The Direction Mixer plug-in offers the following features:

- MS Decoder
- The option of influencing the stereo base
- Variable pan positioning of a stereo recording

Parameters of the Direction Mixer Plug-in

Input

The *LR* and *MS* radio buttons determine whether the input signal is a standard left/right signal, or if you're dealing with an *MS* encoded (middle and side) signal, for example when the two sides of an *MS* stereo mic setup were recorded directly.

Direction

This parameter determines the direction from which the middle of the recorded stereo signal will emanate from within the mix, or in less complicated terms, its pan position.

At a value of *0*, the middle of the stereo recording will be dead center within the mix. Positive values shift the middle of the stereo recording towards the left, negative values towards the right. At 90° , the middle of the stereo recording is panned hard left, at -90° , hard right. Higher values shift the recorded signal back to the center of the stereo mix, except that the stereo sides of the recording are swapped. At values of 180° or -180° , the middle of the recording is yet again dead center of the mix, although the left side of the recording is audible on the right side of the mix, and vice versa.

Basis

This parameter determines the spread of the stereo base. At a neutral value of *1*, the left side of the signal is positioned precisely on the left, and the right side precisely on the right. As the values decrease, the two sides increasingly move towards the center of the stereo image. A value of *0* produces a mono signal (both sides of the input signal are routed to the two outputs at the same level—a true middle signal). At values greater than *1*, the stereo base is extended out to an imaginary point beyond the spatial limits of the speakers. In terms of *MS* levels, this is an involved way of saying that the level of the side signal is increased so that it is higher than the level of the middle signal. At a value of *2*, only the side signal remains audible (on the left you'll hear L-R and on the right R-L).

If you chose to use the Direction Mixer simply to spread the stereo base, please keep in mind that as the *Basis* values increase beyond 1, monaural compatibility decreases accordingly. Once you process a stereo signal with an extreme setting of 2, when you play back the track in mono, the signal will be cancelled out completely—after all, L-R plus R-L doesn't leave you with much.

What is MS?

Relegated to obscurity for a good long while, MS stereo (middle-side as opposed to left-right) has recently enjoyed a renaissance of sorts. To explain; two microphones are stacked on a stand or suspended from the ceiling so that they are positioned as closely together as possible. One of the two is a cardioid (or omnidirectional) mic which faces the sound source that you want to record directly—in straight alignment. The other is a bidirectional mic, the sensitive axes of which point to the left and right at 90° angles. The cardioid mic delivers the middle signal, the bidirectional mic the side signal. Simply record the middle signal to the left side of a stereo track and the side signal to the right. This configuration enables the Direction Mixer to easily decode MS recordings. The advantage that MS recordings have over XY recordings (two cardioid microphones aligned so that they are directed to a point halfway to the left and right of the sound source) is that the stereo middle is actually located on the on-axis (main recording direction) of the cardioid mic. This means that slight fluctuations in frequency response that occur off the on-axis—as is the case with every microphone—are less troublesome.

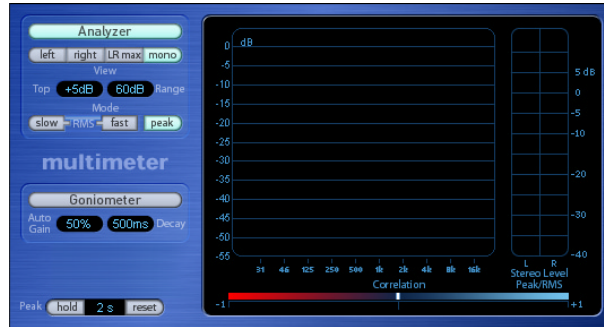
In principle, MS and LR signals are equivalent and can be converted at any time. When “-” signifies a phase inversion, then the following applies:

$$M = L+R$$

$$S = L-R$$

In addition, L can also be derived from the sum of and R from the difference between M and S. Here's some interesting trivia for you: Radio (FM) broadcasts feature M and S stereo. The MS signal is actually converted to a signal suitable for the left and right speakers by the receiver.

Multimeter



This plug-in consists of a collection of professional gauge and analysis tools, namely;

- a 1/3 octave Analyzer
- a Goniometer for judging the phase coherency in the stereo sound field
- a Correlation Meter to spot mono phase compatibility
- an integrated Level Meter

The control panel to the left of the display allows you to switch between the Analyzer and Goniometer, and contains parameter controls for the Multimeter.

The Stereo level and Phase Correlation meter is always displayed.

1/3 Octave Spectrum Analyzer

This tool analyses the level of 31 independent frequency bands. Each frequency band represents one third of an octave. The Analyzer complies with the specifications outlined by IEC document 1260.

Analyzer Button

The *Analyzer* button activates the Analyzer (and deactivates the Goniometer). The four buttons below determine the portion of the Input signal that the Analyzer displays.

You can choose to view the *Left* or *Right* channel independently.

LR max shows the maximum Band levels of either channel, while *Mono* displays the levels of the stereo signal as a summed mono entry.

View

The *View* options determine the level represented by the top line of the scale shown in the display (Top; range: -40 to $+20$ dB) and the overall dynamic range of the Analyzer (range; range: 20 to 80 dB).

These two parameters can also be set directly in the bar graph. To do so, click-hold and drag on the bar graph to move the top line of the display. Click-holding and dragging directly on the dB scale allows you to compress or expand the range of the scale.

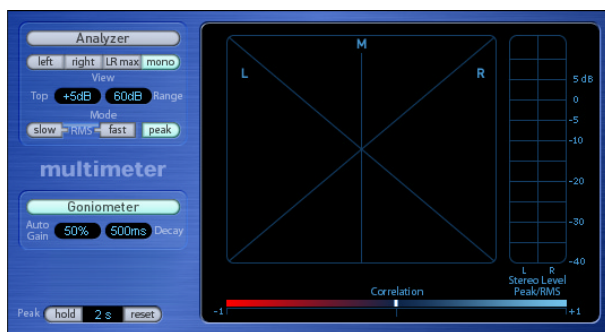
These features are useful when analyzing highly compressed material, as you can identify smaller level differences more easily by moving and/or reducing the display range.

Mode

There are three display response modes: *RMS slow*, *fast*, and *Peak*.

The two *RMS modes* with *slow* and *fast* response settings show the effective signal average (“Root Mean Square”), and provide a good representative overview of the perceived volume levels. The *Peak* mode shows level peaks accurately.

Goniometer



The idea of the Goniometer was born with the advent of early two channel oscilloscopes. To use such devices as Goniometers, users would connect the left and the right stereo channels to the X and Y inputs, while rotating the display by 45 degrees. This would result in a useful visualization of the signal's stereo phase.

A Goniometer helps you to judge the coherence of the stereo image. Phase problems are easily spotted as trace cancellations along the center line (*M—mid/mono*).

The signal trace slowly fades to black. Not only does this imitate the retro glow of the tubes found in older Goniometers, it also significantly enhances the readability of the tool.

Goniometer Button

This button activates the Goniometer (and deactivates the Analyzer).

Auto Gain

In order to obtain a higher readout on low-level passages, you can use the *Auto Gain* display parameter. When activated, this parameter allows the display to automatically compensate for low input levels. The amount of compensation can also be set with the *Auto Gain* parameter. The *Auto Gain* parameter can also be adjusted by click-holding and dragging directly in the display area of the Goniometer.

Note: *Auto Gain* is a display parameter only, and increases display levels in order to enhance readability. The actual audio levels are not affected by this parameter.

Decay

The *Decay* parameter determines the time that it takes for the Goniometer trace to fade to black.

Correlation Meter

This is another phase measurement tool that gauges the phase relationship of a stereo signal. This meter is also available as a separate plug-in.



The Correlation Meter's scale ranges from -1 to $+1$.

- A $+1$ correlation value indicates that the left and right channels correlate 100%. In other words, the left and right signals are in phase.
- Correlation values in the blue zone (between $+1$ and the middle position) indicate that the stereo signal is mono compatible.
- The middle position indicates the highest permissible amount of left/right divergence, which is often audible as an extremely wide stereo effect.
- Once the correlation meter moves into the red area to the left of the center position, out-of-phase material is present. This will lead to phase cancellations if the stereo signal is combined into a mono signal.

Level Meter (Peak/RMS Meter)

The stereo Level Meter shows the current signal level on a logarithmic scale—using two blue bars. If the level is higher than 0 dB , the portion of the bar above the 0 dB point will turn red.

This meter is also available as a separate small plug-in.

The current peak values are displayed numerically (in dB increments), above the Level Meter. The values are reset by clicking into the display.

While the separate *Level Meter* plug-in is switchable between Peak and RMS characteristics, the *Level Meter* of the Multimeter shows Peak and RMS values simultaneously. The *RMS level* is represented by a dark blue bar while the *Peak level* bar is light blue in color.



Peak Hold

This section controls the peak hold behavior of all metering tools.

Hold

The Hold button activates the Multimeter's peak hold function. The duration of the Hold time is set in the parameter field alongside the Hold button.

- In the Analyzer and Level Meter a small yellow segment above each 1/3 octave level bar, and stereo level bar, labels the most recent peak level.
- In the Goniometer, all illuminated pixels of the display are held in place during peak hold.
- In the Correlation Meter, a growing horizontal area around the white correlation indicator represents any phase correlation deviations—in both directions. A vertical red line to the left of the correlation indicator permanently shows the maximum negative phase deviation value. This line can be reset by clicking on it during playback.

(Value)

This parameter field allows the hold time for all metering tools to be set to 2 s, 4 s, 5 s, 6 s or, *infinite* by click/dragging on the value.

Reset

During operation, the *Reset* button resets the peak hold segments of all metering tools.

It should be noted that all metering tools are automatically reset when playback is stopped.

Correlation Meter

The Levelmeter is also part of the MultiMeter plug-in. A detailed description of its parameters can be found in the Multimeter's "Correlation Meter" section, on page 165.

Levelmeter

The Levelmeter is also part of the MultiMeter plug-in. A detailed description of its parameters can be found in the Multimeter's "Level Meter (Peak/RMS Meter)" section, on page 165.

If you are new to vocoders you should read this chapter.

It provides you with basic knowledge about vocoders and their functionality. You will also find tips on using vocoders, and achieving good speech intelligibility.

What Is a Vocoder?

The word Vocoder is an abbreviation for VOice enCODER. As with many technologies in this otherwise beautiful world, it is a child of war. The Vocoder was initially developed for secure speech transmission over telephone lines which couldn't be tapped. To achieve this, the speech signal was analyzed and only the cryptic results of the analysis were transmitted over telephone lines. On the receiving side, these results were used to synthetically rebuild the original voice signal.

Fortunately, Vocoderes are used nowadays for altogether more peaceful purposes—namely for music. A Vocoder analyses and transfers the sonic character of the audio signal arriving at its analysis input to the audio signal present at its synthesis input. The result of this process is heard at the output of the Vocoder.

The classic vocoder sound uses speech as the analysis signal and a synthesizer sound as the synthesis signal. This classic sound was popularized in the late 70's and early 80's. You'll probably know it from tracks such as "O Superman" by Laurie Anderson, "Funky Town" by Lipps Inc. and numerous Kraftwerk pieces—from "Autobahn" and "Europe Endless" up to "The Robots" and "Computer World".

Away from these "singing robot" sounds, vocoding has also been used in many films. As examples: the Cylons in *Battlestar Galactica*, and most famously, on the voice of Darth Vader from the Star Wars saga.

Vocoding, as a process, is not strictly limited to vocal performances. You could use a drum loop as the analysis signal to shape a string ensemble sound arriving at the synthesis input.

How Does a Vocoder Work?

The speech *analyzer* and *synthesizer* referred to above are actually two *filter banks* of bandpass filters. Bandpass filters allow a frequency band (a slice) in the overall frequency spectrum to pass through unchanged, and cut the frequencies which fall outside of the band's range.

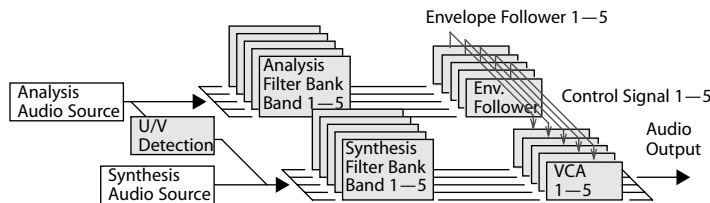
In Logic'S EVOC 20 plug-ins, these filter banks are named the *Analysis* and *Synthesis* sections. These filter banks have a matching number of corresponding bands—if the *Analysis* filter bank has five bands (1, 2, 3, 4, and 5), there will be a corresponding set of five bands in the *Synthesis* filter bank. Band 1 in the *Analysis* bank is matched to band 1 in the *Synthesis* bank, band 2 to band 2, and so on.

The audio signal arriving at the analysis input passes through the *Analysis* filter bank where it is divided into up to 20 bands.

An *envelope follower* is coupled to each filter band. The *envelope follower* of each band tracks (follows) any volume changes in the portion of the audio source allowed to pass by the associated bandpass filter. In this way, the *envelope follower* of each band generates *dynamic control signals*.

These *control signals* are then sent to the *Synthesis* filter bank where they control the levels of the corresponding *Synthesis* filter bands. This is done via *VCAs*—*Voltage Controlled Amplifiers*. This allows the volume changes of the bands—and thus the changes of the original sound—in the *Analysis* filter bank to be imposed on the matching bands in the *Synthesis* filter bank.

The more bands a Vocoder offers, the more precisely the original sound's character will be re-modeled.



How Does a Filter Bank Work?

If you removed all circuits responsible for transferring the sonic characteristics from the analysis to the synthesis signal from a Vocoder, and dispensed with the detection of voiced or unvoiced signals, you'd be left with two filter banks—the analysis and synthesis filters. To use these musically, you would need to ensure that you could control the output level of each bandpass filter. With this level of control, you can apply unique and dramatic changes to the frequency spectrum.

Analyzing Speech Signals

The principles you've been introduced to thus far are insufficient for the transmission of speech signals.

The reason is that human speech consists of a series of voiced sounds (tonal sounds) and unvoiced sounds (noisy sounds). The main distinction between voiced and unvoiced sounds is that voiced sounds are produced by an oscillation of the vocal cords, while unvoiced sounds are produced by blocking and restricting the air flow with lips, tongue, palate, throat, and larynx.

Should speech containing voiced and unvoiced sounds be used as a Vocoder's analysis signal, but the synthesis engine doesn't differentiate between voiced and unvoiced sounds, the result will sound rather toothless. To avoid this, the synthesis section of the Vocoder must produce different sounds for the *voiced* and *unvoiced* parts of the signal.

In Logic's EVOC 20 PS and the EVOC 20 TO Vocoder plug-ins, there is an Unvoiced/Voiced detector. This unit detects the unvoiced portions of the sound in the analysis signal and then substitutes the corresponding portions in the synthesis signal with *Noise*, a mixture of *Noise* + *Synth* or with the original signal (*Blend*). If the U/V Detector detects voiced parts, it passes this information to the *Synthesis* section, which uses the normal synthesis signal for these portions. Control over unvoiced/voiced sound detection, type, and level is found in the *U/V Detection* section of Logic's vocoder plug-ins.

Tips for Better Speech Intelligibility

The classic vocoder effect is very demanding, with regard to the quality of *both* the *analysis* and *synthesis* signals. Furthermore, the vocoder parameters need to be set carefully. Following, are some tips on both topics.

Editing the Analysis and Synthesis Signals

Compressing the Side Chain

The less the level changes, the better the intelligibility of the vocoder. We therefore recommend that compression be used in most cases.

Enhancing High Frequency Energy

The vocoder, in a way, always generates the intersection point of the analysis and synthesis signals. To explain: If there's no treble portion in the *analysis* signal, the resulting vocoder output will also lack treble. This is also the case when the *synthesis* signal features a lot of high frequency content. This is true of each frequency band. As such, the vocoder demands a stable level in *all* frequency bands from *both* input signals, in order to obtain the best results.

Due to the way human beings hear, the intelligibility of speech is highly dependent on the presence of high frequency content. To aid in keeping speech clear, it may be worthwhile using equalization to boost or cut particular frequencies in analysis signals before processing them with the vocoders.

If the Side Chain (analysis) signal consists of vocals or speech, a simple shelving filter should be sufficient. It doesn't require much processing power, and efficiently boosts the high-mid and treble range, which is so important for speech intelligibility.

If the *synthesis* signal lacks treble energy, it can be generated with a distortion effect. The overdrive plug-in is perfect for this purpose.



Avoiding Sonic Artefacts

A common problem with vocoder sounds are sudden signal interruptions (ripping, breaking sounds) and rapidly triggered noises, during speech pauses.

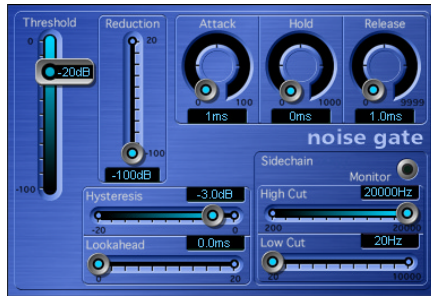
Release Parameter in the Analysis Section

The *Release* parameter defines the speed that a given synthesis frequency band can decrease in level, if the signal level of the respective analysis band decreases abruptly. The sound is smoother when the band levels decrease slowly. To achieve this smoother character, use higher *Release* values in the analysis section of the interface. Longer release times result in a washy sound.



Short *Attack* values are no problem. They may, in fact, even be desirable when a fast reaction to impulse signals by the vocoder is required.

Gating Background Noises in the Side Chain



If the Side Chain signal is compressed, as recommended, the level of breath, rumble, and background noises will rise. These background noises can cause the Vocoder bands to open, but this is normally not intended. In order to eliminate these noises, it's therefore a good idea to employ a noise gate before compression and treble boosting. If the Side Chain signal is gated appropriately, you may find that you want to reduce the analysis *Release* value. When gating speech and vocals, the *Hysteresis* parameter is important. *Threshold* defines the level, above which the gate will open. *Hysteresis* defines a lower *Threshold* level, under which the gate will close. The value is relative to the *Threshold* level. The graphic shows a *Threshold* setting, which is well-adapted to compressed speech. Unwanted triggering by low or high frequency noise is avoided by the Noise Gates' dedicated sidechain filters. The *Hold*, *Release*, and *Hysteresis* values shown are typical level envelopes, suitable for most vocal and speech signals.



Treatment of the Side Chain signal (speech) with Compressor, Shelving Filter and Gate. The Silver Compressor, Silver Gate or another EQ are well-suited for these purposes.

Achieving the Best Analysis and Synthesis Signals

For good speech intelligibility, please keep these points in mind:

- The spectra of the analysis and synthesis signals should overlap almost completely. Low male voices with synthesis signals in the treble range do not work well.
- The synthesis signal must be constantly sustained, without breaks. The track should be played legato, as breaks in the synthesis signal will stop the vocoders output. Alternatively, the *Release* parameter of the synthesis signal (not to be confused with the *Release* time of the analysis section) can be set to a longer time. Nice effects can also be achieved by the use of a reverberation signal as a synthesis signal. Note that the two latter methods can lead to harmonic overlaps.
- Do not overdrive the Vocoder. This can happen easily, and distortion will occur. Lower the *Gain* of the compressor in the Side Chain track, to avoid this problem.
- Enunciate your speech clearly, if the recording is to be used as an analysis signal. Spoken words, with a relatively low pitch, work better than sung vocals—even if the creation of vocoder choirs is your goal! Pronounce consonants well.

Note: A nice example is the rolled *R* of “We are the Robots,” by Kraftwerk, a classic vocoder track. This pronunciation was specifically made to cater to the demands of the vocoder.

Feel free to do what you like when setting the Formant parameters. The intelligibility of speech is surprisingly little affected by shifting, stretching or compressing the formants. Even the number of frequency bands used has a minimal impact on the quality of intelligibility. The reason for this is our ability to intuitively differentiate the voices of children, women, and men, whose skulls and throats vary vastly by nature. Such physical differences cause variations in the formants which make up their voices. Our perception (recognition) of speech is based on an analysis of the *relationships* between these formants. In Logic’s Vocoder plug-ins, these stay intact, even when extreme formant settings are used.

The EVOC 20 PS combines a vocoder with a polyphonic synthesizer, and can be played in real-time. This chapter explains the use of the EVOC 20 TO and its parameters.

The EVOC 20 PS is a sophisticated vocoder, equipped with a polyphonic synthesis engine, and capable of receiving MIDI note input. This allows the EVOC 20 PS to be played, resulting in classic vocoder choir sounds, for example. Single notes and chords played with the polyphonic EVOC 20 PS will sing with the articulation of the *Analysis* audio source. During this process, the sonic characteristics and changes of the audio signal arriving at the Analysis input (the audio track selected as a side chain) are imposed onto the output signal of the integrated synthesizer (the *Synthesis* section).

The signal path of the EVOC 20 PS is shown in the block diagram on page 190.

Using the EVOC 20 PS

To make use of the EVOC 20 PS, you need to insert the EVOC 20 PS into an Audio Instrument channel instrument slot, and you also need to provide an audio signal as the Analysis audio source.

You can do this by following these steps.

- 1 Select or create a new audio track in the Arrange window.
- 2 Insert (or record) an audio file—we'll use a vocal part to start with—onto this audio track by Shift-clicking onto the Arrange area with the Pencil tool. This will launch a file browser, allowing you to select the desired audio file.
- 3 Click once on the name of the audio file you wish to use, and press *Open*. The file will be inserted at the selected location. It may be worthwhile setting up a cycle region in the Arrange window, allowing you to continually cycle the audio part. This will make experimentation easier.
- 4 Insert the EVOC 20 PS into the *Instrument* slot of an Audio Instrument channel.
- 5 In the gray area at the top of the Plug-in window, click-hold on the *Side Chain* pull-down menu, and select the audio track that contains the audio file.

- 6 Ensure that the corresponding Audio Instrument track is selected in the Arrange window. The EVOC 20 PS is now ready to accept incoming MIDI data, and has been assigned to see the output from the selected audio track via a Side Chain.
- 7 In the Track Mixer or Environment Audio layer (not the Arrange!), mute the audio track (the vocal track) serving as the Side Chain input.
- 8 Press the Play button on the Transport Bar, or use the *Play* key command (0 on the numeric keypad).
- 9 And now ... as the audio file is playing back, play your MIDI keyboard.
- 10 In the Track Mixer (or Environment Audio layer), adjust the volume levels of the EVOC 20 PS and the audio track used for the *Side Chain* to taste.
- 11 Do a little experimentation with the knobs, sliders, and other controls. Have fun, and feel free to insert further effect plug-ins on the channel or busses to further enhance the sound.

EVOC 20 PS Parameters



The EVOC 20 PS interface is divided into six main sections. These are the *Synthesis*, *Sidechain Analysis*, *Formant Filter*, *Modulation*, *U/V Detection*, and *Output* areas.

Synthesis Parameters

The EVOC 20 PS is equipped with a polyphonic synthesis engine. It is capable of accepting MIDI note input. The parameters of the *Synthesis* section are described below.

Mode Switches

These switches determine the number of voices used by the EVOC 20:



- When *Poly* is selected, the maximum number of voices is set via the numeric field alongside the *Poly* button. To change the value, click and hold with your mouse, and drag up or down to increase/decrease polyphony.

Note: Increasing the number of voices also increases processor overhead.

- When *Mono* or *Legato* is selected, the EVOC 20 PS is monophonic, and uses only one voice.
 - In *Legato* mode, *Glide* (see page 180) is only active on tied notes. Envelopes are not retrigged when tied notes are played (single trigger).
 - In *Mono* mode, *Glide* is always active and the envelopes are retrigged by every note played (multi trigger).
- The *Unison* button enables/disables Unison mode. In this mode, each EVOC 20 PS voice is doubled, which will cut polyphony in half (to a maximum of 8 voices) as indicated by the numeric *Voices* field. The doubled voices are detuned by the amount defined with the *Analog* parameter. (Also see the “Analog Tuning,” from page 180 onwards.)
- In *Unison-Mono* mode (both the *Unison* and *Mono* or *Legato* buttons are active), up to 16 voices can be stacked and played monophonically. In this mode, the *Voices* field displays the number of stacked voices that sound at the same time.

Warning: Stacking voices in *Unison-Mono* mode will increase the EVOC 20 PS’s output volume. To avoid overloading the Audio Instrument channel output, adjust the EVOC 20 PS’s *Level* slider accordingly.

Oscillator Section

The EVOC 20 PS is equipped with a two oscillator digital synthesizer which features a number of waveforms, and FM (Frequency Modulation). Further to these sound-generators in the *Synthesis* section is an independent *Noise* generator.

There are two oscillator modes.

- *Dual*: Where two oscillators make use of single-cycle digital waveforms to provide the *Synthesis* sound source(s).
- *FM*: A two operator FM engine, with Oscillator 1 as a sine wave carrier, and Oscillator 2 as the modulator. Oscillator 2 can use any of the single-cycle digital waveforms.

You can switch between *Dual* and *FM* modes by clicking on the *Dual* or *FM* label(s) to the top left of the section shown in the diagrams.



As you can see, there are some subtle differences between the two modes. We will look at the common parameters first, and will then look at the mode-specific options.

Wave 1 Parameters

The footages below the *Wave 1* label in both modes harks back to the days of pipe organs. The longer the pipe, the deeper the tone. This also applies to *Wave 1*. Simply click on the 16, 8 or 4 foot value to select the range in which Wave (oscillator) 1 functions. Your selection will be illuminated.

The numerical value beside the *Wave 1* label (shown as 41 in the diagrams) indicates the currently selected waveform type. The EVOC 20 PS features 50 waveforms with different sonic characteristics. To switch between them, simply click-hold on the numerical field and drag up or down. When the desired waveform number is visible, release the mouse button.

Note: When in FM mode, the waveform of *Wave 1* is a fixed sine wave. The waveform parameter of *Wave 1* does not have an effect in this mode.

Wave 2 Parameters

The numerical value beside the *Wave 2* label (shown as 41 in the diagrams) indicates the currently selected waveform type. The EVOC 20 PS features 50 single-cycle digital waveforms with different sonic characteristics. To switch between them, simply click-hold on the numerical field and drag your mouse up or down. When the desired waveform number is visible, release the mouse button.

Noise Parameters



The *Noise* generator provides a further sound source which can be used in addition to the two oscillators (*Wave 1* and *Wave 2*).

The *Level* knob controls the amount of noise added to the signals of the two oscillators, and the *Color* knob controls the timbre of the noise signal. When the *Color* knob is turned fully-left, the *Noise* generator creates a pure *white noise*. When turned fully-right, it generates *blue noise* (high-passed noise). White noise has always been used to create wind and rain sound effects. It has the same energy in each frequency interval. Blue noise sounds brighter, because its bass portion is suppressed by a high pass filter.

It is important to note that the *Noise* generator in the *Oscillator* section is independent of the *Noise* generator in the *U/V Detection* section. For further information on voiced and unvoiced signals, refer to “Analyzing Speech Signals,” from page 169 onwards.

Note: Turn *Color* full-right and *Level* a tiny bit up to achieve a more lively and fresh synthesis signal.

Dual Mode Parameters



The parameters specific to *Dual* mode are found in the *Wave 2* section, and the *Balance* slider to the right.

- The *Semi* parameter adjusts the tuning of the second oscillator (*Wave 2*) in semitone steps. Adjustment is made by using the mouse as a slider directly on the numerical field. Its range: ± 24 , or up/down two octaves.
- The *Detune* parameter fine-tunes *Wave 1* and *Wave 2* in cents. 100 cents equals a semitone step. Doing so will detune *Wave 1* in conjunction with *Wave 2* around the tuning zero point. The range is ± 50 cents, or up/down half a semitone. Adjustment is made by using the mouse as a slider directly on the numerical field.
- The *Balance* slider allows you to blend the two oscillator signals (*Wave 1* and *Wave 2*).

FM Mode Parameters

The parameters specific to the FM mode are found in the *Wave 2* section, and the *FM Int* slider to the right.



- The *Ratio c(arse)* parameter adjusts the coarse frequency ratio of the second oscillator in relation to the first oscillator. Adjustment is made by using the mouse as a slider directly on the numerical field. Range: 0–32.
- The *Ratio f(ine)* parameter adjusts the fine frequency ratio of the second oscillator in relation to the first oscillator. The range is 0–99. Adjustment is made by using the mouse as a slider directly on the numerical field.
- The *FM Int* slider determines the intensity of *Wave 1*'s sine wave modulation by *Wave 2*. Higher *FM Int* settings will result in a more complex waveform with more overtones.

When combined, the *Ratio* and *FM Int* parameters form the resulting complex FM waveform, thus defining the harmonic content.

Analog Tuning

The *Analog* tuning parameter simulates the instability of analog circuitry found in vintage vocoders. *Analog* alters the pitch of each note randomly. This behavior is much like that of polyphonic analog synthesizers. The *Analog* knob controls the intensity of this random detuning.



Tuning

The range of detuning is defined in the *Tune* window. Adjustments are made by using the mouse as a slider. The range is from 425 to 455 Hz.

Glide

The effect of this knob depends on the setting made in the *Bend Range* window. *Glide* determines the time it takes for the pitch to slide from one note to another (portamento). The maximum value is +5000 ms.



Bend Range

Bend Range determines the pitch bend modulation range in semitones. The range is ± 12 semitones.

Cutoff

This parameter sets the cutoff frequency of the lowpass filter. As you turn this knob to the left, an increasing number of high frequencies is filtered from the signal.

Resonance

Turning up *Resonance* leads to an emphasis of the frequency area surrounding the frequency defined by the *Cutoff* parameter. The filter is used for rough signal shaping, before the signal is articulated by the vocoding circuits.

Note: Set Cutoff as high as possible, and add a little *Resonance* to achieve a nice, brilliant high-end.

Envelope



The EVOC 20 PS features an *Attack/Release* envelope generator used for level control of the Oscillator section.

- The *Attack* parameter determines the amount of time that it takes for the Oscillators of the Synthesis section to reach their maximum level.
- The *Release* parameter determines the amount of time that it takes for the Oscillators of the Synthesis section to reach their minimum level.

Sidechain Analysis In Parameters

Attack

The *Attack* parameter determines how quickly each *envelope follower* (coupled to each Analysis filter band) reacts to rising signals. Longer *Attack* times result in a slower tracking response to transients of the Analysis input signal.

Note: A long *Attack* time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulate vocoder effect. Set *Attack* to the lowest possible value to enhance articulation.

Release

The *Release* parameter determines how quickly each *envelope follower* (coupled to each Analysis filter band) reacts to falling signals. Longer *Release* times cause the Analysis input signal transients to sustain longer at the vocoder's output.



Note: A long *Release* time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulate vocoder effect. Note that *Release* times that are too short result in rough, grainy vocoder sounds. *Release* values of around 8 to 10 ms have proven to be useful starting points.

Freeze

The *Freeze* button holds the current Analysis sound spectrum infinitely.

The frozen Analysis signal can capture a particular characteristic of the source signal which is then imposed as a complex sustained filter shape on the Synthesis section.

Using a spoken word pattern as a source, for example, the *Freeze* parameter could capture the attack or tail phase of an individual word within the pattern—the vowel *a*, for example.

With *Freeze* engaged, the Analysis filter bank ignores the input source until it is disengaged.

Another use of the *Freeze* parameter (which can be automated) could be to compensate for people's inability to sustain sung notes for a long period, without taking a breath. If the Synthesis signal needs to be sustained, when the Analysis source signal (a vocal part) isn't, *Freeze* can be used to lock the current formant levels (of a sung note)—even during gaps in the vocal part—between words in a vocal phrase.

Note: When the *Freeze* parameter is used, the *Attack* and *Release* parameters have no effect.

Bands

The *Bands* window determines the number of frequency bands used by the EVOC 20 PS. It ranges from 5 to 20.



The greater the number of bands, the more precisely the sound can be reshaped. As the number of bands is reduced, the source signal's frequency range is divided up into fewer bands—and the resulting sound will be formed with less precision by the Synthesis engine.

Note: Increasing the number of *Bands* also increases the processor overhead.

You may find that a good compromise between sonic precision—allowing incoming signals (speech and vocals, in particular) to remain intelligible—and resource usage, is around 10 to 15 bands.

Formant Filter Parameters

The Formant Filter Window

The Formant Filter window is divided into two sections by a horizontal line. The upper half applies to the Analysis section and the lower half to the Synthesis section. Changes made to the *High/Low* frequency parameters, the *Bands* parameter or the *Formant Stretch* and *Shift* parameters will result in visual changes to the Formant Filter window. This provides you with invaluable feedback on what is happening to the signal as it is routed through the two Formant Filter banks.



High/Low Frequency

The blue bar shown just beneath the EVOC 20 PS logo is a multi-part control which is used to determine the lowest and highest frequencies allowed to pass by the filter section. The length of the blue bar represents the frequency range for both analysis and synthesis. Frequencies of any audio input which fall outside these boundaries will be cut. All filter bands are distributed evenly across the range defined by the *High/Low Frequency* values.

- To adjust the *Low Frequency* value, simply click-hold on the silver slider to the left of the blue bar, and drag to the right (or left). The value range is 75–750 Hz.
- To adjust the *High Frequency* value, simply click-hold on the silver slider to the right of the blue bar, and drag to the left (or right). The value range is 800–8000 Hz.
- To adjust both sliders simultaneously, click on the area between the slider halves (directly on the blue bar) and drag to the left or right.
- You can make changes to the *High/Low Frequency* values directly by using your mouse as a slider on the numerical entries—270 and 7100 Hz in the diagram.

Lowest/Highest

These parameters can be found in the two small windows on either side of the Formant Filter window. These switches determine whether the lowest and highest filter bands act as bandpass filters (like all of the bands between them), or whether they act as lowpass/highpass filters, respectively. Click once on them to switch between the two curve shapes available.

- In the *Bandpass* setting, the frequencies below/above the lowest/highest bands are ignored for both analysis and synthesis.
- In the *Highpass* (or *Lowpass*) setting, all frequencies below the lowest (or above the highest) bands will be considered for analysis and synthesis.

Formant Stretch

This parameter alters the width and distribution of all bands in the Synthesis filter bank, extending or narrowing the frequency range defined by the blue bar (*Low/High Frequency* parameters) for the Synthesis filter bank.

With *Formant Stretch* set to 0, the width and distribution of the bands in the Synthesis filter bank is equal to the width of the bands in the Analysis filter bank. Low values narrow the width of each band, while high values widen them. The control range is from 0.5 to 2 (expressed as a ratio of the overall bandwidth).

Note: You can jump directly to a value of 1 by clicking on its number.

Formant Shift



Formant Shift moves the position of all bands in the Synthesis filter bank up and down. With *Formant Shift* set to 0, the position of the bands in the Synthesis filter bank is equal to the position of the bands in the Analysis filter bank. Positive values will move the bands up in frequency, while negative values will move them down in respect to the Analysis filter bank.

Note: You can jump directly to the values -0.5, -1, +0.5 and +1 by clicking on their numbers.

Note: When combined, *Formant Stretch* and *Formant Shift* alter the formant structure of the resulting vocoder sound, and can lead to some interesting timbre changes. As an example, using speech signals and tuning *Formant Shift* up results in Mickey Mouse effects.

Note: *Formant Stretch* and *Formant Shift* are also useful if the frequency spectrum of the Synthesis signal does not complement the frequency spectrum of the Analysis signal. You could create a synthesis signal in the high frequency range from an analysis signal that mainly modulates the sound in a lower frequency range, for example.

Resonance



Resonance is responsible for the basic sonic character of the vocoder: low settings give it a soft character, high settings will lead to a more snarling, sharp character. Increasing the *Resonance* value emphasizes the middle frequency of each frequency band.

The use of either, or both, of the *Formant Stretch* and *Formant Shift* parameters can result in the generation of unusual resonant frequencies—when high *Resonance* settings are used.

Modulation Parameters



The Modulation (*LFO*) area offers two LFOs to control the *Formant Shift* and *Pitch* parameters of the EVOC 20 PS. The LFOs can run free or synchronized to the song's tempo.

- *Pitch LFO*, on the left-hand side, controls *Pitch* modulation (Vibrato) of the built-in synthesizer's oscillators. It is hardwired to accept data from the modulation wheel of your MIDI keyboard (or corresponding MIDI data) to control modulation intensity.
- *Shift LFO* controls the *Formant Shift* parameter of the Synthesis filter bank to produce dynamic phasing-like effects.

Wave

These two switches allow the selection of the waveform type used by *Pitch LFO* and *Shift LFO*. A selection of Triangle, falling and rising Sawtooth, Square up and down around zero (bipolar, good for trills), Square up from zero (unipolar, good for changing between two definable pitches), a random stepped waveform (S&H), and a smoothed random waveform is available for each LFO.



Intensity/Int via Whl

The *Intensity* slider controls the amount of *Formant Shift* modulation by the *Shift LFO*.

The *Int via Whl* slider for the *Pitch LFO* features a multi-part slider. The intensity of LFO pitch modulation can be controlled by the modulation wheel of an attached MIDI keyboard. The upper half of the slider determines the intensity when the modulation wheel is set to its maximum value, and the lower half when set to its minimum value. By clicking and dragging in the area between the two slider segments, you can simultaneously move both.

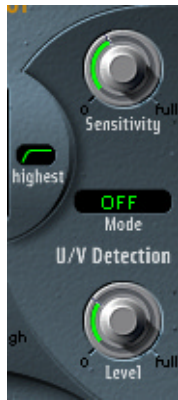
Rate Knobs

These knobs determine the speed of modulation. Values to the left of the center positions are synchronized with the sequencer's tempo and include bar values, triplet values and more. Values to the right of the center positions are non-synchronous, and are displayed in Hertz (cycles per second).

Note: The ability to use synchronous bar values could be used to perform a formant shift every four bars on a cycled one bar percussion part. Alternately, you could perform the same formant shift on every eighth note triplet within the same part. Either method can generate interesting results, and can lead to new ideas, or a new lease of life on old audio material.

Unvoiced/Voiced (U/V) Detection

Please refer to “Analyzing Speech Signals,” from page 169 onwards, for an explanation of the *U/V Detection* principle.



Speech intelligibility is highly dependent on high frequency content, as human hearing is reliant on these high frequencies to determine syllables within words. Bear this fact in mind when using the EVOC 20 PS, and take care with filter frequency settings in the Synthesis and Formant Filter sections.

To aid intelligibility, it may be worthwhile using equalization to boost particular frequencies in the mid to high frequency range, before processing the signal with the EVOC 20 PS. Please see the “Tips for Better Speech Intelligibility” on page 169 for further information.

Sensitivity

This parameter determines how responsive U/V detection is. By turning this knob to the right, more of the individual *unvoiced* portions of the input signal are recognized.

When high settings are used, the increased sensitivity to unvoiced signals can lead to the U/V source—determined by the *Mode* parameter—being used on the majority of the input signal, including *voiced* signals. Sonically, this results in a sound that resembles a radio signal which is breaking up, and contains a lot of static or noise.

Mode

Mode selects the sound source(s) that can be used to replace the *unvoiced* content of the input signal. Possible settings are *Off*, *Noise*, *Noise + Synth*, or *Blend*.



- *Noise*—uses noise alone for the unvoiced portions of the sound.
- *Noise + Synth*—uses noise and the synthesizer for the unvoiced portions of the sound,
- *Blend*—uses the Analysis signal after it has passed through a highpass filter, for the unvoiced portions of the sound. This filtered Analysis signal is then mixed with the EVOC 20 PS output signal. The *Sensitivity* parameter has no effect when this setting is used.

Level

The *Level* knob controls the volume of the signal (*Noise*, *Noise + Synth*, or *Blend*) used to replace the *unvoiced* content of the input signal.



Warning: Care should be taken with this control, particularly when a high *Sensitivity* value is used, to avoid internally overloading the EVOC 20 PS.

Output Parameters

Signal



This pull-down menu offers the choice of *Voc(oder)*, *Syn(thesis)* and *Ana(lysis)*. These options allow you to determine the signal that you wish to send to the EVOC 20 PS main outputs. To hear the vocoder effect, the *Signal* parameter should be set to *Voc*. The other two settings are useful for monitoring purposes.

Ensemble

The three *Ensemble* switches switch the ensemble effect(s) on or off. *Ensemble I* is a special chorus effect. *Ensemble II* is a variation, creating a fuller and richer sound by using a more complex modulation routine.

Level

The *Level* slider controls the volume of the EVOC 20 PS output signal.

Stereo Width

Stereo Width distributes the output signals of the Synthesis section's filter bands in the stereo field.

- At the *left* position, the output of all bands are centered.
- At the *centered* position, the output of all bands ascends from left to right.
- At the *right* position, the bands are output—alternately—to the left and right channels.

MIDI Controllers Received

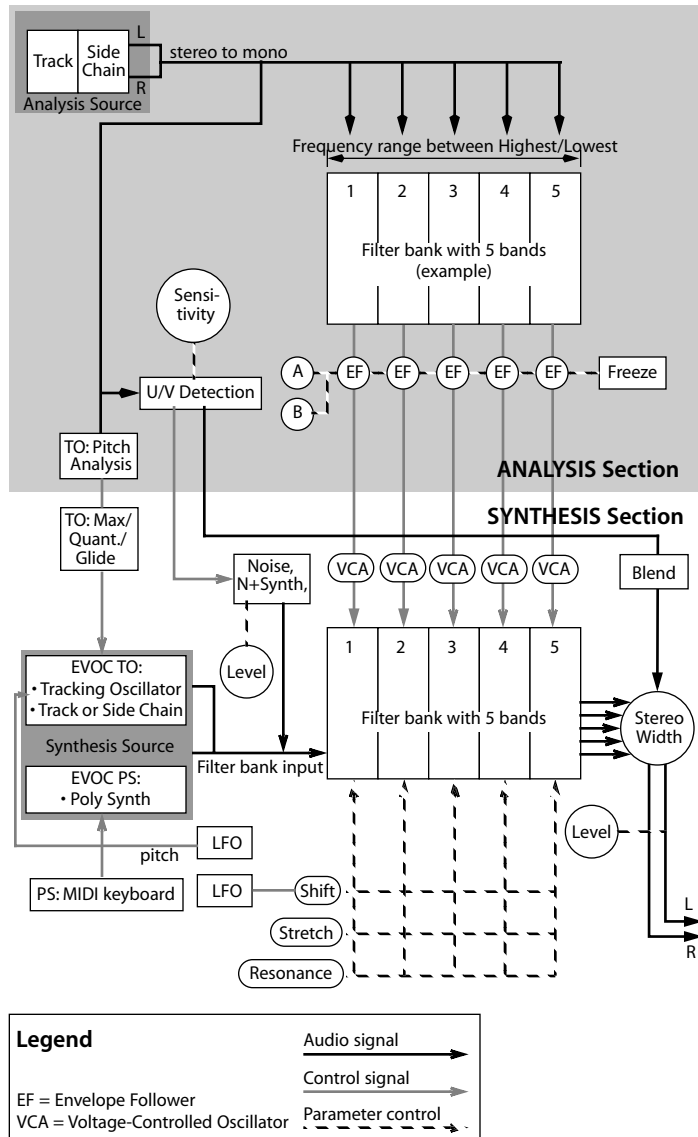
The following tables show the CC numbers used when the following MIDI preference is active: *Options > Settings > MIDI Options > (Version 4.x behavior)*.

Sidechain Analysis	Attack	CC #77
	Release	CC #78
	Freeze	CC #76
Keyboard	Mode	CC #89
	Voices	CC #83
	Unison	CC #108
Synthesis	Oscillator Mode	CC #88
	Osc 1 Octave	CC #90
	Osc 1 Waveform	CC #94
	Osc 2 Waveform	CC #95
	Osc 2 Semitone	CC #93
	Detune/Ratio Fine	CC #91
	Osc Mix/FM Intensity	CC #100
Noise	Level	CC #99
	Color	CC #98
Analog	Tune	CC #84
	Intensity	CC #85

Glide	Time	CC #87
	Bend Range	CC #86
Oscillator Filter	Cutoff	CC #96
	Resonance	CC #97
Osc. Envelope	Attack	CC #106
	Release	CC #107
Formant Filter	FF Low Freq	CC #65
	FF Hi Freq	CC #66
	Formant Shift	CC #67
		CC #71
	FF Resonance	CC #72
	FF Low/Bandpass Select	CC #73
	FF High/Bandpass Select	CC #74
LFO 1 (Shift)	LFO 1 Rate	CC #68
	LFO 1 Waveform Select	CC #69
	LFO 1 Intensity	CC #70
LFO 2 (Pitch)	LFO 1 Rate	CC #102
	LFO 1 Waveform Select	CC #103
	LFO 1 Low Intensity	CC #104
	LFO 1 High Intensity	CC #105
U/V Detection	Sensitivity	CC # 81
	Mode	CC # 80
	Level	CC # 82
Output	Signal Out Select	CC #109
	Ensemble	CC #112
	Level	CC #110
	Mono/Stereo Select	CC #111
	Stereo Width	CC #79

Block Diagram

This block diagram illustrates the signal path in the EVOC 20 TO and EVOC 20 PS.



The vocoder is over 50 years old. This chapter discusses its history.

You may be surprised you to learn that the Voder and Vocoder date back to 1939 and 1940, respectively.

Homer Dudley, a research physicist at Bell Laboratories, New Jersey (USA) developed the Voice Operated reCOrDER as a research machine. It was originally designed to test compression schemes for the secure transmission of voice signals over copper phone lines.

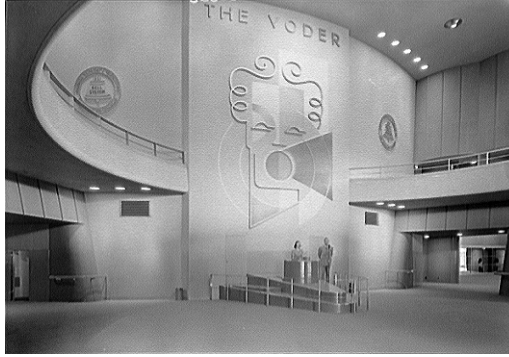
It was a composite device consisting of an analyzer and an artificial voice synthesizer. These were the:

- Parallel Bandpass Vocoder—a speech analyzer and resynthesizer, invented in 1940.
- The Voder speech synthesizer—a voice model played by a human operator, invented in 1939. This valve-driven machine had two keyboards, buttons to recreate consonants, a pedal for oscillator frequency control, and a wrist-bar to switch vowel sounds on and off.

The analyzer detected the energy levels of successive sound samples, measured over the entire audio frequency spectrum via a series of narrow band filters. The results of this analysis could be viewed graphically as functions of frequency against time.

The synthesizer reversed the process by scanning the data from the analyzer and supplying the results to a number of analytical filters, hooked up to a noise generator. This combination produced sounds.

The Voder was demonstrated at the 1939 World Fair, where it caused quite a stir:



In World War II, the Vocoder (now called VOice enCODER) proved to be of crucial importance, scrambling the transoceanic conversations between Winston Churchill and Franklin Delanore Roosevelt.

Werner Meyer-Eppler, the director of Phonetics at Bonn University, recognized the relevance of the machines to electronic music after Dudley visited the University in 1948. Meyer-Eppler used the vocoder as a basis for his future writings which, in turn, became the inspiration for the German “Elektronische Musik” movement.

In the 1950's, a handful of recordings ensued.

In 1960, the Siemens Synthesizer was developed in Munich. Among it's many oscillators and filters, it included a valve-based vocoding circuit.

In 1967, a company called Sylvania created a number of digital machines that used time-based analysis of input signals, rather than bandpass filter analysis.

In 1971, after studying Dudley's unit, Bob Moog and Wendy Carlos modified a number of synthesizer modules to create their own vocoder for the *Clockwork Orange* sound track.

Peter Zinovieff's London-based company “EMS” developed a standalone—and altogether more portable—vocoder. EMS are probably best known for the “Synthi AKS” and VCS3 synthesizers. The EMS Studio Vocoder was the world's first commercially available machine, released in 1976. It was later renamed the EMS 5000. Among it's users were Stevie Wonder and Kraftwerk. Stockhausen, the German “Elektronische Musik” pioneer, also used an EMS vocoder.

Sennheiser released the VMS 201 in 1977, and EMS released the EMS 2000, which was a cut-down version of it's older sibling.

1978 saw the beginning of mainstream vocoder use, riding on the back of popularity created through the music of Herbie Hancock, Kraftwerk, and a handful of other artists. Among the manufacturers who jumped into vocoder production at this time are: Synton/Bode, Electro-Harmonix, and Korg, with the VC-10.

In 1979, Roland released the VP 330 ensemble/vocoder keyboard.

The late 70's and early 80's were the heyday of the vocoder. Artists who used them included: ELO, Pink Floyd, Eurythmics, Tangerine Dream, Telex, David Bowie, Kate Bush, and many more.

On the production side, vocoders could (and can still) be picked up cheaply in the form of kits from electronics stores.

From 1980 through to the present, EMS in the UK, Synton in Holland and PAiA in the USA were, and remain, the main flyers of the vocoding flag.

In 1996, Doepfer in Germany and Music and More joined the vocoder-producing fraternity.

Throughout the 1990's, a number of standalone software-based vocoders have appeared.

As you can see, the history of the vocoder is long and varied, and we're sure that the vocoder will be with us for a while to come.

If you are new to synthesizers, you should read this chapter.

It covers important facts about the synthesizer and explains the difference between analog, digital and virtual analog synthesizers. Important synthesizer terms such as cutoff, resonance, envelope, and waveform are also introduced.

Analog and Subtractive

An analog synthesizer signal is an electrical signal, measured in volts. To give you a brief comparison with a technology you're probably familiar with, we'll look at speakers. The speaker "coils" move when the voltage—amplified by a power amplifier and output to the speaker—changes. When the voltage rises, the speaker coil moves forward. If the voltage falls, the speaker coil moves backwards.

In a digital synthesizer, the signal flow is digital. Binary *descriptions* of the signal (a string of zeros and ones) are fed from one algorithm to another. This is an important distinction to make. It is *not* the signal itself that is fed from a virtual oscillator to a virtual filter and so on.

A virtual analog synthesizer is a digital synthesizer which mimics the architecture, features, and peculiarities of an analog synthesizer. It includes the front panel with all controls, which provides direct access to all sound generation parameters.

Logic's ES1 is an example of a virtual analog synthesizer. Its virtual signal flow is as per that found in analog synthesizers. It includes some of the desirable idiosyncrasies of particular analog circuits—in cases where they tend to sound nice, such as high oscillator levels overdriving the filter. The ES1 also features a graphical control surface on your computer screen. Its signal processing (those "virtual" oscillators and so on) is performed by the central processing unit (CPU) of your computer.

Undesirable analog synthesizer phenomena, such as the habit of going completely out of tune, are not simulated by virtual analog synthesizers. You can, however, set the voices of the ES1 to randomly detune, adding “life” to the synthesizer’s sound. Unlike its analog counterparts, the ES1 is also; completely programmable (you can save sound settings), can be completely automated (you can record and playback fader movements), polyphonic (you can play up to 16 notes at the same time), multitimbral (you can play different sounds at the same time—on different Audio Instrument channels), and velocity sensitive.

These are important benefits, which overcome the limitations of old synthesizers. If you find it more inspirational to avoid the use of these features, you can always switch them off.

What Is Synthesis?

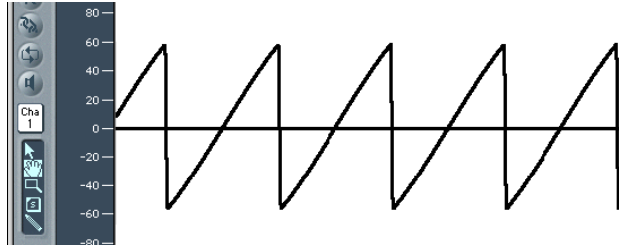
Before we start, synthesis in this context, is the (re)production of a sound which emulates, or synthesizes the sound of another instrument, a voice, helicopter, car, dog bark—in fact, any sound you can think of!

This synthetic reproduction of other sounds is what gives the synthesizer its name. Needless to say, synthesizers can also produce many sounds which would never occur in the “natural” world. This ability to generate sounds which cannot be created in any other way is what makes the synthesizer a unique musical tool. Its impact on modern music has been enormous, and will continue well into the future—although it is more likely to live on in “virtual” form, rather than as hardware.

Subtractive Synthesis

Subtractive synthesis is synthesis using filters. All analog and virtual analog synthesizers use subtractive synthesis to generate sound. In analog synthesizers, the audio signal of each voice is generated by the *oscillator*. The oscillator generates an alternating current, using a selection of waveforms which contain differing amounts of (more or fewer) harmonics. The fundamental (or root) frequency of the signal primarily determines the perceived pitch, its waveform is responsible for the basic sound color, and the amplitude (level) determines the perceived volume.

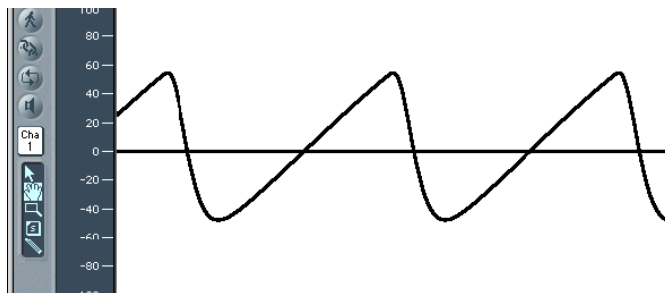
Cutoff and Resonance—illustrated with a sawtooth wave



This picture shows an overview of a sawtooth wave ($a = 220$ Hz); The filter is open, with cutoff set to its maximum, and with no resonance applied. The screenshot shows the output signal of Logic's ES1, routed to a monophonic Logic Output Object. The recording was performed with the Bounce function of this Audio Object, and is displayed in Logic's Sample Editor at a high zoom setting.

When Michelangelo was asked how he would manage to cut a lion out of a block of stone, he answered, "I just cut away everything that doesn't look like a lion." This, in essence, is how subtractive synthesis works: Just filter (cut away) those components of sound which should not sound—in other words, you subtract parts of the oscillator signal's spectrum. After being filtered, a brilliant sounding sawtooth wave becomes a smooth, warm sound without sharp treble. Analog and virtual analog synthesizers are not the only devices that make use of subtractive synthesis techniques. Samplers and sample players also do so, but use modules which play back digital recordings (Samples) in place of oscillators (that supply sawtooth and other waveforms).

The picture below shows a sawtooth wave with the filter half closed (24 dB/Fat). The effect of the filter is somewhat like a graphic equalizer, with a fader set to a given cutoff frequency (the highest frequency being fed through) pulled all the way down (full rejection), so that the highs are damped. With this setting, the edges of the sawtooth wave are rounded, making it resemble a sine wave.

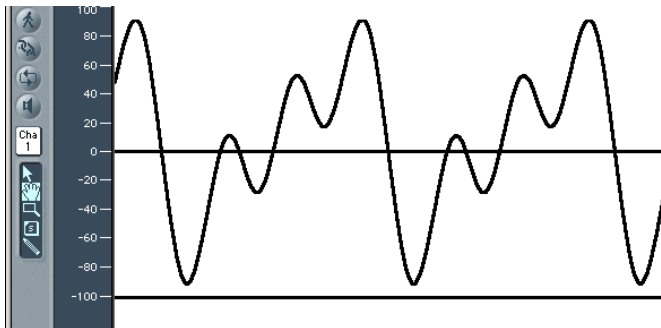


The wave length here is not really higher, but the zoom setting is.

Fourier Theorem and Harmonics

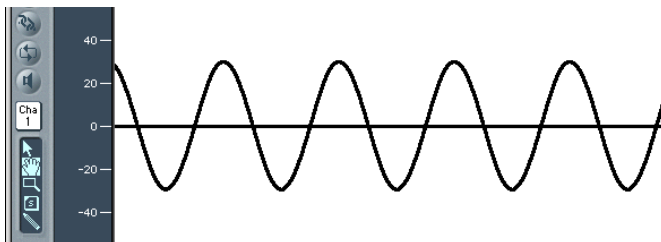
“Every periodic wave can be seen as the sum of sine waves with certain wave lengths and amplitudes, the wave lengths of which have harmonic relations (ratios of small numbers).” This is known as the Fourier theorem. Roughly translated into more musical terms, this means that any tone with a certain pitch can be regarded as a mix of sine partial tones. This is comprised of the basic fundamental tone and its harmonics (overtones). As an example: The basic oscillation (the first partial tone) is an “A” at 220 Hz. The second partial has double the frequency (440 Hz), the third one oscillates three times as fast (660 Hz), the next ones 4 and 5 times as fast, and so on.

You can emphasize the partials around the cutoff frequency by using high *resonance* values. The picture below shows a sawtooth wave with a high resonance setting, and the cutoff frequency set to the frequency of the third partial (660 Hz). This tone sounds a duodecima (an octave and a fifth) higher than the basic tone. It’s apparent that exactly three cycles of the strongly emphasized overtone fit into one cycle of the basic wave:



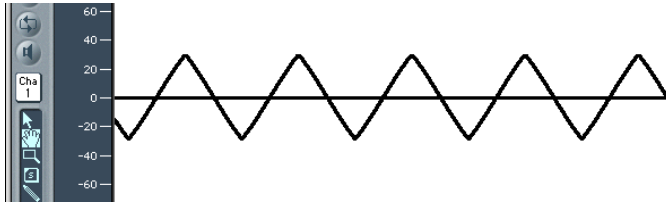
The effect of the resonating filter is comparable to a graphic equalizer with all faders higher than 660 Hz pulled all the way down, but with only 660 Hz (*Cutoff Frequency*) pushed to its maximum position (*resonance*). The faders for frequencies below 660 Hz remain in the middle (0 dB).

If you switch off the oscillator signal, a maximum resonance setting results in the self-oscillation of the filter. It will then generate a sine wave.



Other Oscillator Waveforms

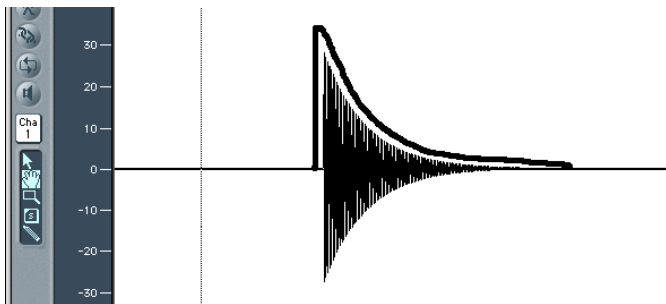
Waveforms (waves) are named sawtooth, square, pulse, or triangular because of their shape when displayed as an oscillogram (as in Logic's Sample Editor). This is the triangular wave:



The triangular wave has few harmonics—which is evident by the fact that it is shaped more like a sine than a sawtooth wave. This wave contains only odd harmonics—which means no octaves.

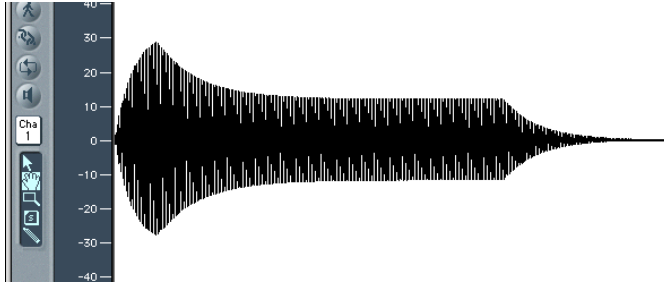
Envelopes

What does the term envelope mean in this context? In the image, you see an oscillogram of a percussive tone. It's easy to see how the level rises immediately to the top of its range, and how it decays. If you drew a line surrounding the upper half of the oscillogram, you could call it the envelope of the sound—a graphic displaying the level as a function of time. It's the job of the envelope generator to set the shape of the envelope.



The screenshot shows a recording of an ES1 sound created with these ADSR (*attack time*, *decay time*, *sustain level*, and *release time*) parameter settings: *attack* as short as possible, medium value for *decay*, zero for *sustain*, medium value for *release*.

When you strike a key, the envelope travels from zero to its maximum level in the *attack time*, falls from this maximum level to the *sustain level* in the *decay time*, and maintains the *sustain level* as long as you hold the key. When the key is released, the envelope falls from its *sustain level* to zero over the *release time*. The brass or string-like envelope of the following sound—the envelope itself is not shown in this graphic—has longer *attack* and *release times*, and a higher *sustain level*.



The envelope generator can also control the rise and fall of the cutoff frequency. You can also use envelope generators to modulate other parameters. In this context, modulation can be thought of as a remote control for a given parameter. There are more sources that can serve as a modulation source: the pitch (note number), velocity sensitivity or the modulation wheel, for example.

The 16-voice polyphonic EFM 1 is a powerful synthesizer based on frequency modulation.

It produces the typically rich bell and digital sounds that FM synthesis has become synonymous with.



Concept and Function

At the core of the EFM 1 engine, you'll find a multi-wave *Modulator* oscillator and a sine wave *Carrier* oscillator. The Modulator oscillator modulates the frequency of the Carrier oscillator within the audio range, thus producing new harmonics. These harmonics are known as sidebands.

The EFM 1 is divided into three areas. The top ring contains the global *Transpose*, *Tune*, *Randomize*, and *Unison* parameters.

The raised T-shaped FM engine in the center consists of the *Modulator*, *Carrier*, and *FM* controls, including the *Modulation Envelope* and *LFO*.

The bottom section of the ring houses the Output section, and features the *Sub Osc Level* and *Stereo Detune* parameters, plus the *Volume Envelope*, *Main Level* and *Velocity* controls.

Global Parameters

Transpose

The base pitch is set with the *Transpose* parameter. You can transpose the EFM 1 by ± 2 octaves.

Tune

Tune will fine-tune the EFM 1 ± 50 cents. A cent is 1/100th of a semitone.

Randomize

The Randomize facility generates new sounds with each mouse click. Click the *Randomize* button to create a new randomized sound, based on the Intensity value. Higher *Intensity* values—set in the numeric field by click-dragging up/down—will produce more random sounds. Experiment with values below 10% for small variations of a given sound.

Unison

Clicking on the *Unison* button will layer two complete EFM 1 voices, making the EFM 1 sound larger and fatter. In Unison mode, the EFM 1 can be played with 8-voice polyphony.

Voices

The number of simultaneously playable voices (polyphony) is determined by the *Voices* parameter. Available values are: *Mono* (one voice), *Legato* (one voice) and 2–16 voices. In the monophonic *Legato* mode, playing overlapping notes will not retrigger the EFM 1 envelopes.

Glide

Glide is used to introduce a continuous pitch bend between two consecutively played notes. The *Glide* value (in ms) determines the time it takes for the pitch to travel from the last played note to the next. *Glide* can be used in both of the monophonic *Mono* and *Legato* Voices modes, as well as with the polyphonic Voices settings (2–16).

FM Parameters

FM (Intensity)

The Modulator oscillator modulates the Carrier frequency, resulting in newly generated sidebands that add new overtones. Turning up the *FM (Intensity)* control (the large dial in the center) produces increasing numbers of overtones—and the sound becomes brighter. The FM (Intensity) parameter is sometimes called the FM Index.

Note: Although the technology behind it is very different, you could compare the FM (Intensity) parameter to the Filter Cutoff parameter of an analog synthesizer.

Modulation Env(elope)

To control the FM (Intensity) parameter dynamically, the EFM 1 provides a dedicated ADSR (FM) Modulation Envelope, consisting of four sliders: *A* (*Attack time*), *D* (*Decay time*), *S* (*Sustain level*) and *R* (*Release time*). The envelope is triggered every time a MIDI note is received. The *Attack* slider sets the time needed to reach the maximum envelope level. The *Decay* slider sets the time needed to reach the Sustain level (determined by the Sustain slider). The *Sustain* level is held until the MIDI note is released. The *Release* slider sets the time needed to reach a level of zero, after the MIDI note has been released.

FM Depth

The strength, or impact, of the Modulation Envelope on the FM intensity is determined by the *FM Depth* control.

Turning the FM Depth control clockwise increases the effect of the Modulation Envelope. Turning the FM Depth control counter clockwise inverts the effect of the Modulation Envelope, meaning that the envelope slopes down during the Attack phase, and slopes up during the Decay and Release time phases.

In the center (0) position, the envelope does not have an effect. You can easily center the FM Depth dial by clicking on the 0.

Modulator Pitch

The impact of the Modulation Envelope on the pitch of the Modulator oscillator is determined by the *Modulator Pitch* control.

Turning the Modulator Pitch control clockwise increases the effect of the Modulation Envelope. Turning the Modulator Pitch control counter-clockwise inverts the effect of the Modulation Envelope, meaning that the envelope slopes down during the Attack phase, and slopes up during the Decay and Release time phases.

In the center (0) position, the envelope does not have an effect. You can easily center the Modulator Pitch dial by clicking on the 0 button.

LFO

The LFO (Low Frequency Oscillator) serves as a cyclic modulation source for FM Intensity or Vibrato. Turning the *LFO* control clockwise increases the effect of the LFO on FM Intensity. Turning it counter clockwise introduces a vibrato.

In the center (0) position the LFO does not have an effect. You can easily center the LFO dial by clicking on the 0.

Rate

The speed/rate of the LFO cycles is set with the *Rate* parameter.

Modulator and Carrier

Harmonic

In FM synthesis, the basic overtone structure is determined by the tuning relationship of the Modulator and Carrier. This is often expressed as a tuning ratio. In the EFM 1, this ratio is achieved with the *Modulator* and *Carrier Harmonic* controls. Additional tuning control is provided by the *Fine (Tune)* parameters.

You can tune the Modulator and Carrier to any of the first 32 harmonics. The tuning relationship (or ratio) greatly changes the base sound of the EFM 1, and is best set by ear.

As a rule of thumb: even tuning ratios tend to sound more harmonic or musical, while odd ratios produce more inharmonic overtones—which are great for bell and metallic sounds.

As an example, the Modulator and Carrier set to the First Harmonic (a 1:1 ratio) will produce a sawtooth-like sound. If the Modulator is set to the Second Harmonic, and the Carrier to the First Harmonic (a 2:1 ratio), the tone produced will sound similar to a square wave. In this respect, the tuning ratio is somewhat similar to the waveform selector of an analog synthesizer.

The *Harmonic* dial of the EFM 1 Carrier can be set to a value of zero. This, in effect, produces a DC (Direct Current) signal. In this scenario, the Carrier actually acts as a wave shaper.

Fine

Fine tune adjusts the tuning in-between two adjacent harmonics (as determined by the Harmonic control). The range of this control is ± 0.5 harmonic. Dependent on the amount of detuning, this will create either a subtle “beating” of the timbre or—if high detuning amounts are used—adds new harmonic and inharmonic overtones.

In the center (0) position *Fine* tune does not have an effect. You can easily center the *Fine* tune control by clicking on the 0.

Fixed Carrier Button

This button allows you to disconnect the carrier frequency from keyboard, pitchbend, and LFO modulations.

Modulator Wave

In classic FM synthesis, sine waves are used as Modulator and Carrier waveforms. To extend its sonic capabilities, the EFM 1 Modulator provides a number of additional digital waveforms.

When turned completely counter clockwise the Modulator produces a sine wave. Turning the *Wave* parameter clockwise will step/fade through a series of complex digital waveforms. These digital waveforms add a new level of harmonic richness to the resulting FM sounds.

The Output Section

Sub Osc Level

For added bass response, the EFM 1 features a sine wave sub oscillator. This operates one octave below the FM engine (as determined by the *Transpose* parameter). Turning up the *Sub Osc Level* control mixes the sub oscillator sine wave with the EFM 1's FM engine output.

Vol(ume) Envelope

The *Volume Envelope* shapes the overall volume contour. The Volume Envelope consists of four sliders: *Attack time*, *Decay time*, *Sustain level*, and *Release time*. The Volume Envelope is triggered every time a MIDI note is received. The *Attack* slider defines the time needed to reach the maximum volume level. The *Decay* slider sets the time needed to reach the Sustain level (as determined by the Sustain slider). The *Sustain* level is held until the MIDI note is released. The *Release* slider controls the time needed to reach a volume level of zero, after the MIDI note has been released.

Stereo Detune

Stereo Detune adds a rich and diverse chorus-like effect to the sound of the EFM 1. This is achieved by doubling the EFM 1 voice with a detuned second FM engine. The amount of detune is adjusted using the Stereo Detune dial. A wide stereo effect is also added, increasing the “space” and “width” of your sound.

Velocity

The EFM 1 is able to respond to MIDI velocity, and reacts with dynamic sound and volume changes—harder playing will result in a brighter and louder sound. The sensitivity of the EFM 1 in response to incoming velocity information is determined by the *Velocity* parameter.

Set the Velocity control all the way to the left (counter-clockwise) if you don't want the EFM 1 to respond to velocity. Turning the control clockwise will increase velocity sensitivity, and with it, dynamic changes to the sound that the EFM 1 is able to produce.

Main Level

The *Main Level* control adjusts the overall output level of the EFM 1. Turning it clockwise makes the EFM 1 output louder. Turning it counter clock-wise will decrease the output level.

Pitch Bend, Modulation Wheel, Aftertouch

The EFM 1 responds to pitch bend, modulation wheel and aftertouch controller data. Pitch bend is hardwired to pitch. The modulation wheel introduces vibrato while aftertouch offers control over FM intensity.

Randomize

The *Randomize* facility generates new sounds with each mouse click. Click the *Randomize* button to create a new randomized sound, based on the *Intensity* value. Higher *Intensity* values will produce more random sounds. Experiment with values below 10% for small variations of a given sound.

This chapter introduces you to Logic's ES M synthesizer.

The monophonic ES M (ES Mono) is a good starting point if you're looking for bass sounds that punch through your mix.



Parameters of the ES M

8, 16, 32

The 8, 16, and 32 buttons set the ES M's octave transposition.

Glide

The ES M permanently works in a fingered portamento mode, with notes played in a legato style resulting in a glide (portamento) from pitch to pitch. The speed of the glide is set with the *Glide* parameter. At a value of 0, no glide effect occurs.

Mix

Mix crossfades between a sawtooth wave and a 50% rectangular wave, which is heard one octave lower.

Cutoff

This parameter sets the cutoff frequency of the resonance-capable dynamic lowpass filter. Its slope is 24 dB/Octave.

Resonance

This parameter sets the resonance of the dynamic lowpass filter. Increasing the *Resonance* value results in a rejection of bass (low frequency energy) when using low pass filters. The ES M compensates for this side-effect internally, resulting in a more bassy sound.

Int

The ES M features two very simple envelope generators with a single Decay parameter. *Int* enables modulation of the cutoff frequency by the filter envelope.

Decay (Filter)

This parameter sets the decay time of the filter envelope. It is only effective if *Int* is not set to 0.

Velo (Filter)

Velo determines the velocity sensitivity of the filter envelope. This parameter is only effective if *Int* is not set to 0.

Decay (Volume)

This parameter sets the decay time of the dynamic stage. The attack, release, and sustain times of the synthesizer are internally set to 0.

Velo (Volume)

This parameter determines the velocity sensitivity of the dynamic stage.

Vol

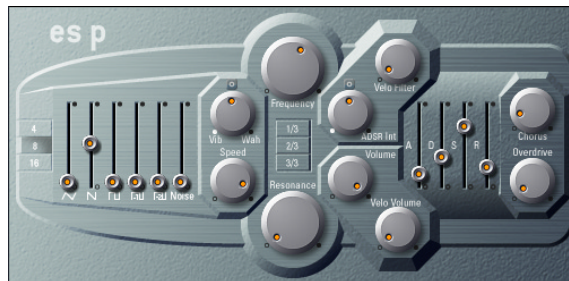
This parameter sets the master volume of the ES M.

Overdrive

This parameter sets the overdrive/distortion level for the ES M output. Caution: The overdrive effect significantly increases the output level.

This chapter introduces you to Logic's eight-voice polyphonic ES P (ES Poly) synthesizer.

Functionally, (despite its velocity sensitivity) this flexible synthesizer is somewhat reminiscent of the affordable polyphonic synthesizers produced by the leading Japanese manufacturers in the 1980s: Its design is easy to understand, it is capable of producing lots of useful musical sounds, and you may be hard-pressed to make sounds with it that can't be used in at least some musical style. The creation of classic analog synthesizer brass sounds are just one of its many strengths.



Parameters of the ES P 8, 16, 32

The 8, 16, and 32 buttons determine the ES P's octave transposition.

Waveform Faders

The faders on the left side of the panel allow you to mix several waveforms, output by the oscillators of the ES P. In addition to triangular, sawtooth, and rectangular waves, the rectangular waves of two sub-oscillators are also available. One of these is one octave lower than the main oscillators, and the other, two octaves lower. The pulse width of all rectangular waves is 50%. The right-most fader adds white noise to the mix. This is the raw material for classic synthesizer sound effects, such as ocean waves, wind, and helicopters.

Vib/Wah

The ES P features an LFO which can either modulate the frequency of the oscillators (resulting in a vibrato), or the cutoff frequency of the dynamic low pass filter (resulting in a wah wah effect). Turn the control to the left in order to set a vibrato, or to the right to cyclically modulate the filter.

Speed

Speed controls the rate of the oscillator frequency or cutoff frequency modulation.

Frequency

This parameter set the cutoff frequency of the resonance-capable dynamic low pass filter.

Resonance

This parameter sets the resonance of the dynamic lowpass filter. Increasing the *Resonance* value results in a rejection of bass (low frequency energy) when using low pass filters. The ES M compensates for this side-effect internally, resulting in a more bassy sound.

1/3, 2/3, 3/3

The cutoff frequency can be modulated by MIDI note number (keyboard position); you may know this parameter as Keyboard Follow on other synthesizers. You have the choice of: no modulation, one third, two thirds, or full keyboard follow (3/3). When set to 3/3, the relative harmonic content of each note is the same, independent of its pitch.

ADSR Int

The ES P features one ADSR envelope generator per voice. *ADSR Int* sets the amount of cutoff frequency modulation by the ADSR envelope generator.

Velo Filter

The cutoff frequency modulation by the ADSR envelope generator is velocity sensitive. The amount of velocity sensitivity is set by this parameter.

Volume

This parameter sets the master volume of the ES P.

Velo Volume

This parameter determines the amount of velocity sensitivity, with each note being louder if struck more firmly.

A

The *A* slider determines the attack time of the envelope generator.

D

The *D* slider determines the decay time of the envelope generator.

S

The *S* slider determines the sustain level of the envelope generator.

R

The *R* slider determines the release time of the envelope generator.

Chorus

This parameter sets the intensity of the integrated chorus effect.

Overdrive

This parameter sets the overdrive/distortion level of the ES P output. Caution: The overdrive effect significantly increases the output level.

This chapter introduces Logic's eight-voice polyphonic ES E synthesizer.

The ES E (ES Ensemble) is designed for pad and ensemble sounds. It is great for adding atmospheric sounds to your music.



Parameters of the ES E

4, 8, 16

The 4, 8, and 16 buttons determine the ES E's octave transposition.

Wave

The left-most setting of the *Wave* parameter causes the oscillators to output sawtooth signals, which can be modulated in frequency by the integrated LFO. Across the remaining range, the oscillators output pulse waves, with the average pulse width being defined by the *Wave* parameter.

Vib/PWM

If *Wave* is set to sawtooth, this parameter defines the amount of frequency modulation, resulting in a vibrato or siren effect, depending on LFO speed and intensity. If *Wave* has been set to a pulse wave, this parameter controls the amount of pulse width modulation (PWM). When the pulse width becomes very narrow, the sound sounds like it is being interrupted. Given this potential artefact, set the PWM intensity with care, and select the *Wave* parameter's 12 o'clock-position (50% rectangular) for the pulse width, if you want to achieve the maximum modulation range.

Speed

Speed controls the frequency of the pitch (sawtooth) or pulse width modulation.

Cutoff

This parameter sets the cutoff frequency of the resonance-capable dynamic lowpass filter.

Resonance

This parameter sets the resonance of the ES E's dynamic lowpass filter.

AR Int

The ES E features one simple envelope generator per voice. It features an *Attack* and a *Release* parameter. *AR Int*, defines the amount of cutoff frequency modulation applied by the envelope generator.

Velo Filter

This parameter sets the velocity sensitivity of the cutoff frequency modulation applied by the envelope generator. This parameter is only effective if *AR Int* is not set to 0.

Attack

This parameter sets the attack time of the envelope generator.

Release

This parameter sets the release time of the envelope generator.

Velo Volume

This parameter determines the amount of velocity sensitivity, with each note being louder if struck more firmly.

Volume

This parameter sets the master volume of the ES E.

Chorus/Ensemble

The ES E features a chorus/ensemble effect, with three switchable variations (plus off).

This chapter introduces Logic's virtual analog ES1 synthesizer.

The ES1's flexible tone generation system and interesting modulation options place an entire palette of analog sounds at your disposal: punchy basses, atmospheric pads, biting leads, and sharp percussion.

Parameters of the ES1



2', 4', 8', 16', 32'

These footage values allow you to switch the pitch in octaves. 32 feet is the lowest, and 2 feet, the highest setting. The origin of the term feet to measure octaves, comes from the measurements of organ pipe lengths.



Wave

Wave allows you to select the waveform of the oscillator, which is responsible for the basic tone color. You can freely set any pulse width in-between the square wave and pulse wave symbols. The pulse width can also be modulated in the modulation section (see the “Router” section, on page 219). Modulating the pulse width with a slowly cycling LFO, for example, allows periodically mutating, fat bass sounds.

Sub

The sub oscillator delivers square waves (one and two octaves below the frequency of the main oscillator), as well as a pulse wave (two octaves below the frequency of the main oscillator). In addition to pure square waves, the waveform switch allows selections between different mixes, and phase relationships of these waves, resulting in different sounds. You can also use white noise, or switch the sub oscillator *OFF*. You can feed a Side Chain signal (from any track!) into the synth’s filter (select EXT). You can select the Side Chain source track from the *Side Chain* panel in the gray area at the top of the Plug-in window.

Mix

This slider defines the mix relationship between the main and sub oscillator signals. When the sub oscillator wave is switched to *OFF*, its output is completely removed from the mix. As a tip, high resonance values allow the filter to self-oscillate, which can be useful if you want to use the filter like an oscillator.

Filter Parameters

Drive

This is an input level control for the lowpass filter, which allows you to overdrive the filter. Its use changes the behavior of the *Resonance* parameter, and the waveform may sound distorted.

Cutoff and Resonance

The *Cutoff* parameter controls the cutoff frequency of the ES1's lowpass filter.

Resonance emphasizes the portions of the signal which surround the frequency defined by the *Cutoff* parameter. This emphasis can be set so intensively, that the filter begins to oscillate by itself. When driven to self-oscillation, the filter outputs a sine oscillation (a sine wave). If *key* is set to 1, you can play the filter chromatically from a MIDI keyboard.

There is another way to set the cutoff frequency: click-hold on the word *Filter* (surrounded by the slope selectors), and move the mouse vertically to adjust the cutoff frequency. Moving the mouse horizontally adjusts the resonance.

Slope buttons

The lowpass filter offers four different slopes of band rejection above the cutoff frequency.

- The *24 dB classic* setting mimics the behavior of a Moog-style filter: turning up the resonance results in a reduction of the low-end of the signal.
- The *24 dB fat* setting compensates for this reduction in low frequency content. Turning up resonance doesn't diminish the low-end of the signal, and thus resembles an Oberheim-style filter.
- *18 dB* tends to resemble the filter sound of Roland's TB-303.
- The *12 dB* setting provides a soft, smooth sound which is reminiscent of the early Oberheim SEM.

Key

This parameter controls the amount of cutoff frequency modulation by the keyboard pitch (note number). If *Key* is set to zero, the cutoff frequency won't change, no matter which key you strike. This makes the lower notes sound relatively brighter than the higher ones. If *Key* is set to maximum, the filter follows the pitch, resulting in a constant relationship between cutoff frequency and pitch.

ADSR Via Vel

The main envelope generator (ADSR) modulates the cutoff frequency over the duration of a note. The intensity of this modulation can be set to positive or negative values, and can respond to velocity information. If you play pianissimo (*Velocity* = 1), the modulation will take place as indicated by the lower arrow. If you strike with the hardest fortissimo (*Velocity* = 127), the modulation will take place as indicated by the upper arrow. The blue bar between the arrows shows the dynamics of this modulation. You can adjust the modulation range and intensity simultaneously by grabbing the bar and moving both arrows at once. Note that as you do so, they retain their relative distance from one another.

Level Via Vel

The upper arrow works like a main volume control for the synthesizer. The greater the distance from the lower arrow (indicated by the blue bars), the more the volume is affected by incoming velocity messages. The lower arrow indicates the level when you play pianissimo (velocity = 1). You can adjust the modulation range and intensity simultaneously by grabbing the bar and moving both arrows at once. Note that as you do so, they retain their relative distance from one another. In order to retain the maximum possible resolution for velocity sensitivity, even when set to a low volume, the ES1 has an additional *Out Level* parameter—available in the Controls view.

Amplifier Envelope Selector

The *AGateR*, *ADSR*, and *GateR* switches define which of the ADSR envelope generator controls have an effect on the amplifier envelope.

- *AGateR* activates the attack and release time controls, but allows the level to remain constant between the time the peak level is reached, and the release of the key—regardless of the decay and sustain settings.
- *ADSR* activates all controls for the amplifier section.
- *GateR* sets the attack time for the amplifier section to zero, with only the release control still having an effect on the envelope level.

All ADSR parameters will always remain active for the filter (ADSR via Vel). *A* stands for attack time, *R* for release time, while *Gate* is the name of a control signal used in analog synthesizers, which tells an envelope generator that a key is pressed. As long as an analog synth key is pressed, the gate signal maintains a constant voltage. Used as a modulation source in the voltage controlled amplifier (instead of the envelope itself), it creates an organ type envelope without any attack, decay or release.



Glide

The *Glide* parameter defines the amount of (portamento) time applied to each triggered note. The *Glide* trigger behavior depends on the value set in *Voices* (see “Voices” on page 221). A value of 0 disables the *Glide* function.

LFO Waveform

The LFO offers several waveforms: triangle, ascending and descending sawtooth, square wave, sample & hold (random), and a lagged, smoothly changing random wave. You can also assign a Side Chain signal (any audio track) as a modulation source (*EXT*). Select the Side Chain source track via the Side Chain pull-down menu in the gray area at the top of the Plug-in window.

Rate

This defines the speed (frequency) of modulation. If you set values to the left of zero, the LFO phase is locked to the tempo of the song—with phase lengths adjustable between 1/96 bar and 32 bars. If you select values to the right of zero, it will run freely. When set to zero, the LFO will output at a constant (and full) level, allowing you to use the modulation wheel to modulate, say, the pulse width: moving the mod wheel changes the pulse width in accordance with the “Int via Whl” setting, without introducing LFO modulation.

Int Via Whl

The upper arrow defines the intensity of the LFO modulation if the modulation wheel (MIDI Controller 1) is set to its maximum value. The lower arrow defines the amount of LFO modulation if the modulation wheel is set to zero. The distance between the arrows (indicated by a green bar) indicates the range of your keyboard’s modulation wheel. You can simultaneously adjust the modulation range and intensity by grabbing the bar and moving both arrows at once. Note that as you do so, they retain their relative distance from one another.

Router

The router defines the modulation target for LFO modulation and the modulation envelope. Only one target can be set for the LFO, and another one can be set for the modulation envelope. You can modulate:

- the pitch (frequency) of the oscillator
- the pulse width of the pulse wave
- the mix between the main and sub oscillators
- the cutoff frequency of the filter
- the resonance of the filter
- the main volume (the amplifier)

The following two targets are only available for the modulation envelope:

- Filter FM (the amount of cutoff frequency modulation by the triangle wave of the oscillator)

The modulation characteristics are non-linear. Thus, you can achieve a pseudo distortion of existing sounds, or, if only the self-oscillation of the resonating filter is audible, create metallic, FM style sounds. Switch *Sub* to *off* and *Mix* to *Sub* in order to do so.

- LFO Amp (the overall amount of LFO modulation)

As one application, you can create a delayed vibrato by modulating the LFO modulation intensity if the LFO router is set to pitch. The shape of the modulation envelope will control the intensity of the vibrato. Select an attack style setting (High value for *form*).

Int Via Vel

The upper arrow controls the upper modulation intensity setting for the modulation envelope, if you strike a key with the hardest fortissimo (velocity = 127). The lower arrow controls the lower modulation intensity setting for the modulation envelope, if you strike a key with the softest pianissimo (Velocity = 1). The green bar between the arrows displays the impact of velocity sensitivity on the (intensity of the) modulation envelope. You can simultaneously adjust the modulation range and intensity by grabbing the bar and moving both arrows at once. Note that as you do so, they retain their relative distance from one another.

Mod Envelope

The modulation envelope itself only has one parameter. You can set a percussive type of decay envelope (low values), or attack type envelopes (high values). A full setting of the modulation envelope delivers a constant, full level. This is useful if you want a parameter to be modulated solely by velocity: select a modulation destination, (LFO Amplitude, for example), set the modulation envelope to full, and adjust *Int via Vel* as needed, in order to obtain a velocity sensitive, yet non time-variable amount of LFO Amplitude modulation.

ADSR

The ADSR envelope affects the filter (ADSR via Vel) and the amplifier (if set to ADSR). The parameters are attack time (A), decay time (D), sustain level (S) and release time (R). If you're unfamiliar with these parameters: set the *amplifier* to ADSR, the *Cutoff* to a low value, *Resonance* to a high value, and move both of the "ADSR via Vel" arrows upwards, in order to check out what these parameters do.

Tune

Tune sets the pitch of the ES1.

Analog

Analog slightly alters the pitch of each note, and the cutoff frequency, in a random manner. Similar to polyphonic analog synthesizers, *Analog* values higher than zero allow the oscillators of all triggered voices to cycle freely. Note that if *Analog* is set to a value of zero, the oscillator cycle start points of all triggered voices are synchronized. This may be useful for percussive sounds, when looking to achieve a sharper attack characteristic. For a warm, analog type of sound, the *Analog*-Parameter should be set to higher values, thereby allowing subtle variations for each triggered voice.

Bender Range

Bender Range selects the sensitivity of the pitch bender in semitones.

Out Level

Out Level is the master volume control for the ES1 synthesizer.

Voices

The number displayed is the maximum number of notes which can be played simultaneously. Each ES1 instance offers a maximum of 16 voice polyphony. Fewer played voices require less CPU power.

If you set *Voices* to legato, the ES1 will behave like a monophonic synthesizer with single trigger and fingered portamento engaged. This means that if you play legato, a portamento corresponding to the *Glide* setting will occur, but if you release each key before you press a new one, there will be no portamento at all. The envelope will not be triggered by the new note. This allows for pitch bending effects without touching the pitch bender. Don't forget to select a higher *Glide* value when using the *Legato* setting.

Chorus

The ES1 offers classic stereo Chorus/Ensemble effects. There are four possible settings: *Off*, *C1*, *C2*, and *Ens*.

Off deactivates the Chorus. *C1* and *C2* are typical Chorus effects. *C2* is variation of *C1* and is characterized by a stronger modulation. In comparison, the Ensemble effect (*Ens*) employs a more complex modulation routing, creating a fuller and richer sound.

The virtual-analog ES2 synthesizer offers an exciting and extensive array of features and functions.

This chapter covers all details of the ES2's powerful tone generation system. A brief summary is followed by an in-depth description of its parameters. At the chapter's end you will find tutorials, where well-known sound designers explain how to program ES2 sounds.

Concept and Function

The ES2 is one of the most versatile virtual-analog synthesizers ever designed. The features and sonic potential of many well-respected, rare, and vintage analog synthesizers pale in comparison to the power offered by the ES2.

Don't be fooled by the compact and easy-to-understand user interface of the ES2. Its lack of additional menu items, windows, and lengthy parameter lists may give you the impression that its sonic power might not be that immense. Belying this simple appearance, the ES2 offers facilities that exceed those found in most of the legends of analog synthesizer history.

So, for the synthesizer connoisseur, here's the "brief" list of key features to whet your appetite:

The ES2 features an advanced modulation matrix (known as the *Router*) in addition to a number of "hard-wired" modulation routings. Basically, the concept of combining any modulation *source* with any modulation *target*, is almost as old as the synthesizer itself. Most important to the concept is a huge set of modulation *targets*, *sources*, a sufficient number of modulation *channels* and modulation *processes* which can be inserted into modulation *channels*. The ES2—featuring 10 modulation channels—represents the new standard when it comes to matrix modulation. You can modulate parameters such as filter resonance or the intensity of cutoff frequency modulation by Oscillator 1.

The ES2's three Oscillator concept is reminiscent of the Minimoog and EMS VCS 3. The oscillators can be synchronized and ring-modulated. Pulswidth modulation is also possible. Oscillator 1 can be modulated in frequency by Oscillator 2, and is capable of producing FM synth sounds.

Further to the classic, standard waveforms, the ES2's Oscillators also feature 155 single-cycle waveforms, known as DigiWaves. Each has a totally different sonic color. The *DigiWave* Parameter can be modulated, allowing dynamic cross-fades between the waves. Intense use of such modulations will scroll through several waves, resulting in sounds somewhat like those of wavetable synthesizers.

Two dynamic multi-mode filters, with a number of selectable slopes, guarantee the fattest analog sounds. The filters can be mixed freely in parallel and series modes. The filters can be overdriven by the *Drive* control.

Distortion, *Chorus*, *Phaser*, and *Flanger* effects are included. The unique *Unison* mode delivers a density of sound normally only associated with big analog synthesizers, including classic machines such as the Roland Jupiter 8, the SCI Prophet V and Oberheim OB 8, amongst others. The great thing about the ES2's *Unison* mode is that it can be used in *both* monophonic and polyphonic modes. The unison voices are intelligently spread across the entire stereo spectrum, but you can still modulate the panorama position of the voices with any modulation source. This latter facility is unprecedented.

As with the classic Yamaha CS Series analog synthesizers, a sine wave derived from Oscillator 1 can be mixed directly into the dynamic stage to fatten up the sound. This makes the penalties normally associated with the use of highpass, bandpass, and band rejection filters much less of an issue.

The ES2 Parameters



If given just a few words to explain the principles behind a subtractive synthesizer, it would go something like this: The *Oscillator* generates the oscillation (or waveform), the *Filter* takes away the unwanted overtones (of the waveform), and the *Dynamic Stage* sets the volume of the permanent oscillation (the filtered waveform) to zero as long as no key is pressed.

In an analog synthesizer, these three sections are commonly called the *VCO*, *VCF*, and *VCA*, with *VC* being the abbreviation for *Voltage Controlled*, and the other letter(s) standing for *Oscillator*, *Filter*, and *Amplifier*, respectively.

The basic parameters of a synthesizer are controlled (modulated) by voltages: pitch in the oscillator, timbre in the filter, loudness in the amplifier.

These voltages are generated by modulation sources. In the ES2, the *Router* controls which parameter is controlled by which modulation source.

Finally, the synthesizer's sound is refined by effects such as distortion or chorus.

Following this simple signal path, we would like to introduce you to the modules of the ES2.

Oscillators

Tune

Tune sets the pitch of the ES2 in cents. 100 cents equals a semitone step. The range is ± 50 cents. At a value of 0 c (zero cents), a' is tuned to 440 Hz or concert pitch.



Analog

Analog alters the pitch of each note, *plus* the cutoff frequency in a random fashion. Much like polyphonic analog synthesizers, all three oscillators used by each synthesizer voice maintain their specific deviation, but are shifted by the same amount randomly. Medium values simulate the tuning instabilities of analog synthesizer circuitry which can be useful in achieving that much sought after “warmth” of the ES2’s analog hardware counterparts.



If the ES2 is set to *mono* or *legato*, *Analog* is only effective with *unison* activated. In this situation, *Analog* sets the amount of detuning between the stacked (unison) voices.

If *voices* is set to 1 and *Unison* is not activated, the *Analog* parameter has no effect.

Read more on these parameters in the “Keyboard Mode (Poly/Mono/Legato)” section, on page 227.

CBD

Fine Tuning detunes Oscillators 1, 2, and 3 in cents (1/100th percentages of a semitone). The detuning results in “beats” (phasing), the speed of which is determined by the difference between the two oscillator frequencies (given that these frequencies are nearly identical). The higher the pitch, the faster the phasing beats are. High notes may therefore seem to be further out of tune than lower notes.



The *CBD* (*Constant Beat Detuning*) parameter matches this natural effect by detuning the lower frequencies in a ratio proportionate to the upper frequencies. Besides disabling *CBD* altogether, four values are at your disposal: 25%, 50%, 75%, 100%. If you choose 100%, the phasing beats are (almost) constant across the entire keyboard range. This value, however, may be too high, as the lower notes might be overly-detuned at the point where the phasing of the higher notes feels right. Try lower values for *CBD* (and *detune*, of course) in cases where the bass appears to be a little too far out of tune.

The reference pitch for *CBD* is C3 (middle C): Its detuning stays the same, regardless of the *CBD* value.

Glide

The *Glide* parameter controls the portamento time. This is the time it takes for the pitch to travel from one note to another. Glide behavior depends on the value of the *Keyboard Mode* parameter setting.



If *Keyboard Mode* is set to *Poly* or *Mono*, and *Glide* is set to a value other than 0, portamento is active. If *Keyboard Mode* is set to *Legato*, and *Glide* is set to a value other than 0, you need to play legato (press a new key while holding the old one) to activate portamento. If you do not play legato, portamento will be inactive. This behavior is also called fingered portamento.

Bend Range

Bend Range determines the pitch range for pitch bend modulation. The range is ± 36 semitones. There are separate range settings for upwards and downwards bends, plus an optional *link* mode.



Keyboard Mode (Poly/Mono/Legato)

A *polyphonic* instrument allows the simultaneous playing of several notes—an organ or piano, as examples. Many synthesizers are *monophonic*, especially the older ones. This means that only one note can be played at a time, much like a brass or reed instrument. This shouldn't be viewed as a disadvantage in any way, because it allows playing styles that are not possible with polyphonic keyboard instruments.



You can switch between monophonic and polyphonic modes by clicking on the *Poly* and *Mono* buttons. *Legato* is also monophonic, but with one difference: The envelope generators are only retriggered if you play staccato (release each key before playing a new key). If you play legato (press a new key while holding the old one), the envelope generators are only triggered with the first note you play legato, and then continue their curve until you release the last legato played key. If you switch to *Mono*, legato or staccato playing does not have this impact: The envelope generators are retriggered with every new note that is played.

Note: If you switch to *Legato*, you need to play legato to actually hear the *Glide* parameter taking effect.

Note: On several monophonic synthesizers, the behavior in *Legato* mode is referred to as *single trigger*, while *Mono* mode is referred to as *multi trigger*.

Voices

The maximum number of notes that can be played simultaneously is determined by the *Voices* parameter. Maximum value for *Voices* is 32.

The value of this parameter has a significant impact on the computer processing resources demanded by the ES2 when played at its maximum polyphony. Reduce this value to the number of voices that you actually *require* for the part. Setting it to a higher value places higher overheads on the CPU, and wastes resources.

Unison

A forte of polyphonic analog synthesizers has always been unison mode. Traditionally, in unison mode, classic analog polysynths run monophonically, with all voices playing a single note simultaneously. As the voices of an analog synthesizer are never perfectly in tune, this results in an extremely fat chorus effect with great sonic depth. Switch the ES2 to *Mono* or *Legato* and switch on *Unison* in order to achieve and hear this effect. The intensity of the unison effect depends on the number of *Voices* selected. Remember that the amount of processing power required is correlated to the number of voices! The intensity of detuning (voice deviation) is set via the *Analog* parameter.



In addition to this classic monophonic unison effect, the ES2 also features a polyphonic unison effect. In *Poly/Unison*, each note played is effectively doubled, or more correctly, the polyphony value of the *Voices* parameter is halved. These two voices are then used for the single triggered note.

Switching on *Poly* and *Unison* has the same effect as setting the ES2 to *Mono*, *Unison*, and *Voices* = 2, except that it can be played polyphonically.

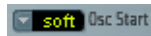
Osc Start

The oscillators can run freely, or they can begin at the same phase position of their waveform cycle each time you hit a key (every time the ES2 receives a note on message).

When *Osc Start* (*Oscillator Start*) is set to *free*, the initial oscillator phase startpoint is random, with each note played. This gives the sound more life and a less static feel—just like an analog hardware synthesizer. On the other hand, the output level may differ each time you play a note, and the attack phase may sound less punchy.



If you set *Osc Start* to *soft*, each initial oscillator phase will start at a zero crossing every time a note is played. This mimics the sonic character of a normal digital synthesizer.



If *Osc Start* is set to *hard*, each initial oscillator phase begins at the highest possible level in its waveform cycle every time a note is played. This punch is only audible if the *ENV3 Attack Time* is set to a minimal value—a very fast attack. This setting is highly recommended for electronic percussion and hard basses.



Note: *Osc Start* *soft* and *hard* result in a constant output level of the initial oscillator phase every time the sound is played back. This may be of particular importance when using Logic's Bounce feature, at close to maximum recording levels.

Flt Reset

If you increase the filter's *Resonance* parameter to higher values, it will begin to internally feed back and will start to self-resonate. This resonance results in a sine oscillation (a sine wave), which you may be familiar with, if you've used subtractive synthesizers before.



In order to start this type of oscillation, the filter requires a trigger. In an analog synthesizer, this trigger may be the noise floor or the oscillator output. In the digital domain of the ES2, noise is all but eliminated. As such, when the oscillators are muted, there is no input signal routed to the filter.

When *Filter Reset* is engaged, however, each note starts with a trigger that is used to make the filter resonate immediately.

Frequency Switch

Switches the pitch in semitone steps over a range of ± 3 octaves. As an octave consists of 12 semitones, the ± 12 , 24, and 36 settings represent octaves. You can click on these options to quickly set the corresponding octave.



The value display works as follows: the left numbers show the semitone (s) setting, the right numbers show the cent (c, 1 cent = 1/100th semitone) setting. You can adjust these two values independently. As an example: an oscillator with the value 12 s 30 c sounds an octave (12 semitones) and 30 cents higher than an oscillator with the value 0 s 0 c.

Note: The fifth (seven semitones), and all settings that correspond with harmonics of an oscillator set to 0 semitones (19, 28 semitones) result in harmonic sounds.

Muting Oscillators

By clicking on the green numbers to the right of the oscillators, you can mute and unmute them independently. This saves processor power.

Wave

Each of the three oscillators features a rotary knob that allows you to select a waveform. This is responsible for the basic harmonic content and tone color of a sound. Oscillators 2 and 3 are almost identical to each other, but differ from Oscillator 1. Oscillator 1 is capable of generating a sine wave, the frequency of which can be modulated in the audio range, for true FM synthesis sounds. Oscillators 2 and 3 can be synchronized to, or be ring-modulated with, Oscillator 1. They also feature rectangular waves with freely definable fixed pulse widths plus pulse width modulation (PWM) facilities. Via the *Router*, the rectangular and pulse waves of Oscillator 1 can be modulated in width in conjunction with the synchronized and ring-modulated rectangular waves of Oscillators 2 and 3.

Note: The *Filter* button disables the entire filter section. This makes listening to the pure oscillator waveforms easier.



Oscillator 1 Waveforms

Oscillator 1 outputs standard waveforms—pulse, rectangular, sawtooth, and triangular waves—or, alternately, any of the 155 available DigiWaves. It can also output a pure sine wave.

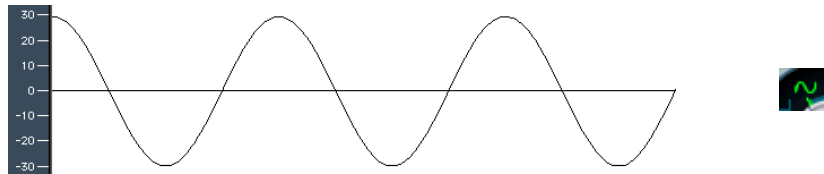


The sine wave can be modulated in frequency by Oscillator 2 in the audio frequency range. This kind of linear frequency modulation is the basis on which FM synthesis works. FM synthesis was popularized by synthesizers such as Yamaha's DX7 (the architecture of which is much more complex, when it comes to FM synthesis).

A click on the Oscillator number disables the output of Oscillator 1.

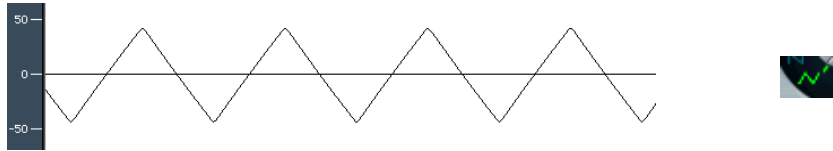
Note: Even when Oscillator 1 is turned off, it is still available for use as a modulation and synchronization source for Oscillators 2 and 3.

We will now take a closer look at the different waveforms available to Oscillator 1.



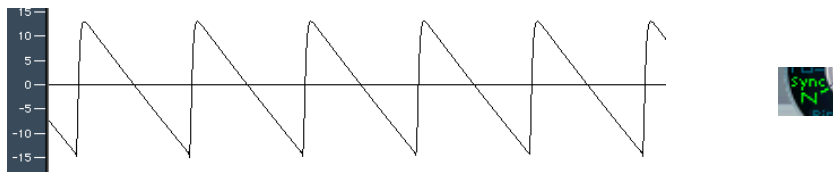
Screenshot of ES2's Oscillator 1, sine wave selected. The sample was created with Logic's Bounce function and is displayed in the Sample Editor. The sine wave shown is in its basic oscillation form. It contains no harmonics. According to the theorem of Jean Baptiste Fourier, all regular waveforms can be interpreted as the sum of sine oscillations with defined frequency, amplitude, and phase position, with their frequencies being "harmonic"—having integer frequency ratios, in other words.

During a Fourier transformation, complex oscillations can be divided into their basic sine components. In additive synthesis, complex oscillation forms can be re-synthesized. The most simple additive synthesizer is the drawbar organ (the Hammond organ, for example). With such an organ, you can mix nine sine choirs with drawbars. Try selecting sine waves for all three oscillators and the following semitone settings: -12 ($16'$), 0 ($8'$) and $+7$ ($5\frac{1}{3}'$), and set all oscillators to the same level. Select an organ envelope, and voila—a classic organ sound!



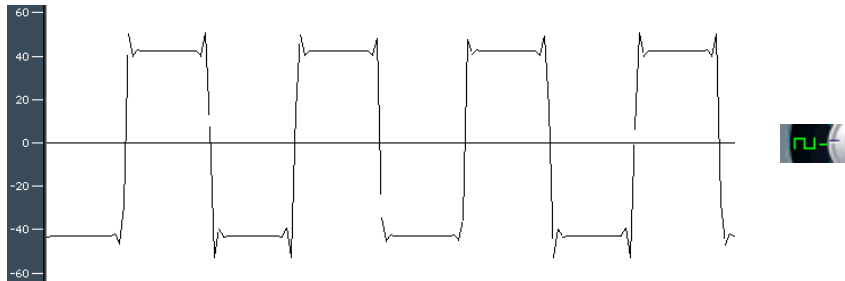
Screenshot of the ES2's triangular wave, created and shown as above. The triangular wave only contains odd harmonics (no octaves), the amplitudes of which decrease square-proportionately to their number. This means that its sound has few overtones. This corresponds with its appearance, which is reminiscent of a sine wave.

Classic synthesizer literature encourages the use of the triangular wave for the creation of flute-like sounds. In the age of sampling, however, it's pretty hard to sell a triangular wave as a flute sound to anyone.



Screenshot of the ES2's sawtooth wave. The sawtooth wave contains all harmonics, the amplitudes of which decrease proportionately with their number.

Classic synthesizer literature indicates the use of the sawtooth wave to create a sound similar to that of a violin. The rich and full sound of the sawtooth wave is the most popular synthesizer waveform, and serves as a basis for synthetic string and brass sounds. It is also handy for synthesized bass sounds.



Screenshot of the ES2's rectangular wave. The 50% rectangular wave contains all odd harmonics, the amplitudes of which decrease proportionately with their number. The pulse width can be set to any value and serves as a modulation address.

Classic synthesizer literature likens the rectangular wave to the sound of a clarinet, as the clarinet doesn't feature any even harmonics in a certain frequency range either. The typical hollow sound of the rectangular wave is achieved with its pulse width value set to 50%. Heavily lowpass-filtered rectangular waves are popular as techno bass sounds. Changing the pulse width to something around 75% results in quite a nasal sound, resembling an oboe.

Digiwaves

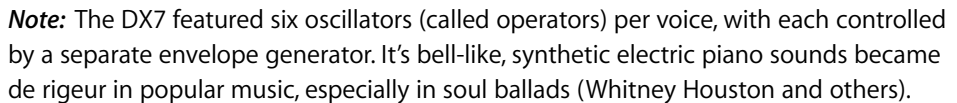
The ES2, as we've mentioned, not only features the most popular synthesizer waveforms, but also a selection of 100 additional waveforms, called Digiwaves.

Note: You can select the Digiwave by touching its name and moving the mouse vertically. You can select it numerically when you hold Shift.

The number of the Digiwave is a parameter that can be modulated. By modulating the *OscWave* target, you can scroll through the list of Digiwaves. Choose a sufficiently low modulation intensity and speed to hear the single Digiwaves being crossfaded. The Digiwaves of the three oscillators can be modulated individually or together. The modulation targets are discussed in "OscWaves", on page 255, through to "OscWaveB", on page 256.

Given the ability to modulate Digiwaves, the ES2 can produce sounds resembling those of the famous wavetable synthesizers from PPG and Waldorf (and also the Korg Wavestation).

The principle of linear frequency modulation (FM) synthesis was developed in the late sixties and early seventies by John Chowning. It's such a flexible and powerful method of tone generation that synthesizers were developed, based solely on the idea of FM synthesis. The most popular FM synthesizer ever built is Yamaha's DX7. FM synthesis is also found in other models of the Yamaha DX range and some Yamaha E-Pianos. In the discipline of pure FM synthesis, the ES2 can't be compared with these synthesizers, but it can certainly achieve some of their signature sounds.

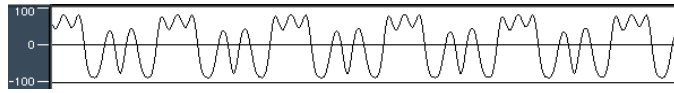


Note: *Osc1Wave* is optimized for subtle FM sounds, utilizing lower FM intensity amounts. For more extreme FM modulations, the Router offers the *Osc1WaveB* target. See the “Modulation Targets,” on page 254.

The effect is similar to an LFO modulation, being used to create a vibrato (a periodic modulation of the frequency) or a slow siren effect. In comparison to an LFO, Oscillator 2 does not oscillate slowly. In the audio domain, it oscillates a little faster than Oscillator 1 itself. Thus, Oscillator 1's oscillation also accelerates and slows down within a single phase, resulting in a distortion of the basic sine shape of Oscillator 1. This sine wave distortion has the side benefit of a number of new harmonics becoming audible.

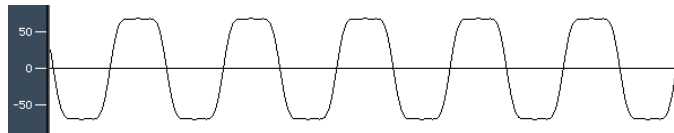


Oscillator 1's frequency modulated sine wave, modulated by Oscillator 2 set to sine wave. Oscillator 2 was set to three times the frequency of Oscillator 1 (+19 semitones). The modulation intensity is low (*Wave* control at about 12 o'clock). As the wavelength (the duration period) of the modulating Oscillator is a third of that of the modulated Oscillator, the sine is accelerated and slowed down three times within a phase.

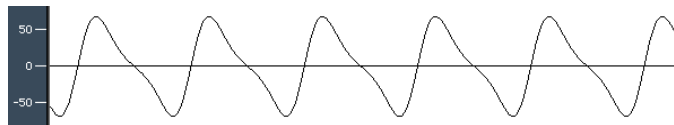


Oscillator 1's frequency modulated sine wave, modulated by Oscillator 2 set to sine wave. Oscillator 2 was set to three times the frequency of Oscillator 1 (+19 semitones). The modulation intensity is much higher (*Wave* control at about 3 o'clock). The distortion of the basic sine wave is much stronger, resulting in more harmonics becoming audible.

The effect of frequency modulation depends on *both* the modulation intensity and frequency ratio of both Oscillators.



The upper graphic shows a slightly deformed sine wave of Oscillator 1, modulated in frequency by Oscillator 2 at double the speed of the carrier (Oscillator 1). The resulting waveform resembles a rectangular wave or clipped sine wave.



The upper graphic shows a slightly frequency modulated sine as the output signal of Oscillator 1, with the modulator frequency being identical to the carrier frequency. The resulting waveform resembles a lowpass filtered sawtooth wave.

The resulting spectrum not only depends on the frequency modulation intensity and frequency ratio, but also on the waveform used by the modulating oscillator (Oscillator 2). The modulation that takes place varies according to the waveform selection for Oscillator 2—it might even be an oscillation that is synchronized to Oscillator 1! Given the 100 available Digiwaves, countless combinations of modulation intensities and frequency ratios, the frequency modulation of the two oscillators delivers an infinite pool of spectra and tone colors.

Waveforms of Oscillators 2 and 3

Basically, Oscillators 2 and 3 supply the same selection of analog waveforms as Oscillator 1: sine, triangular, sawtooth, and rectangular waves. The pulse width can be scaled steplessly between 50% and the thinnest of pulses, and can be modulated in a number of ways (see “Pulse Width Modulation” section, on page 236).

Oscillators 2 and 3 also offer the selection of:

- a rectangular wave, synchronized to Oscillator 1,
- a sawtooth wave, synchronized to Oscillator 1,
- a ring modulator, which is fed by the output of Oscillator 1 and a square wave from Oscillator 2,
- colored noise for Oscillator 3.

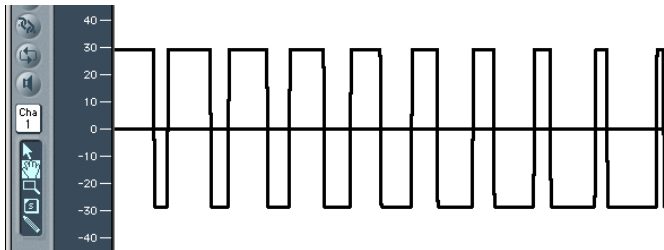
Synchronization and ring modulation afford the creation of very complex and flexible harmonic spectra. The principle of Oscillator synchronization is described on page 237, and ring modulation on page 238.

Pulse Width Modulation

Oscillators 2 and 3 allow you to scale the width of the pulses to any value. The spectrum and tone color generated by these oscillators depends on the pulse width. The pulse width can be modulated. You can even modulate the pulse width of the square and pulse wave of Oscillator 1, the pulse width of the synchronized pulse waves of Oscillator 2 and 3, and the square wave of Oscillator 2's ring modulator.



This width modulation is controlled in the *Router* (the Modulation Matrix). The pulse width is defined by the waveform rotary control. The graphic below shows a pulse wave, with the pulse width modulated by an LFO. You can clearly see how the width of the pulses changes over time.



Screenshot of an ES2 sample, created with the Bounce function of Logic. The pulse width is modulated. It is easy to see how the width of the pulses varies between a rectangular shape and very thin pulses. An LFO is selected as the modulation source, and its waveform is a sine wave. You can see about half a phase of the sine wave. If a rectangular wave had been selected for the LFO, you would see the pulse width periodically changing between the two fixed extreme values.

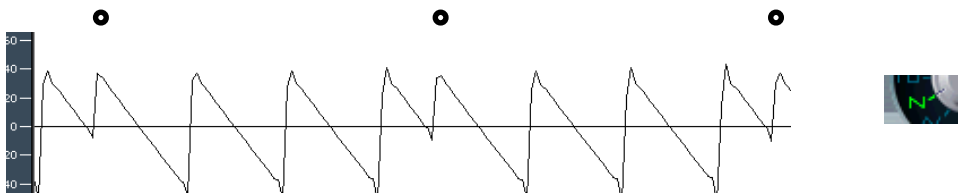
Note: A pulse wave (with its width modulation controlled by an LFO set to a sine wave) makes a single oscillator sound vivid, undulating, and rich with overtones. Sonically, this is somewhat like the sound of two slightly detuned, phasing oscillators. The effect sounds great with sustained bass and pad sounds. Select the intensity and speed of the modulation with care, as the overall volume (and level of the first partial) decreases, and slight detuning occurs, when the pulses become very thin (below 10%).

Note: Pulse width modulations via velocity sensitive envelope generators sound very dynamic—a great effect, especially for percussive bass sounds.

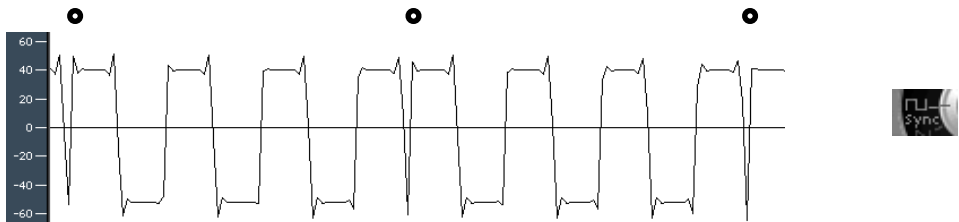
Sync

You can see that the rectangular and sawtooth waveforms also feature a *Sync* option. In this mode, the frequency of Oscillator 2 (or 3, respectively) is synchronized to the frequency of Oscillator 1. This does not mean that their frequency controls are simply disabled. They still oscillate at their selected frequencies, but every time that Oscillator 1 starts a new oscillation phase, the synchronized Oscillator is also forced to restart its phase from the beginning. This is practical while the frequency of the synchronized Oscillator is significantly higher than that of Oscillator 1.

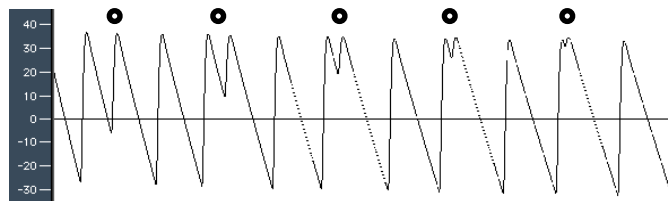
The graphic shows the output signal of Oscillator 2, set to a synchronized sawtooth wave. Its frequency is about three and a half times that of Oscillator 1. Oscillator 1 was set to 0 semitones, and Oscillator 2 to +22 semitones. The oscillogram clearly shows how the phase of Oscillator 2 is forced to restart after about three and a half phases. Between the pulses of Oscillator 1, Oscillator 2 runs freely.



Synchronized sawtooth wave as output by Oscillator 2. Oscillator 1 is set to 0, Oscillator 2 to +22 semitones. The dots in the graphic indicate the phases of Oscillator 1.



Synchronized rectangular wave as output by Oscillator 2. Oscillator 1 is set to 0, Oscillator 2 to +22 semitones. The dots in the graphic indicate the phases of Oscillator 1.



Output signal of Oscillator 2 (sawtooth), synchronized to Oscillator 1. The frequency of Oscillator 2 is modulated by an envelope generator, changing the duration of the phases over time. On the right-hand side, the sawtooth shapes are broader, due to the sinking frequency of Oscillator 2.

At regular intervals, defined by the phase duration of Oscillator 1, the waveform is forced back to its beginning. The dots in the graphic indicate the phases of Oscillator 1. The distance between these dots remains constant, as the frequency of Oscillator 1 is not being modulated.

Synchronized oscillator sounds are especially cool when the frequency of the synchronized oscillator is modulated by an envelope generator. This way, the number of phases within a section (phase) of the synchronization cycle always changes, and so does the spectrum. Typical oscillator sync sounds tend towards the aggressive, screaming leads that synthesizer manufacturers like to talk about.

Ring (Ring Modulation)

The waveform control of Oscillator 2 also features the *Ring* setting. In this mode, Oscillator 2 outputs the signal of a ring modulator. This ring modulator is fed with the output signal of Oscillator 1 and a square wave of Oscillator 2. The pulse width of this square wave can be modulated.

A ring modulator has two inputs. At its output you will find the sum and difference frequencies of the input signals.



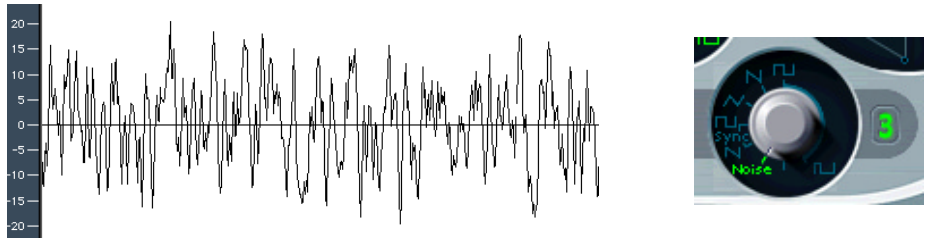
The graphic shows the output signal of the ring modulator, appearing as the output signal of Oscillator 2. The amplitude (or the elongation, to be more exact) of the output of Oscillator 2 changes with the phase of Oscillator 1. Oscillator 2 is set to a higher frequency than Oscillator 1. As the frequency ratio is odd (irrational), the resulting waveform always changes over time. The resulting spectrum is inharmonic and you won't hear a clearly defined pitch.

If you ring-modulate a sine oscillation of 200 Hz with a sine oscillation of 500 Hz, the output signal of the ring modulator will consist of a 700 Hz (sum) and a 300 Hz (difference) signal. Negative frequencies result in a change of the phase polarity of the output signals. With sawtooth and rectangular input signals, the output signal is much more complex, as these harmonically-rich waveforms produce a number of extra side bands.

Note: Ring modulation is a powerful tool for inharmonic, metallic sounds as the spectra resulting from its use are inharmonic with almost every frequency ratio. The ring modulator was the tool of choice for bell-like sounds, and dates back to the early days of the synthesizer (see the “RingShifter—Ring Modulator/Frequency Shifter” section, on page 101).

White and Colored Noise (Oscillator 3 only)

Unlike Oscillator 2, Oscillator 3 is not capable of producing ring modulated signals nor sine waves. Its sonic palette however, is bolstered by the inclusion of a noise generator. By default, Oscillator 3's noise generator generates *white noise*. White noise is defined as a signal that consists of all frequencies (an infinite number) sounding simultaneously, at the same intensity, in a given frequency band. The width of the frequency band is measured in Hertz. Its name is analogous to white light, which consists of a mixture of all optical wavelengths (all rainbow colors). Sonically, white noise falls between the sound of the consonant *F* and breaking waves (surf). Synthesis of wind and seashore noises, or electronic snare drum sounds, requires the use of white noise.



Screenshot of a white noise sample, showing the output signal of Oscillator 3.

Oscillator 3 has more up its sleeve than the output of neutral sounding white noise—you also can make it hiss or rumble. Even better, you can modulate this sound color in real-time without using the main filters of the ES2.

If the waveform of Oscillator 3 is modulated, (*Osc3Wave*) the color of the noise will change. It can be filtered by a dedicated highpass or lowpass filter with a descending slope of 6 dB/octave. At negative values, the sound becomes darker (red); The lowpass filter can be tuned down to 18 Hz with a setting of -1 . When *Osc3Wave* is modulated positively, the noise becomes brighter (blue): At a value of $+1$ for *Osc3Wave*, the highpass filter's cutoff frequency is set to 18 kHz. This filtering of the noise signal is independent of the main filters of the ES2, and can be automatically changed in real-time.

Oscillator Mix Field—the Triangle

By grabbing, and moving the cursor in the Triangle, you can crossfade between the three oscillators. This is pretty self-explanatory. Moving the cursor along one of the Triangle's sides will crossfade between two oscillators, with the third oscillator being muted.



The position of the cursor can be controlled via the vector envelope, just like the cursor position in the Track Pad (the Square), which we'll look at in "The Square" section, on page 272.

Note that the *vector envelope* features a *loop* function. This addition extends its usefulness, allowing you to view it as a luxurious pseudo-LFO with a programmable waveform. It can be used for altering the positioning of the Triangle and Square cursors. Read more about this in the "Vector Mode" section, on page 272, and "The Vector Envelope" section, on page 273.

Triangle Values in Control View

Internally, the position of the cursor in the Triangle is described by two parameters (coordinates, actually) which are relevant when it comes to automating the oscillator mix. These parameters, called *OscLevelX* and *OscLevelY*—are visible in the Controls view. Don't confuse them with the X and Y positions of the *Square*.

If you intended to edit a Region containing oscillator mix automation data in the Hyper Editor, you would use the following MIDI controller values. Take a look at the information below to get a feel for how they operate.

In order to listen to Oscillator 1 only ...

- choose *OscLevelX*= 1.0 (*MIDI*: 127) and *OscLevelY*= 1.154 (*MIDI*: 127).

In order to listen to Oscillator 2 only ...

- choose *OscLevelX*= 0.0 (*MIDI*: 0) and *OscLevelY*= 0.577 (*MIDI*: 64).

In order to listen to Oscillator 3 only ...

- choose *OscLevelX*= 1.0 (*MIDI*: 127) and *OscLevelY*= 0.0 (*MIDI*: 127).

In order to listen to Oscillator 1 and 2 only ...

- choose *OscLevelX*= 0.5 (*MIDI*: 64) and *OscLevelY*= 0.866 (*MIDI*: 95).

In order to listen to Oscillator 1 and 3 only ...

- choose *OscLevelX*= 1.0 (*MIDI*: 127) and *OscLevelY*= 0.577 (*MIDI*: 64).

In order to listen to Oscillator 2 and 3 only ...

- choose *OscLevelX*= 0.5 (*MIDI*: 64) and *OscLevelY*= 0.288 (*MIDI*: 32).

All Oscillators have the same level at ...

- *OscLevelX*= 0.667 (*MIDI*: 85) and *OscLevelY*= 0.577 (*MIDI*: 64).

Filters

The ES2 features two dynamic *filters* which are equivalent to the Voltage Controlled Filters (VCF) found in the world of analog synthesizers. The two filters are *not* identical. *Filter 1* features several modes: lowpass, highpass, bandpass, band rejection, peak. *Filter 2* always functions as a lowpass filter. Unlike Filter 1, however, Filter 2 offers variable slopes (measured in dB/octave).



The *Filter* button bypasses (switches off) the entire *filter* section of the ES2. Disabling the filters makes it easier to hear adjustments to other sound parameters, as the filters always heavily affect the sound. If the word *Filter* is green, the filters are engaged. Disabling the filters reduces processor load.



Series and Parallel Filter Signal Flow

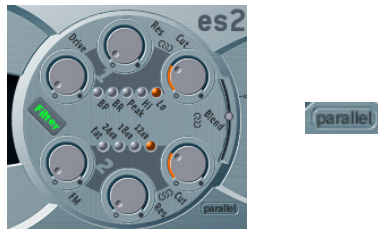
You can rotate the entire (circular) filter segment of the ES2 user interface. Press the button labelled *Parallel* or *Series* to do so. The label and position/direction of the filter controls clearly indicate the current signal flow.



In the position displayed above, the filters are cabled serially. This means that the signal of the *Oscillator Mix* section (Triangle) passes through the first filter, and then this filtered signal passes through Filter 2—if *Filter Blend* is set to 0 (middle position). See the “Filter Blend: Cross-Fading the Filters” section, on page 243 for a detailed description of this parameter.

The mono output signal of Filter 2 is then fed into the input of the dynamic stage (the equivalent of a VCA in an analog synthesizer), where it can be panned in the stereo spectrum, and then fed into the effects processor.

In the graphic below, the filters are cabled in parallel. If *Filter Blend* is set to 0, you'll hear a 50/50 mix of the source signal routed via Filter 1 and Filter 2, which is fed into the mono input of the dynamic stage. There it can be panned in the stereo spectrum, and then fed into the effects processor.



Filter Blend: Cross-Fading the Filters

You can cross-fade the two filters. When wired in parallel, you'll find this quite obvious to look at and understand: If *Filter Blend* is set to its top position, you'll only hear *Filter 1*. If *Filter Blend* is set to its lowest position, you'll only hear *Filter 2*. In between these positions, the filters are cross-faded.

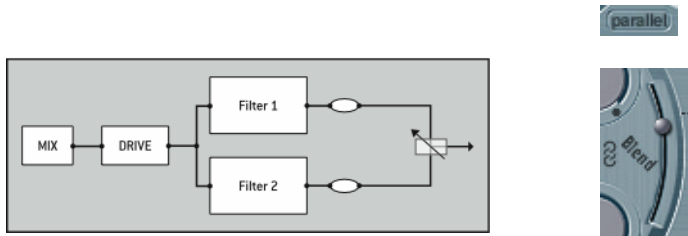
More often, you'll want to cable the filters serially. It should be noted that even in series mode, it's possible to cross-fade the filters! This is achieved through the use of controllable side chains (bypassing lines). In this serial cabling scenario, the distortion circuits controlled by the *Drive* parameter also need to be considered, as the distortion circuits are positioned before or in-between the filters, dependent on *Filter Blend*. All fades occur flawlessly, in very fine steps.

The *Filter Blend* parameter can be modulated dynamically in the Router!

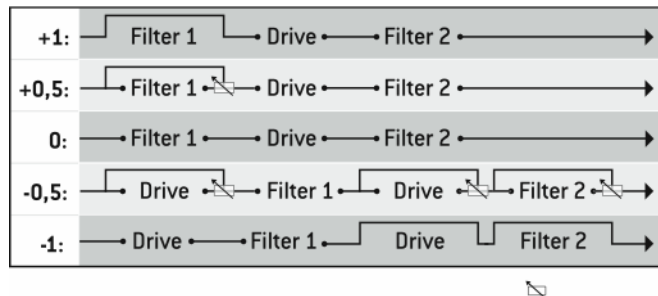
Filter Blend and Signal Flow

No matter whether parallel or series filter configurations are chosen, a *Filter Blend* setting of -1 always results in only *Filter 1* being audible. A *Filter Blend* setting of $+1$ will limit audibility to *Filter 2*. This is reflected in the user interface.

In conjunction with the overdrive/distortion circuit (*Drive*) and a series wiring configuration, the ES2's signal flow is anything but commonplace. The graphics illustrate the signal flow between the *Oscillator Mix* stage (the Triangle) and the dynamic stage, which is always controlled by ENV 3. The signal flow through the filters, the overdrives and the bypassing sidechains is dependent on the *Filter Blend* setting.



Filter Blend in parallel filter mode.



Filter Blend in Serial filter mode. Between 0 and -1 two distortion circuits are active—one “pre” each filter. Filter Blend cross-fades up to three bypassing lines simultaneously.

Filter Blend and Serial Filter Configuration Tips

- With *positive* values for *Filter Blend*, Filter 1 is partially bypassed.



- With *negative* values for *Filter Blend*, Filter 2 (the low pass) is partially bypassed.
- At zero and *positive* values for *Filter Blend*, there is only one overdrive circuit for *both* filters.
- With *negative* values, another overdrive circuit is introduced, distorting the output signal of the *Oscillator Mix* stage before it is fed into the first filter—where *Filter Blend* = -1 .

Note: If *Drive* is set to 0, no distortion occurs.

Filter Blend and Parallel Filter Configuration Tip

The overdrive/distortion circuit is always wired *after* the *Oscillator Mix* stage and *before* the filters. The filters receive a mono input signal from the overdrive circuit's output. The outputs of both filters are mixed to mono via *Filter Blend*.



Note: If *Drive* is set to 0, no distortion occurs.

Drive

The filters are equipped with separate overdrive modules. Overdrive intensity is defined by the *Drive* parameter. If the filters are connected in parallel, the overdrive is placed before the filters. If the filters are connected serially, the position of the overdrive circuits depends on the *Filter Blend* parameter—as described above.



Polyphonic Distortions In the Real World

The ES2 features a *distortion* effect, equipped with a tone control, in the *Effects* section. Given the inclusion of this effect, you may be wondering what benefit the *Drive* function in the *Filter* section brings?

The distortion circuit in the *Effects* section affects the sum of the entire polyphonic synthesizer performance. Thus, more complex chords (other than major chords, parallel fifths and octaves) sound rough, when using distortion. Every rock guitarist knows this. Due to these intermodulation distortions, distorted guitar playing is generally performed by using few voices or parallel fifths and octaves.

The *Filter Drive* affects every voice individually—and when every voice in the ES2 is overdriven individually (like having six fuzz boxes for the guitar, one for each string), you can play the most complex harmonies over the entire keyboard range. They'll sound clean, without unwanted intermodulation effects spoiling the sound.

Furthermore, appropriate *Drive* parameter settings lead to character. To explain, the way analog filters behave when overdriven forms an essential part of the sonic character of a synthesizer. Each synthesizer model is an individual with regard to the way its filters behave when overdriven. The ES2 is extremely flexible in this respect, allowing for the most subtle fuzz through to the hardest distortions.

Finally, in series mode, the distortion always takes place *before* the lowpass filter (Filter 2). As the lowpass filter (Filter 2) can filter (cut) away the overtones introduced by the distortion, the *Drive* feature can be seen (and used) as another tool for deforming the oscillator(s) waveforms.

To check out how the overdrive circuit between the filters works, program a sound as follows:

- Simple static waveform (a sawtooth)
- Filter set to *Serial* mode
- *Filter Blend* set to 0 (center position)
- Set Filter 1 to *peak Filter Mode*
- Set a high *Resonance* value for Filter 1
- Modulate *Cutoff Frequency 1* manually or in the *Router*
- Set *Drive* to your taste
- Filter away (cut) the high frequencies with Filter 2 to taste

The sonic result resembles the effect of synchronized oscillators. With high resonance values, the sound tends to scream. Modulate the *Resonance* of Filter 1, if you wish.

Filter Parameters

Cutoff and Resonance

With every lowpass filter (in the ES2: *Lo* mode for Filter 1 and *all* of Filter 2's modes), all frequency portions *above* the *Cutoff Frequency* (*Cut*) are suppressed, or cut off, hence the name. The *Cutoff Frequency* controls the brilliance of the signal. The higher the *Cutoff Frequency* is set, the higher the frequencies of signals that are allowed to pass through the lowpass filter.



Resonance (*Res*) emphasizes the portions of the signal which surround the frequency defined by the *Cutoff Frequency* value. This emphasis can be set so intensively in Filter 2, that the filter begins to oscillate by itself. When driven to self-oscillation, the filter outputs a sine oscillation (sine wave). This self-oscillation can be supported by the *Filter Reset* parameter. See “Flt Reset” on page 229 for details.



Note: If you are new to synthesizers, experiment with a simple saw wave, using Oscillator 1, and Filter 2 (lowpass filter, *Filter Blend* = +1) on its own. Experiment with the *Cutoff Frequency* and *Resonance* parameters. You'll quickly learn how to emulate a number of recognizable sounds, and will pick up the basic principles of subtractive synthesis intuitively.

Note: The dynamic lowpass filter is the most essential module in any subtractive synthesizer. This is why Filter 2 always operates in lowpass mode.

Note: As opposed to the filter and EQ effect plug-ins in Logic, the ES2's filters are dynamic, which means that the *Cutoff Frequency* parameter can be modulated extremely quickly and severely in real-time—even on modulation signals in the audio frequency range.

The Chain Symbols

Manipulating the *Cutoff* and *Resonance* controls in real-time is one of the key ingredients in the creation of expressive synthesizer sounds. You'll be pleased to know that you can control two filter parameters at once by dragging on one of the three little chain symbols in the filter graphic.

- The chain between *Cut* and *Res* of Filter1 controls *Resonance* (horizontal mouse movements) and *Cutoff* (vertical mouse movements) of the first filter simultaneously.
- The chain between *Cut* and *Res* of Filter2 controls *Resonance* (horizontal mouse movements) and *Cutoff* (vertical mouse movements) of the second filter simultaneously.
- The chain between Filter1's *Cut* and Filter2's *Cut* controls *Cutoff* (vertical mouse movements) of the first filter, and *Cutoff* (horizontal mouse movements) of the second filter simultaneously.

Filter Slope

A filter can not completely suppress the signal portion outside the frequency range defined by the *Cutoff Frequency* parameter. The slope of the filter curve expresses the amount of rejection applied by the filter (beneath the cutoff frequency) in dB per octave.



Filter 2 offers three different slopes: 12 dB, 18 dB and 24 dB per octave. Put another way, the steeper the curve, the more severely the level of signals below the *cutoff frequency* are affected in each octave.

Fat

Increasing the *Resonance* value results in a rejection of bass (low frequency energy) when using lowpass filters. The *Fat* switch compensates for this side-effect, delivering a bassier sound.

Filter Mode (Lo, Hi, Peak, BR, BP)

Filter 1 can operate in several modes, allowing the filtering (cutting away) and/or emphasis of specific frequency bands.

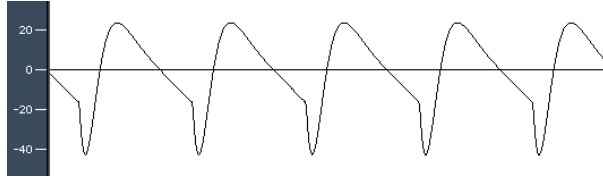


- A *lowpass filter* allows frequencies that fall below the cutoff frequency to pass. When set to *Lo*, the filter operates as a lowpass filter. The slope of Filter 1 is 12 dB/octave in *Lo* mode.
- A *highpass filter* allows frequencies above the cutoff frequency to pass. When set to *Hi*, the filter operates as a highpass filter. The slope of Filter 1 is 12 dB/octave in *Hi* mode.
- In *Peak* mode, Filter 1 works as a *Peak Type Filter*. This allows for a level increase in a frequency band, the width of which is controlled by the *Resonance* parameter.
- The abbreviation *BR* stands for *band rejection*. In this mode, the frequency band (a range of adjacent frequencies) directly surrounding the cutoff frequency is rejected, whilst the frequencies outside this band can pass. The *Resonance* parameter controls the width of the rejected frequency band.
- The abbreviation *BP* stands for *bandpass*. In this mode, only the frequency band directly surrounding the cutoff frequency can pass. All other frequencies are cut. The *Resonance* parameter controls the width of the frequency band that can pass. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the band.

Impact of the Filters on the Waveform

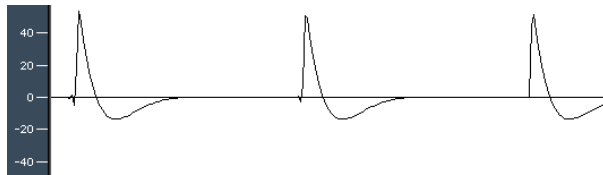
Below, you'll find a number of oscillograms of a sawtooth wave generated by Oscillator 1. These images illustrate the effect of the various modes of Filter 1. The cutoff frequency was set so that it is equal to the frequency of the first overtone (twice the frequency of Oscillator 1). The duration and wavelength of this overtone (the second harmonic) is half as long as the duration and wavelength of the first harmonic (the fundamental tone).

Lowpass Filter



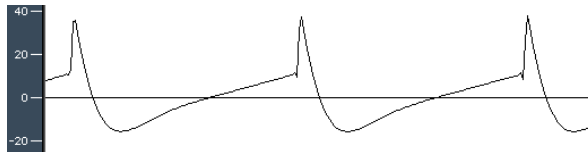
Lowpass-filtered sawtooth wave, Cutoff Frequency one octave above the frequency of the sawtooth. As the slope isn't infinite, harmonics are still visible. Note that the waveform is rounded.

Highpass Filter



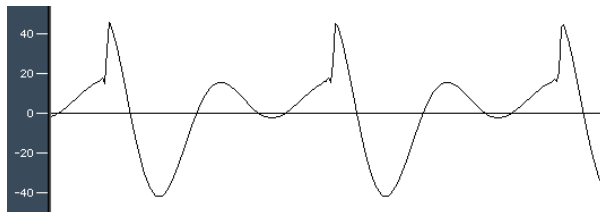
Highpass-filtered sawtooth wave, Cutoff Frequency one octave above the frequency of the sawtooth. The basic harmonic is rejected by about 12 dB, as the slope is 12 dB/octave. Try to mentally add a sine curve with the basic wavelength to the graphic: The sum of both would result in a sawtooth wave.

Peak Type Filter



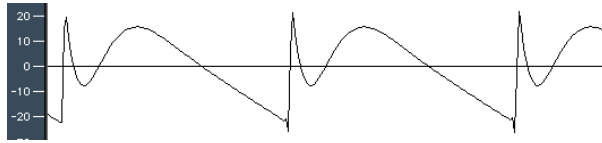
The second partial is not easy to see, even though all other partials are rejected.

Peak Type Filter with Resonance



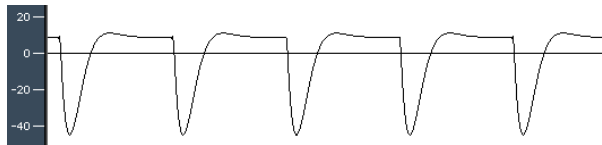
Here, the second partial is strongly emphasized.

Band Rejection



In this image, the second partial is rejected. The basic oscillation and all other harmonics are present. Sawtooth is still the input waveform.

Bandpass Filter



The bandpass filter distorts the picture of the sawtooth wave.

Filter 2 FM

The cutoff frequency of Filter 2 can be modulated by the sine wave of Oscillator 1, which means that it can be modulated in the audio frequency range.



The effect of such filter modulations in the audio spectrum is unpredictable, but the results tend to remain harmonic as long as the modulation intensity doesn't get too high. *FM* defines the intensity of frequency modulation. This parameter can be modulated in real-time: In the *Router*, this modulation is abbreviated as *LPF FM*.

A clean sine wave, at the frequency of Oscillator 1, is always used as the modulation source.

Note: Don't confuse this type of filter frequency modulation with the FM feature of Oscillator 1, which can be modulated by Oscillator 2, as described in the "Linear Frequency Modulation" section, on page 234.

Note: If a frequency modulation of Oscillator 1 by Oscillator 2 is used, it does not influence the (sine wave) signal used to modulate the cutoff frequencies.

Note: Filter 2 can be driven to self-oscillation. If you set a very high value for *Resonance*, it will produce a sine wave. This self-oscillating sine wave will distort at the maximum *Resonance value*. If you mute all oscillators, you'll only hear this sine oscillation. By modulating the Cutoff Frequency, you can produce effects similar to those produced by modulating the frequency of Oscillator 1 with Oscillator 2.

Handling Processing Power Economically

The ES2 is designed to make the most efficient use of computer processing power. Modules and functions that are not in use don't use processing power. This principle is maintained by all elements of the ES2.

As examples: If only one of the three oscillators is in use, and the others are muted, less processing power is required. If you do not modulate Digiwaves, or if you disengage the filters, processing power is saved. When it comes to filtering, here are some hints that will help you to use processing power most efficiently:

- If you can achieve the same lowpass-filtered sound with Filter 1 as with Filter 2, use Filter 1. Filter 1 uses less processing power, although it differs a little, sonically.
- *Filter FM* uses additional processing power. If you don't need it, don't use it.
- Modulation of the *Filter Blend* parameter requires quite a bit of additional processing power, as soon as it is engaged in the Router.
- *Drive* requires additional processing power. This is especially the case when it comes to filters wired in *series* and *Filter Blend* settings with two distortion circuits. See "Filter Blend and Signal Flow," on page 243 for details.

Remember that you'll never be forced to make compromises in your sound! You can always make use of Logic's Bounce features in order to convert a processor-intensive Audio Instrument track into an audio track, playing back a bounced audio file. To do so, route the Audio Instrument (the ES2) to an Output Object. Switch the ES2 track solo. Set the locators in the Transport window. Press *Bounce* in the Output Object (in the Audio Mixer). Select *Bounce & Add*. After the Bounce procedure, drag the resulting file from the Audio window into the Arrange window, onto a stereophonic audio track. Save the ES2 setting. Mute the bounced Audio Instrument track. Don't delete anything. Save the song. If you want to change the notes, or tempo, recording level, or sound, repeat the entire procedure.

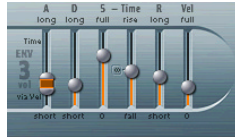
You can also use Logic's Freeze facility to perform individual offline bounce processes for each track. To Freeze, simply click on the *Freeze* button (the ice crystal) of the desired Audio Instrument track(s) in the Arrange window track list. The next time you hit Play, Logic will Freeze the tracks, saving massive amounts of processing power.

Dynamic Stage (Amplifier)

The dynamic stage defines the level—which means the perceived volume—of the played note. The change in level over time is controlled by an *envelope generator*.

ENV3 and the Dynamic Stage

ENV3 is hard-wired to the dynamic stage—envelope generator 3 is always used for control over the level of the sound. For detailed explanations of the envelope parameters, see “The Envelopes (ENV 1—ENV 3)” section, on page 268.



Router Modulation Target: Amp

The dynamic stage can be modulated by any modulation source in the *Router*. The modulation *target* is called *AMP* in the *Router*.

Note: If you select *AMP* as the *target*, *LFO1* as the *Source*, and leave *via* set to *Off* in the *Router*, the level will change periodically, based on the current *Rate* of the LFO—and you’ll hear a tremolo.

Sine Level



Sine Level allows the mixing of a sine wave (at the frequency of Oscillator 1) directly into the dynamic stage, independent of the filters. Even if you have filtered away the basic partial tone of Oscillator 1 with a highpass filter, you can reconstitute it through use of this parameter. Please note:

- In cases where Oscillator 1 is frequency modulated by Oscillator 2 (if you have turned up *FM* with the waveform selector), *only* the pure sine wave is mixed into the dynamic section, not the distorted FM waveform.
- Low frequency modulations of Oscillator 1’s pitch, set in the *Router*, affect the sine wave frequency mixed in here.

Note: *Sine Level* is well suited for adding warmth and a fat bass quality to the sound. Thin sounds can be fattened up with this function, given that Oscillator 1 actually plays the basic pitch.

The Router

The ES2 features a modulation matrix, called the *Router*. Any modulation *Source* can be connected to any modulation *Target*—much like an old-fashioned telephone exchange or a studio patchbay. The modulation intensity—how strongly the *Target* is influenced by the *Source*—is set with the associated vertical slider.

Note: To set the modulation intensity to zero, just click on the little zero symbol (the small circle) right beside *via*.

The intensity of the modulation can also be modulated: The *via* parameter determines yet another modulation source, which defines the amount or intensity of the modulation.

Ten such modulations of *Source*, *via*, and *Target* can take place simultaneously, in addition to those which are hard-wired outside the Router. The *bypass (b/p)* parameter allows the disabling/enabling of individual modulation routings without losing settings.

Note: The bypass parameter is not available in Logic 5.x. If you would like to exchange ES2 settings that utilize bypassed modulation routings with Logic 5.x users, you should set these *Targets* to *off*.

Note: Some modulations aren't possible, due to technical reasons. As an example, the envelope times can be modulated by parameters that are only available during a note-on message. Therefore, there are situations where the envelopes are not available as *Targets*. Furthermore, the LFO 1 can't modulate its own frequency. Values that are not available are grayed out.

Note: You may need to switch from the Vector Envelope display to that of the Router.

The Range of Via

The modulation intensity is set with the vertical slider. This is self-explanatory, as long as the *via* parameter is set to *off*. This ensures that the modulation intensity is constant, if not affected by any other controller (such as the modulation wheel or aftertouch).

As soon as you select a value other than *off* for *via*, the slider is divided into two halves. The lower half defines the minimum intensity of the modulation, when the *via* controller is set to its minimum value. The upper half defines the maximum modulation intensity when the *via* modulator (the modulation wheel, in this case) is set to its maximum value. The area between the two slider halves defines the range that is controlled by the *via* controller.

You can grab the area between the two slider halves with the mouse and drag both halves simultaneously. If this area is too small to be grabbed with the mouse, just click on a free part of the slider track and move the mouse up or down to move the area.

In the example below, the lower half of the slider knob defines the vibrato intensity when the modulation wheel is turned down. The upper half defines the vibrato intensity that takes place when the modulation wheel is set to its maximum value.



Note: To invert the effect of the *via* modulation source, simply activate the *Via invert (inv)* parameter in the Router.

A Modulation Example

Say you've chosen these settings:

- *Target* = *Pitch 123*
- *via* = *Wheel*
- *Source* = *LFO1*
- Modulation intensity = slider position, set as desired

In this configuration, the modulation *source*—*LFO1*—is used to modulate the frequency (pitch) of all three Oscillators (*Pitch 123*). (*Pitch 123*) is the modulation *target* in this example. You'll hear a vibrato (a modulation of the pitch) at the speed of LFO 1's *Rate*. The *modulation intensity* is controlled by the (modulation) *wheel*, which is determined by the *via* parameter. This provides you with control over the depth of vibrato (pitch modulation) via the modulation wheel of your keyboard. This type of configuration is well-known in countless sound settings (patches).

It does not matter which of the ten *Router Channels* you use.

You can select the same *target* in several *Router Channels*, in parallel. You can freely use the same *Sources* as often as you like, and the same *via* controllers can be set in one or multiple *Router* channels.

Modulation Targets

The following *targets* are available for real-time modulation.

Note: These modulation targets are also available for the X and Y axes of the X/Y modulator (the Square). See "The Square" section, on page 272.

Pitch 123

This *target* allows the parallel modulation of the frequencies (pitch) of all three oscillators. If you select an LFO as the *source*, this *target* leads to siren or vibrato sounds. Select one of the envelope generators with zero *attack*, short *decay*, zero *sustain*, and short *release* as the *source* for tom tom and kick drum sounds.

Pitch 1

This *target* allows modulation of the frequency (pitch) of Oscillator 1. Slight envelope modulations can make the amount of detuning change over time, when Oscillator 1 is sounding in unison with another (unmodulated) Oscillator. This is useful for synthesizer brass sounds.

Pitch 2

This *target* allows modulation of the frequency (pitch) of Oscillator 2.

Pitch 3

This *target* allows modulation of the frequency (pitch) of Oscillator 3.

Detune

This *target* controls the amount of detuning between all three oscillators.

Note: The sensitivity of all the pitch modulation *targets* described above depends on the modulation intensity. This sensitivity scaling allows for very delicate vibrati in the cent range (one cent equals 1/100 semitone), as well as for huge pitch jumps in octave ranges.

- Modulation intensity from 0 to 8: steps are 1.25 cents.
- Modulation intensity from 8 to 20: steps are 3.33 cents.
- Modulation intensity from 20 to 28: steps are 6.25 cents.
- Modulation intensity from 28 to 36: steps are 12.5 cents.
- Modulation intensity from 36 to 76: steps are 25 cents.
- Modulation intensity from 76 to 100: steps are 100 cents.

This leads to the following rules of thumb for modulation intensity settings.

- Modulation intensity of 8 equals a pitch shift of 10 cents.
- Modulation intensity of 20 equals a pitch shift of 50 cents, or one quarter tone.
- Modulation intensity of 28 equals a pitch shift of 100 cents, or one semitone.
- Modulation intensity of 36 equals a pitch shift of 200 cents, or two semitones.
- Modulation intensity of 76 equals a pitch shift of 1,200 cents, or one octave.
- Modulation intensity of 100 equals a pitch shift of 3,600 cents, or three octaves.

OscWaves

Dependent on the *Waveforms* set in the three Oscillators, this *target* can be used to modulate:

- the pulse width of rectangular and pulse waves,
- the amount of frequency modulation (Oscillator 1 only),
- Noise color (Oscillator 3 only)
- and the position of the Digiwaves.

OscWaves affects all oscillators simultaneously. The *Osc1Wave*, *Osc2Wave*, and *Osc3Wave* targets only affect the specified Oscillator. Check out the ensuing paragraphs to see what wave modulation does in the three Oscillators.

For further information on the effects of these modulations, please read the “Pulse Width Modulation” section, on page 236. Also take a look at the “Linear Frequency Modulation” section, on page 234, “White and Colored Noise (Oscillator 3 only)”, on page 240, and the “Digiwaves” section, on page 233.

Osc1Wave

Dependent on the *waveform* selected, you can control the pulse width of rectangular and pulse waves of Oscillator 1, the amount of frequency modulation (with Oscillator 1 being the carrier and Oscillator 2 being the modulator), or the position of the Digiwave. The pulse width of the rectangular and pulse waves is not restricted to two fixed values in Oscillator 1.



Note: In classic FM synthesizers, the amount of FM is controlled in real time by velocity sensitive envelope generators. Select one of the *ENVs* as the *source* for such sounds.

Osc2Wave

As per *Osc1Wave*, except that Oscillator 2 does *not* feature FM. Please note that pulse width modulation also works with both the synchronized rectangular and ring modulated rectangular waves.



Osc3Wave

Oscillator 3 is as per *Osc1Wave* and *Osc2Wave*, but it does not feature FM or ring modulation. Oscillator 3 features *Noise*, the color of which can be modulated with this parameter.



OscWaveB

The transitions between Digiwaves during a wavetable modulation are always smooth. Depending on the modulation intensity, an additional *OscWaveB* target can be used to continuously modulate the shape of the transitions from smooth to hard. It applies to all Oscillators.

Osc1WaveB

If wavetable modulation is active for a DigiWave using *Osc1Wav*, you can use this target to modulate the shape of the transition.

Osc1 FM mode: When compared with the hardwired Osc1-FM and the *Osc1Wave* modulation target, the *Osc1WaveB* modulation target offers much higher FM intensities.

Osc2WaveB

If wavetable modulation is active for a Digiwave using *Osc2Wav*, you can use this target to modulate the shape of the transition.

Osc3WaveB

If wavetable modulation is active for a Digiwave using *Osc3Wav*, you can use this target to modulate the shape of the transition.

SineLevl

SineLevl (Sine Level) allows modulation of the *sine wave* level of Oscillator 1, which can be mixed directly into the input of the dynamic stage—without being affected by the filters. The parameter defines the level of the first partial tone of Oscillator 1. See the “Sine Level” section, on page 252.

OscLScle

OscLScle (Osc Level Scale) allows modulation of the levels of *all three* oscillators simultaneously. A modulation value of 0 mutes all oscillators, while a value of 1 raises the gain of the entire mix by 12 dB. The modulation is applied *before* the overdrive stage, allowing for dynamic distortions.

Osc1Levl

Osc1Levl (Osc 1 Level) allows modulation of Oscillator 1’s level.

Osc2Levl

Osc2Levl (Osc 2Level) allows modulation of Oscillator 2’s level.

Osc3Levl

Osc3Levl (Osc 3Level) allows modulation of Oscillator 3’s level.

Cutoff 1

This *target* allows modulation of the *Cutoff Frequency* of Filter 1. See the “Cutoff and Resonance” section, on page 246.



Resonance 1 (Reso 1)

This *target* allows modulation of the *Resonance* of Filter 1. See the “Cutoff and Resonance” section, on page 246.

Cutoff 2

This *target* allows modulation of the *Cutoff Frequency* of Filter 2.



Resonance 2 (Reso2)

This *target* allows modulation of the *Resonance* of Filter 2.

LPF FM

A sine signal, at the same frequency as Oscillator 1, can modulate the *Cutoff frequency* of Filter 2 (which always works as a low pass filter). LPF FM (Lowpass Filter Frequency Modulation) allows the modulation of Filter 2's FM modulation intensity. This filter FM parameter is described in the “Filter 2 FM” section, on page 250.

Cut 1+2

Cut 1+2 (Cutoff 1 and 2) modulates *both* filter's *Cutoff* frequencies in parallel, much like applying the same modulation to Cutoff 1 and Cutoff 2 in two *Router* channels.

Cut1inv2

Cut1inv2 (Cutoff 1 normal and Cutoff 2 inverse) simultaneously modulates the *Cutoff* frequencies of the first and second filters *inversely* (in opposite directions). Put another way, while the first filter's *Cutoff* frequency is rising, the *Cutoff* of the second filter will fall—and vice versa.

Note: In cases where you have combined Filter 1, defined as a highpass filter, and Filter 2 (which always works in lowpass mode) in *Serial* mode, both will serve as a bandpass filter. Modulating the *Cut1 inv 2* target will result in a modulation of the bandpass filter's bandwidth in this scenario.

FltBlend

FltBlend (Filter Blend) modulates the *FilterBlend* (the cross-fading of the two filters), as described in the “Filter Blend and Signal Flow” section, on page 243.

Note: If *FilterBlend* is set as a *target* in one or several *Router* channels, the modulation data for *both* filters will be calculated—even if the *FilterBlend* parameter is set to 1.0 or +1.0. As such, we advise caution when choosing *FilterBlend* as a modulation target because it may increase the need for processing power.

Amp

This target modulates the dynamic stage, or level (the voice's volume). If you select *Amp* as the *target* and modulate it with an *LFO* as the *Source*, the level will change periodically, and you will hear a tremolo.

Pan

This target modulates the panorama position of the sound in the stereo spectrum. Modulating *Pan* with an *LFO* will result in a stereo tremolo (*auto panning*).

In Unison Mode, the panorama positions of all voices are spread across the entire stereo spectrum. Nevertheless, pan can still be modulated, with positions being moved in parallel.

Lfo1Asym

Lfo1Asym (Lfo1 Asymmetry) can modulate the selected waveform of LFO 1. In the case of a square wave, it changes its pulse width. In the case of a triangle wave, it sweeps between triangle and sawtooth. In the case of a sawtooth wave, it shifts its zero crossing.

Lfo1Curve

This target modulates the waveform smoothing of the square and random wave. In the case of a triangle or sawtooth wave, it changes their curves between convex, linear, and concave.

Scaled Modulations

All following modulation *targets* result in a *scaled* modulation, which means that the modulation value isn't simply *added* to the *target* parameter's value, but rather that the *target* parameter value will be *multiplied*. This works as follows: a modulation value of *0.0* results in no effect, while a modulation value of *+1.0* equals a multiplication by 10, and a modulation value of *-1.0* equals a multiplication by 0.04.

LFO1Rate

This target modulates the frequency (speed, rate) of LFO 1.

Note: Say you've created a vibrato with another *Router* channel by modulating the *Target Pitch 123* with *LFO 1*. If desired, you can have LFO 1's speed (the vibrato speed) automatically accelerated or slowed down. To do so, modulate the *LFO1Rate* target with one of the envelope generators (*ENV*). Select *LFO 2* as a *source* and reduce its *Rate*, in order to periodically accelerate and slow down the vibrato.

Env2Atck

Env2Atck (Envelope 2 Attack) modulates the *Attack* time of the second envelope generator.

Env2Dec

Env2Dec (Envelope 2 Decay) modulates the *Decay* time of the second envelope generator.

In cases where you've selected *ENV2 Dec* as the *target* and *Velocity* as the *source*, the duration of the decaying note is dependent on how hard you strike the key. Selecting *Keyboard* as the *source* will result in higher notes decaying more quickly (or slowly).

Env2Rel

Env2Rel (Envelope 2 Release) modulates the *Release* time of the second envelope generator.

Env2Time

Env2Time (Envelope 2 All Times) modulates *all* of ENV2's time parameters: *Attack time*, *Decay time*, *Sustain time*, and *Release time*.

Env3Atck

Env3Atck (Envelope 3Atck) modulates the *Attack* time of the third envelope generator.

Env3Dec

Env3Dec (Envelope 3 Decay) modulates the *Decay* time of the third envelope generator.

Env3Rel

Env3Rel (Envelope 3 Release) modulates the *Release* time of the third envelope generator.

Env3Time

Env3Time (Envelope 3 All Times) modulates *all* of Env3's time parameters: *Attack time*, *Decay time*, *Sustain time*, and *Release time*.

Glide

This *target* modulates the duration of the *Glide* (portamento) effect.

Note: If you modulate *Glide* with *Velocity* selected as the *source*, the velocity used to (how hard) strike the key will define how long it takes for the played notes to “find their way” to the target pitch. See the “Glide” section, on page 227.

Modulation Sources

Some modulation sources are *unipolar*, delivering values between 0 and 1. Others are *bipolar*, and output values between —1 and +1. The following modulation sources are available:

LFO1

... LFO 1 is described in “The LFO's” section, on page 265.

LFO2

... LFO 2 is described in “The LFO's” section, on page 265.

ENV1

... Envelope Generator 1 is described in “The Envelopes (ENV 1—ENV 3)” section, on page 268.

ENV2

... Envelope Generator 2 is described in “The Envelopes (ENV 1—ENV 3)” section, on page 268.

ENV3

... Envelope Generator 3 is described in “The Envelopes (ENV 1—ENV 3)” section, on page 268.

Note: Envelope Generator 3 always controls the level of the overall sound.

Pad-X, Pad-Y

These modulation *sources* allow you to define the axes of the Square, for use with the selected modulation *target*. The cursor can be moved to any position in the Square, either manually or controlled by the vector envelope. See “The Square” on page 272 and “The Vector Envelope” on page 273.

Max

If you select *Max* as a *source*, the value of this *source* will permanently be set to +1. This offers interesting options in conjunction with *via*, as the possible values available for *via* control the modulation intensity.

Kybd

Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0). Five octaves below and above, an output value of —1 or +1, respectively is sent.

Note: This could be used to control the *Cutoff frequencies* of the filters in parallel with the keyboard position—as you played up and down the keyboard, the *Cutoff frequencies* would change. Modulate the *Cut 1+2* target with the *Keyboard* source to do so. At a modulation intensity of 0.5, the *Cutoff frequencies* scale proportionally with the pitches played on the keyboard.

Velo

If you select Velo (Velocity), velocity sensitivity serves as modulation *source*.

Bender

The pitch bender serves as a bipolar modulation *source*, if *Bender* is selected. This is also true when the Oscillators' *Bend Range* parameter is set to 0.

ModWhl

The modulation wheel serves as an unipolar modulation source, if *ModWhl* is selected.

Note: For most standard applications, you'll probably use the wheel as the *via* controller. Traditionally, it can be (and is) used for control over the intensity of periodic LFO modulations. Used here, it can be employed for direct, static modulations, such as controlling the Cutoff frequencies of the filters (*Target = Cut 1+2*).

Note: The Least Significant Bit (LSB) controller for the modulation wheel is recognized correctly, as well.

Touch

Aftertouch serves as modulation source. The ES2 reacts to poly pressure (polyphonic aftertouch). It uses the sum of channel pressure and the note-specific poly pressure value.

Note: If you set the *Target* to *Cut 1+2*, the *Cutoff frequencies* will rise and fall, dependent on how firmly you press a key on your touch-sensitive MIDI keyboard *after* the initial keystroke.

Whl+To

The modulation wheel *and* aftertouch serve as modulation sources.

MIDI Controllers A–F

MIDI controllers available in the mod matrix are named Ctrl A–F and can be assigned to arbitrary controller numbers (via the MIDI Controllers Assignment menus at the bottom of the GUI).

Note: Earlier ES2 versions offered the Expression, Breath, and MIDI Control Change Messages 16–19 as modulation sources. These MIDI controllers are the default values for the assignment and guarantee backwards compatibility.

The values of the MIDI Controllers Assignment menus are only updated if the default setting is loaded, or if a setting that was saved with a song is loaded. If you simply step through settings, the assignment will remain unchanged.

The MIDI Controllers Assignment menus allow you to assign your favorite MIDI controllers as Ctrl A, Ctrl B, and so on.

All MIDI Controller Assignment menus feature a *Learn* option. If this is selected, the parameter will automatically be assigned by the first appropriate incoming MIDI data message.

Note: If none of the controller assignments (Ctrl A–F) is assigned to Expression, the Expression CC message (Ctrl #11) controls the output volume.

Note: The “Vector Stick” (Joystick) of the Korg Wavestation synthesizer generates Controllers 16 and 17, for example. If you use this instrument as your master keyboard, you can control any two ES2 parameters directly with its Joystick.

Note: In the MIDI specification, for all controllers from 0 to 31, there also is a LSB-Controller defined (32 to 63). This “Least Significant Bit”-controller allows for a resolution of 14 bit instead of 7 bit. The ES2 recognizes these control change messages correctly; the breath or expression controllers, for example.

RndNO1

RndNO1 (Note On Random1) outputs a random modulation value between -1.0 and 1.0 (same range as an LFO), that changes when a note is triggered or re-triggered. The (random) note-on modulation remains constant throughout the duration of the note until the next note-on trigger.

Note: There is no value change when playing legato, while in legato mode.

RndNO2

RndNO2 (Note On Random2) behaves like Note On Random1 with the exception that it glides to the new random value using the *Glide* time (inclusive of modulation). It also differs from *NoteOnRandom1* in that the (random modulation) value changes when playing legato, while in legato mode.

SideCh

SideCh (Side Chain modulation) uses a Side Chain (tracks, inputs, busses) to create a modulation signal. The Side Chain source can be selected in the upper gray area of the window. It is fed to the internal envelope follower, which creates a modulation value based on the current Side Chain input signal level.

Via—Controlling the Modulation Intensity

Some modulation sources are *unipolar*, delivering values between 0 and 1 . Others are *bipolar*, and output values between -1 and $+1$. The following *sources* may be used to modulate the modulation intensity.

LFO1

The modulation undulates at the speed and waveform of LFO1, which controls the modulation intensity.

LFO2

The modulation undulates at the speed and waveform of LFO2, which controls the modulation intensity.

ENV1

ENV1 controls the modulation intensity.

ENV2

ENV2 controls the modulation intensity.

ENV3

ENV3 (the level envelope) controls the modulation intensity.

Pad-X, Pad-Y

Both axes of the Square (the Vector Envelope) are available as *via sources* as well, enabling you to control modulation intensities with them.

Kybd

Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0). Five octaves below and above, an output value of -1 or $+1$, respectively is sent.

If you select *Pitch 123* as the *target*, modulate it with the *LFO1 source*, and select *Keyboard* as the *via* value, the vibrato depth will change, dependent on key position. Put another way, the vibrato depth will be different for notes higher or lower than the defined *Keyboard* position.

Velo

If you select *Velo (Velocity)* as the *via* value, the modulation intensity will be velocity sensitive—modulation will be more or less intense dependent on how quickly (hard) you strike the key.

Bender

The pitch bender controls the modulation intensity.

ModWhl

If you select *ModWhl (Modulation Wheel)* as the *via* value, the modulation intensity will be controlled by your MIDI keyboard's modulation wheel.

The Least Significant Bit (LSB) Controller for the modulation wheel is recognized correctly, as well.

Touch

If you select *Touch (Aftertouch)* as the *via* value, the modulation intensity will be touch sensitive—modulation will be more or less intense dependent on how firmly you press the key of your touch-sensitive MIDI keyboard *after* the initial keystroke (aftertouch is also known as pressure sensitivity).

Whl+To

Both modulation wheel and aftertouch control the modulation.

MIDI Controllers A–F

MIDI controllers available in the mod matrix are named Ctrl A–F (rather than Expression, Breath, General Purpose 1–4. MIDI Control Change Messages 16–19 are also known as “General Purpose Slider 1/2/3/4”) and can be assigned to arbitrary controller numbers (via menus at the bottom of the GUI). The default values for the assignment guarantee backwards compatibility. The values of these assignment menus are only updated if the default setting is loaded, or if a setting that was saved with a song is loaded. If you simply step through settings, the assignment will remain unchanged.

This facility is especially helpful if you've always wanted to use Controller #4 (foot pedal), for example, as a modulation source. This feature allows you to assign your favorite MIDI real-time controllers as Ctrl A, Ctrl B, and so on.

All parameters that allow you to select a MIDI controller feature a *Learn* option. If this is selected, the parameter will automatically be assigned by the first appropriate incoming MIDI data message.

Note: As the new entry is added to the top of the list, existing automation data is increased by one. Should further assignments be made, this will increment by one for each entry.

Note: If none of the controller assignments (Ctrl A–F) is assigned to “Expression,” the Expression CC message (Ctrl #11) controls the output volume.

Note: The “Vector Stick” (Joystick) of the Korg Wavestation synthesizer generates Controllers 16 and 17, for example. If you use this instrument as your master keyboard, you can control modulation intensities directly with its Joystick.

Note: In the MIDI specification, for all controllers from 0 to 31, there also is a LSB-Controller defined (32 to 63). This “Least Significant Bit”-controller allows for a resolution of 14 bit instead of 7 bit. The ES2 recognizes these control change messages correctly, for instance the controllers for breath or expression.

RndN01

RndN01 (NoteOnRandom1) controls the modulation intensity (See RndN01 on page 263).

RndN02

RndN02 (NoteOnRandom2) controls the modulation intensity (See RndN02 on page 263).

SideCh

A Side Chain source (tracks, inputs, busses) is used to create the modulation signal.

The LFO's

LFO is the abbreviated form of *Low Frequency Oscillator*. In an analog synthesizer, LFO's deliver modulation signals below the audio frequency range—in the bandwidth that falls between 0.1 and 20 Hz, and sometimes as high as 50 Hz. LFO's serve as modulation sources for periodic, cyclic modulation effects. If you slightly modulate the pitch of an audio oscillator at a rate (speed, LFO frequency) of, say, 3–8 Hz, you'll hear a vibrato. If you modulate the *cutoff frequency* of a lowpass filter, you'll hear a wah wah effect, and modulating the dynamic stage results in a tremolo.

The ES2 features two LFO's, the outputs of which are available as *sources* in the *Router*.

- LFO 1 is polyphonic, which means that if used for any modulation of multiple voices, they will *not* be phase-locked. Furthermore, LFO 1 is key-synced: Each time you hit a key, the LFO 1 of this voice is started from zero. To explain, when used on polyphonic input (a chord played on the keyboard) the modulation is independent for each voice (note). Where the pitch of one voice may rise, the pitch of another voice might fall and the pitch of a third voice may reach its minimum value.



- LFO 2 is monophonic, which means, that the pitch of all voices will rise and fall synchronously, if you modulated the *Pitch123* target with the *LFO2* source, for example.



Both LFO's feature a number of waveforms. LFO 1 can fade in or out automatically, without the need to engage a separate envelope generator. The LFO parameters are detailed below:

EG (LFO1)

At its center position—which can be accessed by clicking the middle mark—the modulation intensity is static: it won't be faded in or out at all. At positive values, it is faded *in*. The higher the value, the longer the delay time is. At negative values, it is faded *out*. The lower (onscreen) the slider is positioned, the shorter the fade out time is.

The function is abbreviated as *EG* because the fading in or out is internally performed by an ultra-simple Envelope Generator.

Note: Chaotic and fast modulations of the oscillator(s) frequencies (*Pitch123*) by *LFO 1* with a *delayed Sample&Hold* selected as the *waveform*, a high *Rate*, and short fade out, make the attack phase of the note sound Moog “Rogue-ish”—and quite similar to the attack phase of brass instruments.

Note: Most commonly, this is used for delayed vibrato—many instrumentalists and singers intonate longer notes this way. To set up a delayed vibrato: Place the slider at a position in the upper half (*Delay*) and modulate the *Pitch123* target with the *LFO1* source. Set a slight modulation intensity. Select a *Rate* of about 5 Hz and the *triangular* wave as the *LFO waveform*.

Rate

This parameter defines the frequency or speed of the modulation. The value is displayed in Hertz (Hz) beneath the slider.

Wave

This is where you select the desired LFO waveform. Check out the waveforms while a modulation of *Pitch123* is engaged and running. You should find the symbols quite self-evident.

Triangular Wave

The triangular wave is well suited for vibrato effects.

Sawtooth Wave and Inverted Sawtooth

The sawtooth is well suited for helicopter and space gun sounds. Intense modulations of the oscillator(s) frequencies with a negative (inverse) sawtooth wave leads to bubbling and boiling, underwater sounds. Intense sawtooth modulations of lowpass filters (such as Filter 2) create rhythmic effects.

Rectangular Waves

The rectangular waves make the LFO periodically switch between two values. The upper rectangular wave switches between a positive value and zero. The lower wave switches between a positive and a negative value set to the same amount above/below zero.

Note: An interesting effect you may wish to try out is achieved by modulating *Pitch123* with a suitable modulation intensity that leads to an interval of a fifth. Choose the upper rectangular wave to do so.

Sample & Hold

The two lower waveform settings of the LFO's output *random* values. A random value is selected at regular intervals, defined by the LFO rate. The upper waveform delivers exact steps of randomization. At its lower setting, the random wave is smoothed out, resulting in fluid changes to values.

Note: A random modulation of *Pitch123* leads to the effect commonly referred to as a random pitch pattern generator or sample and hold. Check out very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies!

Note: The term *Sample & Hold* (abbreviation—*S & H*) refers to the procedure of taking samples from a noise signal at regular intervals. The voltage values of these samples are then *held* until the next *sample* is taken. When converting analog audio signals into digital signals, a similar procedure takes place: Samples of the voltage of the analog audio signal are taken at the rate of the sampling frequency.

Rate (LFO 2)

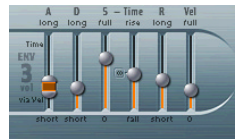
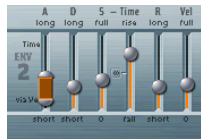
The *LFO2 Rate* (frequency) control allows for the free (in the upper half of the slider range) or song-tempo synchronized (in the lower half of the slider range) running of LFO 2. The rate is displayed in Hertz or rhythmic values, dependent on whether the song tempo synchronization is engaged or not. Rates range from speeds of 1/64-notes through to a periodic duration of 32 bars. Triolic and punctuated values are also possible. LFO 2 is ideally suited for rhythmic effects which retain perfect synchronicity, even during tempo changes to the song.

The Envelopes (ENV 1—ENV 3)

In addition to the complex Vector Envelope, described in “The Vector Envelope” section, on page 273, the ES2 also features three envelope generators per voice. On both the front panel and as a *source* in the *Router*, they are abbreviated as ENV 1, ENV 2 and ENV 3, respectively.

Note: The roots of the term *envelope generator* and its basic functionality are described in the “Envelopes” section, on page 199.

The feature sets of ENV 2 and ENV3 are identical. ENV 3 defines the changes in level over time for each note played. You can regard ENV 3 as being “hard-wired” to the *Router’s AMP* modulation target.



The parameters of ENV 2 and ENV 3 are identical—but ENV 3 is always used for control over level.

Unlike many other synthesizers, there is no hard-wired connection between any of the envelope generators and the cutoff frequencies of the ES2 filters. Modulation of the cutoff frequencies *must* be set separately in the *Router*. In the default setting, this is already the case—in the *Router* channel just below the *Filter* (see graphic).



Note: Set up a *Router* channel as follows, in order to establish this type of modulation: Set *target* to *Cutoff 1*, *Cutoff 2* or *Cut 1+2*, set *source* to, say, *ENV 2*. Once set as described, the slider of the *Router* channel will function as the filter’s *EG Depth* parameter.

Note: Both *ENV 2* and *ENV 3* are velocity sensitive, making it unnecessary to set *via* to *Velo* in the *Router* channel: You can leave *via* switched off.

The Parameters of ENV 1

At first glance, ENV 1 appears to be rather poorly equipped. Its few parameters, however, are useful for a vast range of synthesizer functions.

Trigger Modes: Poly, Mono, Retrigger

In *Poly* mode, the envelope generator behaves as you would expect on any polyphonic synthesizer: Every voice has its own envelope.



In *Mono* and *Retrig* modes, a single envelope generator modulates all voices in parallel—identically, in other words.

- If ENV 1 is set to *Mono*, all notes must be released before the envelope can be triggered again. If you play legato, or any key remains depressed, the envelope won't start its attack phase again.
- In *Retrig* mode, the envelope will be triggered by any key you strike, no matter whether other notes are sustained or not. Every sustained note is affected by the retriggered envelope.

Note: The design of early analog polysynths led to polyphonic instruments where all voices passed through a single lowpass filter. This design was primarily due to cost factors. The best known example of these polyphonic instruments were the Moog Polymoog, the Yamaha SK20 and Korg Poly 800. The sole lowpass filter of such instruments is controlled by a single envelope generator. To simulate this behavior, use the *Mono* or *Retrigger* modes.

Note: Say you've modulated the *Cutoff 2* target with a percussive *source*, such as *ENV1*, which is set to *Retrig*. If you play and sustain a bass note, this note will receive a percussive filter effect every time you hit another key. The newly struck key is also shaped by the same filter. Playing a sound set up in this way feels like you're playing a polyphonic synthesizer with *one* filter. This is despite the fact that the ES2's filters remain polyphonic and can be simultaneously modulated by different polyphonic sources.

Note: If you want to simulate the percussion of a Hammond Organ, you will also need to use the *Mono* or *Retrigger* modes.

Decay/Release

ENV 1 can be set to act as an envelope generator with an *Attack time* and *Decay time* parameter or with an *Attack time* and *Release time* parameter.

Switching between both modes is achieved by clicking on the *D* or the *R* above the right ENV 1 slider.

- In its *Attack/Decay* mode, the level will fall to zero after the attack phase has completed, no matter whether you sustain the note or not. It will decay at the same speed even if you release the key. The decay time is set with the *Decay time* slider, abbreviated as *D*.
- In its *Attack/Release* mode, the envelope level remains at its maximum after the attack phase is over, as long as the key remains depressed. Following the release of the key, its level decreases over the time set with the *R* slider—the abbreviation for *Release time*.

Attack Time and Attack via Vel

The *Attack time* slider is divided into two halves. The lower half defines the attack time when the keys are struck hard (at maximum velocity). The upper half defines the attack time at minimum velocity.

You can grab the area between the two slider halves with the mouse and drag both halves simultaneously. If this area is too small to be grabbed with the mouse, just click in a free part of the slider track, and move the mouse up or down to move the area.

The Parameters of ENV 2 and ENV 3

The feature sets of ENV 2 and ENV 3 are identical, but it is always the task of ENV 3 to define the *level* of each note—to modulate the dynamic stage. ENV 3 is available for simultaneous use as a *source* in the *Router* as well. The envelope's *time* parameters can also be used as modulation *targets* in the *Router*.

Note: See the “Envelopes” section, on page 199 for information on the basic functionality and meaning of envelope generators.

Attack Time

As per the *Attack* slider of ENV 1, the *Attack time* sliders of ENV 2 and ENV 3 are divided into two halves. The lower half defines the attack time when the keys are struck at maximum velocity. The upper half defines the attack time at minimum velocity.

You can grab the area between the two slider halves with the mouse and drag both halves simultaneously. If this area is too small to be grabbed with the mouse, just click in a free part of the slider track, and move the mouse up or down to move the area.

Decay Time

The *Decay time* parameter defines the length of time it takes for the level of a sustained note to fall to the *Sustain level*, after the attack phase is over. If *Sustain level* is set to its *maximum*, the *Decay* parameter has no effect. When the *Sustain level* is set to its *minimum value*, *Decay* defines the duration or fade-out time of the note.

The *Decay* parameter appears as a modulation *target* in the *Router* for ENV 2 and ENV 3 individually (*ENV2Dec*, *ENV3Dec*).

Note: On pianos and plucked string instruments, high notes decay faster than low notes. In order to simulate this effect, modulate the *Decay Time* target with the *Kybd* source in the *Router*. The *Router* channel slider should be set to a negative value.

Sustain and Sustain Time

When the *Sustain Time* parameter is set to its center value—which can be achieved by clicking the ∞ symbol—the *Sustain level* behaves like the *Sustain* parameter of any synthesizer ADSR envelope. In this position, the *Sustain level* (abbreviated as *S*) defines the level that is sustained for as long as the key remains depressed, following the completion of the *Attack time* and *Decay time* phases.

The *Sustain Time* slider defines the time it takes for the level to rise to its maximum—or to fall to zero—after the decay phase is over. Settings in the lower half of its range (*Fall*) determine the speed at which the level decays from the *Sustain level* to zero. The lower the slider position, the faster the decay speed. Settings in the upper half of its range (*Rise*) determine the speed at which the level rises from the *Sustain level* to its maximum value. The higher the slider position, the faster the speed of the rise.

Release Time

As with any synthesizer ADSR envelope, the *Release time* parameter (*R*) defines the time the level takes to decay to zero after the key is released.

Vel

The *Vel* (*Velocity Sensitivity*) parameter defines the velocity sensitivity of the entire envelope. If it is set to *maximum*, the envelope will only output its maximum level when the keys are struck at maximum velocity.

The Square

The Square has two axes: The X and Y axes have positive and negative value ranges. They are bipolar, in other words. By touching and moving the cursor with the mouse, the values of both axes are continuously transmitted. As you can modulate one freely selectable parameter with the X value, and another freely selectable parameter with the Y value, you can use the mouse like a Joystick.



As an alternative to this real-time control, the position of the cursor can be modulated by the *Vector Envelope*—just like the mix between the three Oscillators in the *Triangle*. The *Loop* function of the Vector Envelope generator allows for cyclic movements. This opens a number of doors, allowing it to operate as a two-dimensional, luxurious pseudo-LFO with a programmable waveform. More on this is found in “The Vector Envelope” section, on page 273.

Vector Mode

Vector Mode allows you to disable the control of the Square cursor by the vector envelope. The same parameter also defines whether or not the Triangle (the oscillator mixer) shall be controlled by the vector envelope.

- *Vector Mode Off*—The vector envelope does not influence the Triangle or the Square. It's simply switched off. This allows you to set and control the cursors of the Triangle and the Square in real-time.
- *Vector Mode Mix*—The vector envelope controls the Triangle (the oscillator mix), but not the Square.
- *Vector Mode XY*—The vector envelope controls the Square, but not the Triangle.
- *Vector Mode Mix+XY*—The vector envelope controls both the Square and Triangle.

Note: Like all of the ES2's parameters, the movements of the cursors in the Triangle and Square can be recorded and automated by Logic. This automation data can be edited and looped in Logic. This is completely independent of the cyclic modulations of the Vector Envelope. Vector modulation of the Square and Triangle should be disabled for this type of use (*Vector Mode = off*).

Vector Target—Modulation Destinations

The *Vector X* and *Vector Y target* parameters determine the effect of cursor movements in the Square. The modulation targets are identical to those available in the *Router*, so we won't repeat ourselves here. Please see the “Modulation Targets” section, on page 254 for descriptions. The position of the cursor in the *Square* is also available in the *Router*, as the *Pad-X* and *Pad-Y Source* and *Via* options.

Vector Int—The Modulation Intensity

The maximum intensity, sensitivity, *and* polarity of the modulation is set with the *Vector X Int* and *Vector Y Int* parameters.

The Vector Envelope

The *Triangle* and *Square* are the most special and unusual elements of the ES2's graphical user interface. Whilst the Triangle controls the mix of the three oscillators, the X and Y axes of the Square can modulate any (modulation) target.



The *Vector Envelope* can control the movement of the cursors in the Triangle and the Square in real-time. Each voice is equipped with its own Vector Envelope, which is triggered from its startpoint with every new keystroke (or with every MIDI note-on message, to be more precise).



The concept of the Vector Envelope, the Square and the Triangle may seem strange at first glance, but believe us: Combined with the other synthesis possibilities of the ES2, and you will end up with sounds that are truly unique and literally, moving.

Envelope Points, Times, and Loops

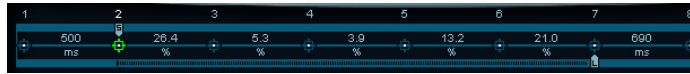
The Vector Envelope consists of up to 15 points on the time axis. Each point can control the position of both the Triangle and Square's cursors.

The points are numbered sequentially. Point 1 is the starting point. In order to edit a point, simply select it—by clicking on it.

Sustain Point

Any point can be declared the *Sustain Point*. Given that the note played is sustained long enough and there's no loop engaged, any envelope movement will stop when this Sustain Point is reached. It will be sustained until the key is released (until the MIDI note-off command).

To define a point as the *Sustain Point*, click on the turquoise strip above the desired point. The selected point will be indicated by an *S* between the point and its number, on the turquoise strip.



Loop Point

Any point can be declared Loop Point. Given that the note is sustained long enough, the envelope can be repeated in a loop.

The looped area is the time span between Sustain Point and Loop Point. In between, you can define several points which describe the movements of the Square and Triangle cursors.

In order to define a point as the Loop Point, click on the turquoise strip below the desired point. A Loop Point is indicated by an *L* in the strip below.



Note: In order to see and define the Loop Point, the loop must be activated. See the "Loop," on page 277.

Note: With loop activated, the Vector Envelope works like a multi-dimensional, polyphonic LFO with a programmable waveform.

Vector Envelope Times

With the exception of the first point, which is tied to the beginning of each played note, every point has a *Time* parameter. This parameter defines the period of time required for the cursor to travel from the point which preceded it. The times are normally displayed in milliseconds (ms).

To adjust a time value, you can click directly on the numerical value and use your mouse as a slider.

Default Setting of the Vector Envelope

The default setting of the Vector Envelope consists of three points. Point 1 is the startpoint, point 2 is defined as the Sustain Point, and point three is the end point, by default.

The impact of the Vector Envelope on the *Oscillator Mix* or on the Square is switched off by default. This allows the ES2 to behave as a synthesizer without a Vector Envelope generator. This traditional starting point is more convenient when creating patches from scratch.

There are two ways to switch off the Vector Envelope:

- You can switch *on* the *Solo Point* parameter (described on page 276). If it is *on*, only the Triangle and Square cursor positions of the currently selected point are active.
- You can disable the Vector Envelope altogether (or only the Triangle or Square) as described in the “Vector Mode” on page 272.

Setting and Deleting Points

The more points you set, the more complex the Vector Envelope movements that can be designed. You can:

- Create a new point by Shift-clicking between two existing points. The segment that previously existed between the two old points is divided at the mouse position. The sum of the two new segment times is equal to the time used by the old undivided segment. As such, the ensuing points retain their absolute time positions. In addition, the existing cursor positions in the *Triangle* and *Square* are fixed, ensuring that the creation of new points does not affect any previously defined movements.
- Delete points by clicking on them while holding Control.

Setting Vector Envelope Times

By clicking a time value and moving the mouse, you will alter the envelope time—the time it takes for the Vector Envelope to travel from the point before this time value to the point after this time value. You have two ways of doing this.

- Simple vertical dragging of the time parameter results in reaching all following points later (or sooner, respectively) in time.
- Dragging with Control held, you will shorten or lengthen the time of the following point by the same amount. This ensures that the adjacent, and all following, points retain their absolute time positions.

Resetting the Values of a Point

Reverting to default cursor positions in the Triangle and the Square is done by:

- Clicking in the *Triangle* while holding down the Option key. This sets all Oscillators to output the same level. The cursor is set to the middle of the Triangle.
- Clicking onto the *Square* while holding down the Option key sets the cursor to the center of the Square. Both axis values are set to zero.

Solo Point

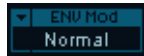
This function basically switches off the entire Vector Envelope generator. If *Solo Point* is set to *on*, no dynamic modulations are applied by the Vector Envelope. In this scenario, the currently visible cursor positions of the *Triangle* and *Square* are permanently in effect. These cursor positions match the currently selected Vector Envelope point.

If you select another Vector Envelope point (by clicking on it), you will engage its *Triangle* and *Square* cursor positions immediately. If *Solo Point* is set to *on*, the newly selected point will become the solo point.

Note: You can independently switch off the vector modulation of the *Square* by setting *Vector Mode* off, as outlined on page 272.

Env Modes Normal and Finish

If *Env Mode* is set to *normal*, the release phase (the phase after the sustain point) will begin as soon as you release the key (note off). The release phase will start from the Vector Envelope point where you released the key.



- If the loop is switched off, and the Vector Envelope reaches the Sustain Point *S*, the Sustain Point *S* will be played for as long as you hold the key.
- If the loop is switched on (see “Loop”, on page 277), and the Loop Point *L* is positioned before the Sustain Point *S*, the loop will be played for as long as you hold a key.
- If the loop is switched on, and the Loop Point *L* is positioned after the Sustain Point *S*, the loop will be played after the release of a key.

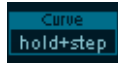
In *Env Mode finish*, the Vector Envelope will not immediately commence the release phase when you release the key. Rather, it will play all points (for their full duration) until the last point is reached, regardless of whether you hold the key or release it.



- If the loop is switched off, the Sustain Point *S* will be ignored. The Vector Envelope will end on its last point, regardless of whether you hold the key or release it.
- If the loop is switched on, the Vector Envelope will play all points until it reaches the Loop Point, and then play the loop for as long as the sound ends. It does not matter if the Loop Point *L* is before or after the Sustain Point *S*.
- If the loop is switched on, and *Loop Count* is set to a value other than *infinite*, the Vector Envelope will continue on to following points after the selected number of loop repeats. If *Loop Count* is set to *infinite*, the number of segments after the loop is irrelevant. See “Loop Count”, on page 279.

Curve

The *Curve* parameter sets the shape of the transition from point to point. You can choose between nine convex and nine concave shapes. There are also the two extreme forms; *hold+step* and *step+hold*, which allow stepped modulation. Where *step+hold* jumps at the beginning of the transition time, *hold+step* jumps at the end.



Note: You can use *hold+step* to create stepped vector grooves with up to 15 steps.

Vector Envelope Loops

The Vector Envelope can—like any envelope—run in one shot (as long as the note is sustained) mode. It can also run several times or in an infinite cycle, much like an LFO. You can achieve this through the use of *loops*.

Note: The loop parameters might remind you of the loop parameters available for samples. Just to avoid any misunderstandings: The Vector Envelope only supplies control signals used for moving the cursor positions of the Triangle and Square. The audio of the ES2 is *not* looped at all.

Loop

The ES2 features these loop options:

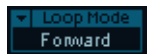
- Off

If *Loop Mode* is switched *Off*, the Vector Envelope runs in one shot mode from its beginning to its end—given that the note is held long enough. The other loop parameters are disabled.



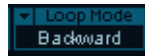
- Forward

When *Loop* is set to *Forward*, the Vector Envelope runs to the Sustain Point and begins to repeat the section between the Loop Point and Sustain Point periodically, always in a forward direction.



- Backward

When *Loop* is set to *Backward*, the Vector Envelope runs to the Sustain Point and begins to repeat the section between the Sustain Point and Loop Point periodically, always in a backward direction.



- Alternate

When *Loop* is set to *Alternate*, the Vector Envelope runs to the Sustain Point and returns to the Loop Point and back to the Sustain Point periodically, alternating in both a backward and forward direction.



Loop Rate

Just as every LFO has a speed (or *Rate*) parameter, the loop can be set to cycle at a defined *Loop Rate*. And just like an LFO, the Vector Envelope *Loop Rate* can be synchronized to the song tempo automatically.



- If you switch the *Loop Rate* to *as set*, the duration of the loop cycle is equal to the sum of the times between the Sustain and Loop Points. Click on the field labeled *as set* (below the *Rate* slider) to select.
- If you set the *Loop Rate* to one of the rhythmic values (*sync*, left half of the slider, 32 bars up to 64th Triplet Note), the *Loop Rate* fits to the song tempo.
- You also can set the *Loop Rate* in the small panel to the right half of the slider (*free*). The value indicates the number of cycles per second. Use the mouse as a slider to adjust.

Note: If *Loop Rate* is not switched to *as set*, and the loop is activated (*Loop Mode Forward, Backward* or *Alternate*), the times of the points between the Loop and Sustain Points as well as the value for *Loop Smooth* are indicated as a percentage of the loop duration, rather than in milliseconds.

Loop Smooth

When *Loop Mode* is set to *Forward* or *Backward*, there will inevitably be a moment when a transition from the Sustain Point to the Loop Point occurs. In order to avoid abrupt cursor position changes, this transition can be smoothed through use of the *Loop Smooth* parameter.

- Set the value by touching and dragging it with the mouse.

- If *Loop Rate* is set to *Sync* or *Free*, the Loop Smoothing Time will be displayed as a percentage of the loop cycle duration.



- If *Loop Rate* is set to *as set*, the Loop Smoothing Time will be displayed in milliseconds (ms).



Loop Count

The loop cycle of the Vector Envelope doesn't need to be performed infinitely—you can have it cycle just a few times. Following the defined number of repetitions, the Vector Envelope will run from the sustain point onwards, just as with *Loop Mode off*.

- Use the mouse as a slider to set the *Loop Count* value. Possible values are 1 to 10 and *infinite*.



Time Scaling

You can stretch and compress the entire Vector Envelope. As an example, to double the Vector Envelope's speed, it's not necessary to halve the time values of every point. All you need to do is set *Time Scaling* to 50%.

- Adjust *Time Scaling* by using the mouse as a slider.



- The range for the *Time Scaling* parameter is from 10% to 1000%. It is scaled logarithmically.
- If the *Loop Rate* is *as set*, the scaling also affects the loop. If not (*Loop Rate* = *free* or *sync*), the setting will not be affected by *Time Scaling*.

Fix Timing—Normalizing Time Scaling and Loop Rate

By clicking *Fix Timing*, the *Time Scaling* value will be multiplied by all time parameters, and *Time Scaling* will be reset to 100%. There will be no audible difference. This is simply a normalizing procedure, most similar to the normalizing function of the Region playback parameters in Logic.



In cases where a loop which was synchronized to the song tempo had been engaged (*Loop Rate = sync*), pressing *Fix Timing* will also switch the *Loop Rate* to *as set*, thus preserving the absolute rate.

Effect Processor

The ES2 is equipped with an integrated effect processor. Any changes to this processor's effects settings are saved as an integral part of each sound program. The entire output of the dynamic stage is processed in true stereo.

Despite the inclusion of this integrated effects processor, please feel free to process the ES2 with the other effect plug-ins included in Logic. The sound and parameter set of the integrated effects unit is reminiscent of classic pedal effects, designed for the electric guitar. The use of guitar pedal effects on classic analog synthesizers was a standard practice amongst gigging musicians.

Distortion

At its *Soft* setting, the distortion circuit is somewhat like a tube overdrive, whilst *Hard* sounds like a fully-transistorized fuzz box. The *Distortion* control defines the amount of distortion, and *Tone* controls the treble portion of the output of the distortion process.



Chorus, Phaser, Flanger

These classic modulation effects and their parameters (*Intensity* and *Speed*) won't need any explanation. These sophisticated algorithms simulate the sound of analog effects of this kind, with one exception: They don't produce as much noise.



Note: A *chorus* effect is based on a delay line, the output of which is mixed with the original, dry signal. The short delay time is modulated periodically, resulting in pitch deviations. The modulated deviations, in conjunction with the original signal's pitch, produce the chorus effect.

Note: The *flanger* works in a fashion similar to that of a chorus, but with even shorter delay times, and the output signal being fed back into the input of the delay line. This feedback results in the creation of harmonic resonances which wander cyclically through the spectrum, giving the signal a metallic sound.

Note: The *phaser* is based on a mix of a delayed and an original signal. The delayed element is derived from an all-pass filter, which applies a frequency-dependent delay to the signal. This is expressed as a phase angle. The effect is based on a comb filter, which is basically an array of inharmonic notches (rather than resonances, as with the flanger), which also wander through the frequency spectrum.

Random Sound Variations

The ES2 offers a unique feature that allows you to vary the sound parameters randomly. You can define the amount of random variation, and can restrict the variations to specific sonic elements. The random sound variation feature will inspire and aid (or occasionally amuse) you when creating new sounds.



Pressing the *RND* button randomly alters the sound. The process is triggered by a single click and can be repeated as often as you like.

Note: To avoid possible misunderstandings: This feature has nothing to do with random real-time modulations. The random feature changes the parameters randomly with each mouse click. Real-time random modulations are performed with the random waveforms of the LFO's and by the *Analog* parameter, for random pitch settings.

RND Int

RND Int (*Random Intensity*) defines the amount of random parameter alteration. As you move the slider to the right, you will increase the amount of random variation.

The random sound variation feature always alters the parameters as they are currently set, not based on the memorized *Setting* file. As such, clicking *RND* repeatedly will result in a sound which increasingly differs from the original setting. If want to check out several slight alterations of the current setting, you can reload the original *Setting*, after each random alteration.

RND Destination

Some aspects of your sound may already be ideal for the sound you had in mind. As such, it may not be desirable to alter them. Say your sound setting has a nice percussiveness, and you'd like to try a few sonic color variations while retaining this percussive feel. To avoid the random variation of any attack times, you can restrict the variation to the oscillator or filter parameters, with the envelope parameters excluded from the variation process. To do so, set the *RND Destination* to *Waves* or *Filters*.

Please note:

- The *Master Level*, *Filter Bypass* as well as the three *Oscillator On/Off* parameters and the *Vector/Router* display options are never randomized.
- During randomizations of the Vector Envelope, the *Solo Point* will always be set to *off*.

You can restrict the random sound variation to the parameter groups listed below:

All

All ES2 parameters, with the exception of the parameters mentioned above, will be altered.

All except Router and Pitch

All ES2 parameters, with the exception of all Router parameters and the basic pitch (*semitone* settings of the oscillators), will be altered. The oscillator fine tuning will be varied. This will result in more musically useful sounds.

All except Vector Env

All ES2 parameters, with the exception of all Vector Envelope parameters, will be altered. This maintains the rhythmic feel of a given setting.

Waves

Only the *Wave* and *DigiWave* parameters of the Oscillators will be altered. The other Oscillator parameters (tuning, mix, and modulations in the *Router*) are excluded.

DigiWaves

DigiWaves will be selected in all Oscillators. The *DigiWave* number will be altered. The other Oscillator parameters (tuning, mix, and modulations in the *Router*) are excluded.

Filters

The filter parameters are varied. The parameters included are: *Filter Structure* (series or parallel wiring), *Blend*, *Filter Mode*, *Cutoff Frequency*, and *Resonance* for Filter 1 and 2, *Fatness*, *Filter FM* of Filter 2.

Envs

All envelope parameters of all three envelopes ENV 1, ENV 2 and ENV 3 are varied. The Vector Envelope is *excluded*.

LFOs

All LFO parameters of all LFOs are varied.

Router

All Router parameters of all Router channels are varied with all *Intensities*, *Target*, *via*, and *Source* parameters.

FX

All effects parameters are varied.

Vector Envelope

All Vector Envelope parameters are varied, including the XY routing of the Square.

Vector Env Mix Pad

The Oscillator mix levels (Triangle cursor positions) of the Vector Envelope points are altered. The rhythm and tempo of the modulation (the time parameters of the points) will not be altered.

Vector Env XY Pad

The Square cursor positions of the Vector Envelope points are altered. The XY routing won't be altered. The rhythm and tempo of the modulation (the time parameters of the points) will not be altered.

Vec Env Times

Only the time parameters of the Vector Envelope points will be altered.

Vec Env Structure

The Vector Envelope structure will be altered: All times, the Sustain Point, the number of points and all loop parameters.

Note: It's recommended that you save any good sounds resulting from the *RND* process, as you work. Do this under a new name (*Setting > Save Setting as...*) in the Plug-in window.

Tutorials

You will find the settings for these tutorials in the Tutorial Settings folder in the settings menu (in the head of the ES2 Plug-in window).

Sound Workshop: Logic ES2

Tutorial Setting: Analog Saw Init

Topics: Sound Design from Scratch, Filter Settings, DigiWaves

This is designed for use as a starting point when programming new sounds from scratch. Professional sound designers like to use such scratch settings when programming entirely new sounds, usually as follows: An un-filtered sawtooth wave sound without envelopes, modulations or any gimmicks. This type of setting is also well-suited to the purpose of getting to know a new synthesizer. It allows you to access all parameters without having to consider any pre-set values.

- Start with the filters, the heart of any subtractive synthesizer. Check out the four lowpass filter types 12 dB, 18 dB, 24 dB and *fat* (Filter 2) with different values for *Cut* (Cutoff Frequency) and *Res* (Resonance). Define Env 2 as filter envelope. This modulation wiring is pre-set in the router.

- Set *Filter Blend* to its left-most position, which will allow you to listen to Filter 1 in isolation. In many circumstances, you'll probably prefer to use Filter 2, but Filter 1 has its advantages. In addition to the lowpass filter with 12 dB/octaves slope (*Lo*), Filter 2 also offers: a highpass, peak, bandpass (*BP*), and band rejection mode (*BR*). Filter 1's lowpass sounds softer in comparison to Filter 2. It is best-suited to sounds where the filter effect is/should be less audible (Strings, FM-Sounds). Distorted TB-303 style sounds are more easily achieved with Filter 1.
- This setting is also ideal for checking out the oscillator waveforms. The analog waveforms can be set in the Editor view. In order to select the DigiWaves, set *Osc 1 Wave* to *DigiWave*.

Tutorial Setting: Analog Saw 3Osc

Topics: Three Detuned Sawtooth Oscillators, Unison Mode

Fat synthesizer sounds have always been popular, and are likely to remain so, given their use in modern trance and techno styles. This setting features three detuned oscillators, and sounds fat as it is. The following will introduce you to some additional tools to fatten the sound even more.

- Check out the 3-oscillator basic sound with different filter and envelope settings.
- Check out the chorus effect at different *Intensities* and *Speeds*.
- Engage *Unison* mode and select a higher setting for *Analog*. As the sound is polyphonic, each note is doubled. The number of notes that can be played simultaneously will be reduced from 10 to 5. This will make the sound rich and broad. Combining *Unison* and higher values for *Analog* will spread the sound across the stereo spectrum.

In many factory settings, the *Unison* mode is active. This demands a lot of processing power. If your computer isn't fast enough, you can switch off the *Unison* mode and insert an Ensemble effect in a bus, for use with several plug-ins. This will save lots of processing power. Another way to save CPU resources is to bounce several Audio Instrument tracks—which place high demands on the processor—to an audio track.

Tutorial Setting: Analog Unison

Topics: Extremely Detuned Monophonic Analog Sounds, Effects

There's nothing fatter than this heavily detuned, un-filtered basic sound. As with the example above, three sawtooth oscillators are used, but they are detuned further. The combination of *Unison* and *Analog* (set to a high value) is essential again, but this time, monophonic mode is used to stack ten voices. Without further effects, the result is an extremely fat lead sound, as used in countless dance and trance productions. With appropriate filter and envelope settings, trendy arpeggio and sequencer sounds can easily be set up.

- Set the *Cutoff Frequency* of Filter 2 to 0. This will activate the preset filter envelope. Feel free to check out different envelope settings.
- Switch *Osc 1* to sound one or two octaves lower.

- Increase *Drive* or *Distortion*.
- Set *Env 2* to be velocity sensitive. This allows for velocity sensitive filter modulations.
- Insert a stereo delay effect in the audio instrument channel strip of the ES2. In order to delay several Audio Instruments, you might prefer to insert the effect into a *Bus*, which is accessed via each channel's *Send*.

Logic incorporates reverb and delay effects which are essential for many synthesizer sounds. These aren't integrated into the ES2, ensuring that no processing power is unnecessarily wasted.

Tutorial Setting: Analog Bass clean

Topic: Clean Bass Settings with One Oscillator Only

Not every sound needs to comprise of several oscillators. There are numerous simple and effective sounds which make use of a single oscillator. This is especially true of synthesizer bass sounds, which can be created very quickly and easily through use of this basic setting.

The basic sound is a rectangular wave, transposed down by one octave. The sound is filtered by *Filter 2*. What's special about this sound is its combination of *Legato* and *Glide* (portamento). As long as you play staccato, no glide effect will occur. If you play legato, the pitch will smoothly glide from one note to another. All keys must be released before striking a new key, in order to retrigger the envelopes.

- Check out different filter and envelope settings.
- Replace the rectangular wave with a sawtooth.
- Vary the *Glide* settings.

Editing works best while a bass line is running. Create a monophonic MIDI Region, with most notes played staccato and some legato. This can provide some interesting results with very long *Glide* values.

Tutorial Setting: Analog Bass distorted

Topic: Distorted Analog Bases

Filter 1 is engaged, with high settings for *Drive* and *Distortion*. This filter is better suited to the creation of distorted analog sounds than *Filter 2*.

- Check out *Filter 2* by setting *Filter Blend* to its right-most position. You'll hear that *Filter 1* works better with distorted sounds.
- In order to control the filter modulation, move the green sliders of the first modulation channel in the router. This controls the modulation intensity.

Tutorial Setting: FM Start

Topic: FM Intensity and Frequency

These are the first steps to take when learning about linear Frequency Modulation (FM) synthesis. You'll first hear an un-modulated sine sound, generated by Oscillator 1. Oscillator 2 is switched on and set to produce a sine oscillation as well, but its level is set to 0: Just push the cursor in the triangle in the uppermost corner.

In the ES2, Oscillator 1 is always the carrier, and Oscillator 2, the modulator. So Oscillator 2 modulates Oscillator 1.

- Adjust the intensity of the frequency modulation by slowly moving the *wave selector* from *Sine* to *FM*. You will hear a typical FM spectrum, with the carrier and modulator set to the same frequency.
- Alter the modulator frequency (Oscillator 2) by adjusting *Fine Tune* from 0 c to 50 c. You'll hear a very slow frequency modulation that can be compared to the effect of an LFO. The frequency modulation, however, takes place in the audio spectrum. It is adjusted in semitone steps by the frequency selector. Check out the entire range from -36 s to +36 s for Oscillator 2. You'll hear a broad spectrum of FM sounds. Some settings will remind you of classic FM synthesizer sounds.
- Select other waveforms for Oscillator 2. *Sine* is the classic, standard FM waveform, but other waveforms lead to interesting results as well, especially the *DigiWaves*.
- You will achieve further interesting results by altering the carrier (Oscillator 1) frequency. Check out the entire range: from -36 s to +36 s semitones here, as well. The odd intervals are especially fascinating. Note that the basic pitch changes when doing so.

Tutorial Setting: FM Envelope

Topic: Controlling FM Intensity by an Envelope and FM scaling

In this setting, you can control the FM intensity with an Envelope, generated by Envelope 2. The modulation target is the range which falls between *Sine* and *FM* in the Oscillator *wave selector*. The first *Router* channel is used for this. You can control a wider range through the use of additional modulations, which have been pre-prepared for you. All you need to do is set their values. As these modulations work without velocity sensitivity, you can set them in the editor view by moving both the lower and upper fader halves to their top-most positions.

- Set the second modulation channel to 1.0. You'll hear how the modulation will now "wander" through a broader sound range.
- Set modulation channels 3 and 4 to a value of 1.0 as well, and listen to the increase in the sound range.

- Following such drastic augmentations to the modulation range, the sound will have become uneven. In the lower and middle ranges, it sounds nice, but in the treble range the FM intensity appears to be too severe. You can compensate for this effect by setting the target *Osc 1 Wave* to be modulated by the keyboard position (*kybd*) in modulation channels 5 and 6. This results in a keyboard scaling of the FM intensity.
- As the sound range is so vast (due to the 4 modulations), two modulation channels are required to compensate for this. Set the lower slider halves to their lowest positions. Good keyboard scaling is essential for any FM sound.

Tutorial Setting: FM Drive

Topic: FM with Drive and Filter-FM

You can dramatically alter the character of FM sounds by applying *Drive* and *Filter FM*. The results are reminiscent of the feedback circuits of classic FM synthesizers.

- Check out different *Drive* and *Filter FM* settings.
- Lower the *Cutoff Frequency* of Filter 2 to 0. Envelope 2 modulates Filter 2. This modulation routing is preset in the setting.

Tutorial Setting: FM DigiWave

Topic: FM with Digiwaves

In this example, a DigiWave is used as an FM modulator. This results in a bell-like spectra out of only two operators. Normally, this type of timbre could only be produced through the use of a larger number of sine oscillators.

In order to create a fatter, undulating, and atmospheric quality to the sound, the polyphonic *unison* mode has been engaged. Filter and amplitude envelopes have been preset to shape the sound.

- Check out the variety of DigiWaves, as FM modulation sources.
- Check out different *Analog* parameter values.

Tutorial Setting: FM Wavetable

Topic: FM with Wavetables

You can program the most vivid FM sounds when the modulation source “morphs” between different Digiwaves. The morphing in this setting is controlled by *LFO 2* in this setting. The tempo of *LFO 2* (and therefore the morph) depends on the sequencer tempo (here: 2 bars).

- Set *LFO 2* to different waveforms. *Lag S/H* (smooth random), in particular, should be fun.
- Check out different FM intensities and Oscillator frequencies.
- Alter the modulation intensity of the first modulation channel (*LFO2* modulates *Osc2 Wave*) and the *LFO 2 rate*.

Tutorial Setting: FM Megafat

Topic: Distorted FM in monophonic Unison

This sound is hard core, and is well-suited for distorted basses and guitar-like sounds. In its treble range, this sound gets rather “rude.” This cannot be compensated for by scaling, but not every sound has to be “nice” over the entire keyboard range!

- Check out extreme detunings by adjusting the *Analog* parameter.
- Check out the *Flanger* with this sound.
- Engage the filter envelope by lowering the *Cutoff Frequency* of Filter 2 down to 0.
- Add some *Glide* to lead sounds.
- As always, when it comes to FM: You can dramatically alter the sound by varying the frequencies of the oscillators. Make sure you check out the odd intervals, as well.

Tutorial Setting: FM Out of Tune and FM Tuned

Topic: FM with Unusual Spectra

If you’re unconcerned with the pitch of your sound, you can get the weirdest spectra out of odd frequency ratios (oscillator intervals).

This setting offers a bell-like sound, reminiscent of a ring modulator. It was achieved through a setting of *30 s 0 c*, with the modulator set to a value of *0 s 0 c*. Sounds like this were very commonly used in the electronic music of the eighties, and have undergone a resurgence in popularity in modern ambient and trance music styles.

You can further develop the sound by applying filtering, envelope modulations and effects. There is, however, one little problem—the sound is out of tune.

- Use Oscillator 3 as a reference for the tuning of the FM sound, by moving the cursor in the Triangle.
- You’ll notice that the sound is 5 semitones too high (or 7 semitones too low, respectively).
- Transpose both oscillators 1 and 2 five semitones (500 ct) lower. Transposing them upwards is not practical, as you’d need to select *37 s 0 c* for Oscillator 1, which maxes out at *36 s 0 c*.
- It’s important to maintain the frequency ratio (interval) between Oscillators 1 and 2. This means that Oscillator 1 will sound at *25 s 0 c* and Oscillator 2 at *–5 s 0 c*.

Tutorial Settings: PWM Start, PWM Slow, PWM Fast, and PWM Scaled

Topic: Slow and Fast Pulse Width Modulations with Oscillator 2

Pulse Width Modulation (PWM) is one of the most essential features of any sophisticated analog synthesizer. Use this setting to manually control the pulse width of a rectangular wave, set via the *Wave* control

Note: Avoid Drive and Distortion with PWM sounds.

- Select the “PWM Start” setting, and move the *Wave* control slowly back and forth between the rectangular and the pulse wave symbols. Both are green. What you will hear is a (manual) pulse width modulation.
- Select the “PWM Slow” setting. Here, LFO 1 controls the pulse width modulation source, not your manual movements. The result should be quite similar.
- Raise the LFO 1 rate from its pre-set value of 0.230 to 4.400. The result is a classic, fast PWM.
- In this, and the next step, the PWM shall be set so that it sounds slower in the lower keyboard range, and faster in the upper range. This is desirable for many sounds, such as synthetic strings. First, reduce the *LFO 1 Rate* to 3,800.
- Change the modulation intensity of the second router channel (*target* = *LFO1 Rate*, *Source* = *Kybd*) to 0.46. This will alter the scaling of the PWM, making it sound faster in the treble range. You can also hear this type of effect in the “PWM Scaled” setting.

Tutorial Settings: PWM 2 Osc and PWM Soft Strings

Topics: Pulse Width Modulation with Two Oscillators, PWM Strings

In order to make the sound fatter, add Oscillator 3, which can also be modulated in pulse width. In fact, even the first oscillator can deliver PWM. In the “PWM 2 Osc” setting, both oscillators are detuned in a relatively strong way. Develop your own personalized PWM string sound, using this setting as your “base”.

- Adjust the *Chorus intensity*. You’ll probably choose higher values which make the sound rather broad.
- Program Envelope 3 according to your taste. You should, at the very least, raise the *attack* and *release* times. Define it to react to *velocity*, if you prefer. If you do not intend to solely use the sound as a simple pad, a shorter *Decay Time* and a lower *Sustain Level* of about 80 to 90% may be more appropriate.
- Reduce the *Cutoff Frequency* and *Resonance* of Filter 1 to make the sound softer.
- Save the new setting.
- Compare the result with the “PWM 2 Osc” setting. You’ll hear that the sound has undergone a remarkable evolution.
- Compare it to “PWM Soft Strings”, which was created as described above. You’ll probably notice a few similarities.

Tutorial Setting: Ringmod Start

Topic: Ring Modulation

A ring modulator takes its two input signals and outputs the sum and difference frequencies of them.

In the ES2, Oscillator 2 outputs a ring modulator, which is fed with a square wave of Oscillator 2 and the wave of Oscillator 1, when *Ring* is set as Oscillator 2's waveform.

Odd intervals (frequency ratios) between the oscillators, in particular, result in bell-like spectra, much like those heard in the "RingMod Start" setting.

As discussed in the "FM Out of Tune" section, on page 288, the third oscillator can be used as a tuning reference, in order to maintain a kind of basic tuning. On occasion, you may find that it is nice to leave the sound out of tune, for use as a source of overtones and harmonics for another basic wave, supplied by Oscillator 3.

Try to program an atmospheric bell sound, using your own imagination. Some hints:

- Experiment with the various frequency ratios of Oscillators 1 and 2. You may want to use the *29 s 0 c / 21 s 0 c* ratio, which doesn't sound out of tune at all. Ring modulation is not only useful for bell-like sounds, It's also good for a great variety of spectra which tend to sound pretty "weird" at lower frequency settings. Also try alterations to the fine tuning of the Oscillators.
- Check out an *Intensity* of 50% and a *Rate*, set to around 2/3 of the maximum value, for the Chorus effect.
- Set the *Attack* and *Release Times* of Envelope 3 to taste.
- Check out *Drive* and *Filter FM*, if you like your sounds a little "out of control".
- The rest is up to you!

Tutorial Setting: Sync Start

Topic: Oscillator Synchronization

If you select the synced square and sawtooth waveforms for Oscillators 2 and 3, they will be synchronized with Oscillator 1. In the "Sync Start" setting, only Oscillator 2 is audible and Oscillator 3 is switched off.

"Typical" sync sounds feature dynamic frequency sweeps over wide frequency ranges. These frequency modulations (the "sweeps") can be applied in various ways.

- Try the pre-programmed pitch modulation, assigned to the modulation wheel, first.
- In the second router channel, an envelope pitch modulation has been pre-programmed (*target = Pitch 2, Source = Env 1*). Setting the minimum value to 1.0 results in a typical sync envelope. Also check out shorter *Decay Times* for Envelope 1.
- In order to avoid a "sterile", and lifeless sound after the decay phase of the envelope, you may want to modulate the oscillator frequency with an LFO as well. Use the third router channel and set the minimum modulation applied by *LFO 1* to about 0.50.
- Check out the synchronized square wave in place of the synced sawtooth wave.

Note: Pulse width modulation is also available via the synchronized square wave of Oscillators 2 and 3. A modulation of the wave parameters of these two Oscillators results in a PWM if the synced square is selected.

Tutorial Setting: Vector Start and Vector Envelope

Topic: First Steps in Vector Synthesis

In this tutorial section you'll find some useful hints for the programming of vector envelopes. In the "Vector Start" setting, the mix of the Oscillators is controlled by the vector envelope. Each Oscillator has been set to a different waveform.

- Switch from the *Router* view to *Vector* view.
- In its basic (default) setting, the vector envelope has 3 envelope points. Point 1 is the start point, Point 2 the sustain point, and Point 3 is the target in the release phase. By clicking the points, you can see that the mix is always set to 100% for Oscillator 1, in the Triangle.
- Click Point 2, and move the Triangle cursor to Oscillator 2. You'll hear a square wave, instead of Oscillator 1's sawtooth.
- Engage the Vector Envelope by switching the *Solo Point* parameter *off*. As long as it is switched *on*, you will only hear the selected point, with no dynamic modulation. Having switched *Solo Point off*, you'll hear the sound moving from saw to square, with every triggered note.
- Alter the pre-set time of 498 ms, between points 1 and 2.
- While holding Shift, click between points 1 and 2. This will create a new Point 2, and the point formerly known as "Point 2" will become Point 3. The total time span between Point 1 and Point 3 is divided into the times between Points 1 and 2, and 2 and 3. The division takes place at the click location. If you clicked at the exact mid-point, the new time spans are equal.
- Grab the newly created Point 2, and move its cursor in the Triangle to Oscillator 2.
- Grab Point 3, and move its cursor in the Triangle to Oscillator 3. Listen to the three oscillators morphing from sawtooth to square to a triangular wave at the final sustain point.
- Grab Point 4 (the end point) and move its cursor in the Triangle to Oscillator 1 (if it is not already there). Listen to how the sound returns to Oscillator 1's sawtooth wave, following the release of the key.

Tutorial Settings: Vector Envelope and Vector XY

Topic: Vector Synthesis—XY Pad

This example starts where the first one left off. You have a simple Vector Envelope consisting of 4 points, which is set to modulate the oscillator mix (the Triangle).

In this example, the Vector Envelope will be used to control two additional parameters: The *Cutoff Frequency* of Filter 2 and *Panorama*. These are pre-set as the "X" and "Y" *Targets* in the Square. Both have a value of 0.50.

- Switch on *Solo Point*, in order to more easily listen to the settings for the single points.
- Click Point 1. You will only hear Oscillator 1's sawtooth.
- Move the cursor in the Square to the hard left, which results in a low *Cutoff Frequency* for Oscillator 2.
- Click Point 2. You will only hear Oscillator 2's rectangular wave.
- Move the cursor in the Square all the way down, which results in the right-most *Panorama* position.
- Click Point 3. You will only hear Oscillator 3's triangular wave.
- Move the cursor in the Square all the way up, which results in the left-most *Panorama* position.
- Switch on *Solo Point*. The sound begins with a strongly filtered sawtooth wave and turns into an-unfiltered square wave. It initially sounds from the right, and then moves to the left while morphing into a triangular wave. After releasing the key, the saw sound will be heard.

This example sound isn't very dramatic or interesting, but we wanted it to be as clear as possible for the purposes of the tutorial.

Tutorial Settings: Vector Loop

Topic: Vector Synthesis Loops

This example is much more spectacular. The basic sound, without the Vector Envelope, consists of three elements:

- Oscillator 1 delivers a metallic FM spectrum, modulated by Oscillator 2's wavetable.
- Oscillator 2 outputs cross-faded DigiWaves (a wavetable), modulated by LFO 2.
- Oscillator 3 plays a PWM sound at the well-balanced, and keyboard-scaled, speed of LFO 1.

Unison and *Analog* make the sound fat and wide.

These heterogenic sound colors shall be used as sound sources for the vector loop.

A slow *forward* loop is pre-set. It moves from Oscillator 3 (PWM sound, Point 1) to Oscillator 1 (FM sound, Point 2), then to Oscillator 3 again (PWM, Point 3), then to Oscillator 2 (Wavetable, Point 4) and finally, it returns to Oscillator 3 (PWM, Point 5). Points 1 and 5 are identical, avoiding any transition from Point 5 to Point 1 in the *forward* loop. This "transition" could be smoothed out with *Loop Smooth*, but this would make the rhythmic design more difficult to program.

The distances between the points of the Vector Envelope have been set to be rhythmically exact. Given that *Loop Rate* has been engaged, the time values are not displayed in ms, but as percentages. There are four time values (each at 25%), which is a good basis for the transformation into note values.

- Switch off the Vector Envelope by setting *Solo Point* to *on*. This allows you to audition the individual points in isolation.
- Take the opportunity to alter the cursor positions in the Square according to your taste. As in the example above, the *X/Y* axes of the Square control the *Cutoff Frequency* of Filter 2, and the *Panorama* position. Adjustments to these make the sound more vivid.
- Activate the Vector Envelope by setting *Solo Point* to *off*. Check the result, and fine-tune the cursor positions in the Square.
- Alter the *Loop Rate* from the pre-set value of 0.09 up to 2.00. You will hear a periodic modulation, much like that of an LFO. At this point, the modulation is not synchronized with the song tempo. To synchronize the loop speed with the song tempo, move the *Rate* cursor to the very left, and set a note or bar value.
- You can create faster rhythmic note values by clicking between two points and setting the new time values (resulting from the division which occurs) to, say, 12.5%.

Tutorial Setting: Vector Kick

Topic: Bass Drum with Self-Oscillating Filter and Vector Envelope

Electronic kick drum sounds are quite commonly created with modulated self-oscillating filters. This approach can also be taken with the ES2, particularly when the Vector Envelope is used for filter modulation. An advantage of the Vector Envelope, in comparison with conventional ADSR envelopes, is its ability to define/provide two independent decay phases. The distortion effect applies the right amount of “drive” without losing the original sonic character of the drum sound.

In order to make this setting really “punchy”, you must make sure to activate *Flt Reset*. Because all Oscillators are switched off in this setting, the filter needs some time to start oscillating. At the start of each note, *Flt Reset* sends a very short impulse into the filter to make it oscillating right from the start.

Through “tweaks” to the “Vector Kick” setting, you’ll probably be able to create any dancefloor kick drum sound your heart desires. These are the parameters which allow for the most efficient and significant variations:

- Filter2 slopes 12 dB, 18 dB, 24 dB,
- Distortion Intensity, Soft/Hard,
- Envelope 3’s Decay Time (D),
- Vector Envelope Time 1 > 2 (Pre-set: 9.0 ms),
- Vector Envelope Time 2 > 3 (Pre-set: 303 ms),
- Vector Time Scaling.

Tutorial Settings: Vector Perc Synth and Vector Punch Bass

Topic: Percussive Synthesizers and Basses with Two Filter Decay Phases (Vector Envelope)

As with the “Vector Kick,” this setting uses the Vector Envelope to control the Filter Cutoff Frequency (with two independently adjustable decay phases). This would not be possible with a conventional ADSR envelope generator. Try creating further percussive synthesizers and basses by varying these parameters:

- *Vector Envelope Time 1* > 2 (= Decay 1),
- *Vector Envelope Time 2* > 3 (= Decay 2),
- *Vector Time Scaling*,
- Cursor positions in the Square for points 1, 2, and 3 (= *Cutoff Frequency*),
- Choosing other waveforms.

Templates for Logic’s ES2

Welcome to a brief programming tour of the ES2!

While working on the factory preset programming for the ES2, a number of beta testers, sound programmers and other people involved in the project indicated that it would be nice to start their programming work from templates, rather than entirely from scratch.

Needless to say, creating templates which covered all sound genres is something of a mission impossible. As you spend time familiarizing yourself with the ES2’s architecture, you’ll start to understand why ...

Nevertheless, we kept this basic goal in mind, and included this programming tour for the ES2 as a part of the “toolbox” to help you learn and understand the ES2’s architecture through experimentation. You’ll find that this approach is fun. You’ll also discover, as you’re working through a number of simple operations, that results will come quickly when starting to create your personal sound library.

As you become more familiar with the ES2, and what its myriad of functions and parameters do, you can create your own templates for use as starting points when designing new sounds.

-1- Clean Stratocaster (Slap Strat)

The target of this preset was the sound of a Stratocaster, with the switch between bridge and middle pickup in the middle position (in phase). We tried to emulate the noisy “twang,” typical of this sound’s characteristics.

This might be a useful template to start work on emulations of fretted instruments, harpsichords, clavinetts, and so on.

Let's have a look at its architecture:

Osc 1 and 3 provide the basic wave combination within the *DigiWave* field. Changing the *DigiWaves* of both (in combination) delivers a huge number of basic variations—some also work pretty well for electric piano-type keyboard sounds.

Osc 2 adds harmonics with its synced waveform, so you should only vary its pitch or sync waveform. We kept the “noisiness” we were after in mind. There are a couple of values which can be changed here, which will give you a much stronger, more balanced signal.

An old trick which delivers a punchy attack that the use of a “naked” wave wouldn't deliver was used—even with the best and fastest filters available: You use an envelope (in this case No. 1) for a quick “push” of a wavetable's window (or all wavetables together, where it makes sense).

So set up Envelope 1's *Decay* time for this short “push”, moving the wave selectors for all Oscillators on the attack. (... actually it makes no sense on the synced sawtooth Oscillator, No. 2, but what an effort to exclude it from the selection!—it just works this way ...)

So you can vary the “punchiness” of the content between:

- envelope 1's contribution to overall attack noise, changing decay speed (a slow one gives you a peak, a long one gives you a growl, as it is reading a couple of waves from the wavetable).
- modulation destination: you can always assign this to each of the Oscillators separately.
- start point (you vary the wave window start with minimum/maximum control of EG1/Osc.waves modulation: negative values for a startwave before the selected wave, positive starts from a position behind the selected wave and rolls the table back) ...
- Feel free to try out a couple of experiments with this wavetable-driving trick. The growl effect works pretty well for brass sounds, and some organs absolutely shine with a little “click”, courtesy of a wavetable “push”.

Envelope 2, which controls the filter, provides a slight attack used for the slapped characteristics. Setting it to the fastest value eliminates the wah-like attack (and don't worry, there's enough “punch” left).

For playing purposes, you'll find that LFO 2 is used as a real-time source for vibrato. It is assigned to the mod wheel and pressure.

Don't concern yourself too much with the different settings for wheel and pressure—this is only how we felt most comfortable bringing in the controls. Feel free to change them!

Velocity is set up to be very responsive because many synthesizer players don't touch keys in the manner of a piano player's weighted-action-punch. As such, we ask that you play this patch softly, or you may find that the "slap" tends to sweep a little. Alternately, you can adjust the sensitivity of the filter modulation's velocity value to match your personal touch.

Also feel free to increase the *Voices* to maximum; we thought six strings would be enough for a guitar, but for held or sustained notes, a few extra voices may come in handy.

-2- The big twirl, basically (Wheelrock)

This quite ordinary organ patch doesn't hold any deep, high-end sound design secrets: it is just a combination of three Oscillators with their wave levels mixed together. We are positive that you'll easily find a different combination which more closely matches your vision of what an organ sound is like. Check out the *DigiWaves*.

Focus your attention on the mod wheel's response: please hold a chord, and bring the wheel in by moving it slowly upwards, until you reach the top (maximum).

What we intended to program with this mod wheel modulation was a simulation of an accelerating rotor speaker (or "Leslie").

The modulation routings do the following jobs:

- Modulation 1 (Cutoff 1) assigns envelope 2 to Filter 1 (the only one used for this patch), and produces a little organ key "click" with the envelope. We also opened the filter (with Keyboard as via) when playing the higher ranges of the keyboard (with the maximum value).
- Modulation 2 and 3 (Pitch 2/Pitch 3) bring in LFO 1 vibrato, and both Oscillators are modulated out of phase.
- Modulation 5 reduces the overall volume—according to personal taste, the organ's level shouldn't increase too drastically when all modulations are moved to their respective maximums.
- Modulations 6 and 7 (Pitch 2/Pitch 3) detune Oscillators 2 and 3 against each other, within symmetrical values (to avoid the sound getting out of tune, overall). Again, both work out of phase with modulations 2 and 3; Oscillator 1 remains at a stable pitch.
- Modulation 8 brings in LFO1 as a modulator for panorama movement—this patch changes from mono to stereo. If you would prefer a full stereo sound, with a slowly rotating Leslie in its idle position, just set an amount equal to the desired minimum value, thereby achieving a permanent, slow rotation. Another modification you may wish to try is a higher value, resulting in more extreme channel separation.
- Modulation 9 speeds up LFO 2's modulation frequency.
- Modulation 10: for increasing the intensity of the "big twirl" we added a little Cutoff to Filter 1.

Please feel free to find your own values for customization of this setup. While doing so, keep in mind the fact that there are two modulation “couples,” which should only be changed symmetrically (Mod. 2 and 3 work as a pair of “twins,” and also Mod. 6 and 7). So, if you change Pitch 2’s maximum to a lower minus value, remember to set Pitch 3’s maximum value to the same positive amount (same goes for modulation pair 6 and 7).

You can also bring in LFO 2 to increase the pitch diffusion against LFO 1’s pitch and pan movements. Just exchange it for LFO 1 on modulation 2 and 3—but note that there will be no modulation source for the Leslie acceleration—so you’ll need to use it in a static way, just fading it in. Alternately, you’ll need to sacrifice one of the other modulations in favor of a second swirl.

For another stereo modification of the idle sound, you can use the patch in *Unison* mode with a slight detune (please adjust the “analog” parameter for this).

-3- Something Horny (Crescendo Brass)

First of all—the tasks of the Oscillators:

- Oscillator 1 provides the basic brass wave—“sawtooth.”
- Oscillator 2 provides a—not that brassy—“pulse” wave, which brings in the “ensemble.” It is pulse-width modulated by LFO 1 (Modulation 4).

Please note that for any modifications, the following *critical* point should be taken into account. There are four (4) parameters, which behave in an entirely different fashion, once one of them is changed. As such, all four must be changed when making adjustments:

- You may adjust the initial pulse width of Oscillator 2’s wave parameter. We selected a sort of “fat” position, close to the ideal square wave because we wanted to program a full, voluminous synth-brass sound.
- Modulation 4 adjusts the modulation intensity, which means: how far does the range differ from “fat” to “narrow,” when being pulse width modulated? Set with the *Minimum* parameter.
- The rate of LFO1 directly controls the speed of the movement of the pulse width modulation. For this patch, both LFO’s are used, to achieve a stronger diffusion effect at different modulation speeds. As a bit of general advice, we suggest that you use LFO1 for all permanent, automatic modulations because you are able to delay its “job” with its EG parameter. You may use LFO 2 for all real-time modulations, which you intend to access via modwheel, pressure or other controls while playing.
- Additionally, we set up a keyboard assignment as the Modulation 4 source because all pitch or pulse-width modulations tend to cause a stronger detuning in the lower ranges, while the middle and upper key zones feature the desired diffusion effect. When using this parameter, you should initially adjust the lower ranges until an acceptable amount of detuning resulting from the modulation is reached. Once set, check whether or not the modulations in the upper zones work to your satisfaction. Adjust the relationship between intensity (Max) and scaling (Min) values.

Oscillator 3 generates a *DigiWave*, which we considered “brassy” enough within the overall wave mix. As an alternative to the *DigiWave*, we could have used another modulated pulse wave to support the ensemble, or another sawtooth wave to achieve more “fatness” when detuning it with Oscillator 1’s sawtooth wave.

What we were after, however, was to have a little bit of growl, achieved through a short wavetable “push”, as described for the Stratocaster patch, on page 294. This configuration is set up in Modulation 3 (Oscillator 3 Wave moved by Envelope 1’s Decay).

Other controls

Envelope 1 also effects the pitch of Oscillator 2 against Oscillator 3. This results in the clash of both pitches against each other and also against the stable pitch of Oscillator 1, in the attack phase of the sound.

The filter envelope’s design closes with a short stab in the attack phase and then opens again for a slower crescendo phase.

We’ve assigned another real-time crescendo to the mod wheel, which also brings in an overall pitch modulation, controlled by LFO 2.

In addition to all of this, we programmed a sort of “contradictory” real-time modulation (by pressure) which closes the filters. This allows you to play with an additional decrescendo, remotely controlled by touch. Try to get a “feel” for the patch’s response. You’ll find that it offers quite a few controls for “expression”: velocity, pressure after note on, and pressure in advance. Listen to what happens when pressing with the left hand before hitting a new chord with the right hand, and allowing the swell come in.

-4- ... making our job redundant ...? (MW-Pad-Creator 3)

This is an attempt to create a patch which is able to create patches by itself—actually we are still working on the question of how we can get it to save its own results ...

The Basics

Again, Oscillator 2 is used for a pulse width modulation to create a strong ensemble component (please refer to “-3- Something Horny (Crescendo Brass)” section, on page 297, for further information).

Oscillators 1 and 3 are set to an initial start wave combination within their respective *DigiWave* tables. You can modify these and start with a different combination of *DigiWaves* from the outset.

On Modulation 3, we assigned a wavetable “drive” to all three Oscillators via the mod wheel. What this actually means is that you scroll through Oscillator 1 and 3’s wavetables, and change Oscillator 2’s pulse width by moving the mod wheel.

Try a careful, very slow movement of the modwheel, and you'll hear drastic changes within the wave configuration. Each incremental position of the wheel offers a different complete digital pad sound. No frantic movements, please, otherwise this will sound like an AM-radio.

Another potential modification procedure is hidden in the modulation intensity of Oscillator 1, 2, and 3 wave's parameter. As already mentioned with the Stratocaster patch, the value of this intensity parameter assigns step width and direction through the wavetables. You may try modifications to the amount and positive or negative values.

We also discovered an interesting side-effect of FM assignment to Filter 2 (Modulation 4/Lowpass Filter FM). Moving to higher positions of the mod wheel, we increased the frequency modulation on the filter, causing all cyclical "beats" (vibrating pitches, detunes, pulse width) to be emphasized. This also results in a rough and hissing touch to the overall sound character.

FM offers vast scope for experimentation, and you can decide between:

- An initial FM, using Filter 2's FM parameter, which you can "redraw" (set a negative modulation amount for Modulation 4's maximum) by moving the mod wheel to its top position.
- Or you may have permanent FM (and another modulation setup, saved for a different assignment). You can also switch off FM, if you consider its effect too "dirty" sounding.

Real-time control is via pressure for a vibrato (Modulation 10), and also for a slight opening of the Cutoff to emphasize the modulation (Modulation 9).

-5- Another approach to "Crybaby" (Wheelsyncer)

Never obsolete—and undergoing a renaissance in new popular electronic music: Sync Sounds

The technical aspects of forcing an Oscillator to sync are described in "Sync" on page 237. Here's the practical side of the playground.

Wheelsyncer is a single-oscillator lead sound, all others are switched off.

Although Oscillator 2 is the only one actively making any sound, it is directly dependent on Oscillator 1.

If you change Oscillator 1's pitch or tuning, the overall pitch of the sound will go out of tune, or will be transposed.

The pitch of Oscillator 2 provides the tone-color (or the harmonics) for the sync sound. Pitch changes are controlled by modulation 7's setup, where we assigned Oscillator 2's pitch to the mod wheel.

If you move the wheel, you can scroll through the spectrum of harmonics that we've programmed for real-time changes. Any modification here starts with the pitch of Oscillator 2 itself, which we've set to three semitones below the overall pitch. Feel free to start with a different pitch for Oscillator 2; it won't effect the patch's tuning.

The next modification may be modulation 7's intensity (or the interval). We have selected the maximum value—maybe this is too extreme for your needs, so feel free to reduce it.

Another modification lies in the tone color of the lead sound itself. We have switched Oscillator 1 off, as we are already satisfied with the result. If you switch it on, it will offer you the entire range of Oscillator 1's waveforms; from *DigiWaves*, through the standard synth stuff, up to a sine wave, which can be further modulated by FM.

All real-time controls are via the mod wheel: It is used for opening the filter on modulation 6, a panning movement on modulation 8, and acceleration of panning movement on modulation 9. If you have deeper modulation interests, please refer to “-2- The big twirl, basically (Wheelrockers)” section, on page 296, where we used a similar setup for the Leslie simulation.

Ultrabeat is a rhythm synthesizer with integrated step sequencer.

Ultrabeat's synthesis engine is optimized for creating electronic and acoustic drum and percussion sounds.

The sonic diversity possible with Ultrabeat is due to its various synthesis engines. In addition to a new type of Phase Oscillator, sample playback, FM, and physical modeling are also put to use.

Special attention has been paid to achieving the greatest possible range of dynamics for its sounds. Ultrabeat features many extremely versatile functions that can vary the timbre of the sound, dependent on the dynamics of the performance or other modulation sources.

Ultrabeat is integrated into Logic as a software instrument. Its synthesis engine can be directly controlled from within Logic. Ultrabeat also features an integrated step sequencer, used to create rhythmic grooves comprised of patterns. The sequencer displays "running light" type controls like those of classic drum machines.

In addition to the entry and playback of note information, the sequencer plays an important role in the dynamic shaping of rhythms and sounds produced with Ultrabeat.

The Structure of Ultrabeat

Before taking a closer look at the user interface, here are a few further comments on the structure of Ultrabeat:

Most software synthesizers offer one synthesizer per plug-in instance. Ultrabeat, however, places 25 independent synthesizers at your disposal. These synthesizers—called *drum voices* in Ultrabeat—are optimized for the generation of drum and percussion sounds.

The distribution of drum voices across the MIDI keyboard is simple and easily explained: the first (starting from the bottom) 24 MIDI keys are each assigned a single drum voice. The 25th drum voice is an exception, and can be played chromatically over three octaves.

You can compare Ultrabeat to a drum machine that features 24 drum pads plus a built-in three octave keyboard.

Ultrabeat's 24 drum pads are assigned to the first 24 keys of a standard MIDI keyboard (corresponds to MIDI notes C1-B2). The three octave keyboard for the 25th synthesizer begins (lowest note in the range) at C3.

For the sake of simplicity (and to stay with the drum machine analogy), we'll refer to the independent synthesizers (drum voices) as *sounds* which, combined, form an Ultrabeat *drum kit*.

Overview of Ultrabeat



Ultrabeat's user interface is divided into three functional sections. The physical arrangement of these sections illustrates the signal flow within the plug-in:

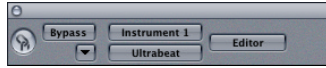
The *assignment* section of the plug-in is found on the left hand side. Here, the individual sounds are selected, mixed, and organized.

The *parameters* of an Ultrabeat sound are displayed to the right of the assignment section. A sound's parameters are displayed once a drum name (in the assignment section) has been clicked with the mouse or is triggered via MIDI.

The *step sequencer* is located at the bottom of the Ultrabeat window. It can be used in place of, or in addition to, MIDI notes entering Ultrabeat's input section (from Logic) to control sounds.

Saving and Loading Settings

In order to save and reload Settings, Ultrabeat uses the procedure common to all Logic plug-ins: as is customary, settings are saved and loaded via the *Settings* menu.



An Ultrabeat setting contains:

- The drum kit, which consists of 25 sounds, inclusive of assignment and mixer settings.
- The complete settings of all parameters for all 25 sounds.
- The sequencer settings and all 24 patterns, including the trigger and velocity/gate rows for all 25 sounds.

Note: The joint recall of all of this data when you load an Ultrabeat setting makes a lot of sense as the musical effect of the patterns, especially those with sequenced gate and velocity parameters, are often intimately connected to the sounds being used.

The Assignment Section

The assignment section of Ultrabeat is found on the left hand side. Individual sounds are selected, mixed, and organized here.

MIDI Control

The first 24 voices of an Ultrabeat instrument are triggered by MIDI notes C1 to B2. Each of these notes triggers a different sound. The first drum voice is assigned to the lowest note (C1) and is displayed at the bottom of the onscreen keyboard. The keys found above this correspond to the ensuing drum voices and MIDI notes, in ascending order. While the first 24 drum voices are each triggered by a single key, the 25th sound is chromatically playable over a span of three octaves (from C3 upwards). The 25th sound is represented by the C3 key on the onscreen keyboard.

The assignment of MIDI notes to drum voices is pre-configured and can't be changed. In order to quickly adapt an Ultrabeat drum kit to play a finished pattern/MIDI Region, the individual drum voices can be exchanged or copied within Ultrabeat, or reassigned by using a Mapped Instrument Object in the Logic Environment.

Selecting the Sounds

Click on the name of the sound in the assignment section to select the relevant drum voice. The sound's parameters will be displayed in the synthesizer section to the right. Click the corresponding note of the onscreen keyboard to play the drum voice.

You can also select the sound by using MIDI note input. To do so, activate the *Voice Select* button in the upper left hand corner of the Plug-in window.

Note: The automatic sound selection function activated by the *Voice Select* button is useful for quick selection of different sounds, which are then displayed for editing.

Note: Given the obvious situation of Ultrabeat receiving numerous trigger notes from Logic or the integrated step sequencer, automatic sound selection would result in constant, visually disturbing changes to the parameter display. To avoid this issue, the *Voice Select* function turns off automatically when a rapid succession of trigger notes occurs.

Naming and Organizing the Sounds

Double-clicking on the name of a drum voice opens its text entry field, allowing you to (re)name it. Press Return or click anywhere outside the text entry field to complete the naming operation.

Control-clicking on the sound name opens a contextual menu that allows the copying and replacement of sounds.

- *Copy*: This command copies the selected sound to the Clipboard.
- *Paste*: This command replaces the selected sound with the sound from the Clipboard.
- *Swap with Clipboard*: This command exchanges and replaces the selected sound with the sound from the Clipboard.

Note: You can also open this menu by right-clicking on the drum sound name.

Note: Naturally, the contextual menu's *Paste* and *Swap with Clipboard* commands require an initial Copy command (to place data in the Clipboard) before functioning.

Copying and replacing drum sounds within an individual drum kit always includes any step sequencer data that may be present. This ensures that copied drum sounds are not separated from their sequencer data, which often contains velocity or gate information for a specific sound.

The Drum Mixer

The assignment section contains a mixer for the 25 sounds found in an Ultrabeat drum kit. It allows you to adjust each sound's volume and pan position, and also offers a Mute and Solo button.



Volume

The individual volumes of all sounds are indicated by blue bars, providing a complete overview of all levels within the kit. You can adjust the volume of the sound, in relation to Ultrabeat's total output level, by dragging the blue bar beneath the sound name. The *Master (Volume)* control is found above the 25th drum voice and controls the total output of the Ultrabeat kit at the main output.

Mute

You can mute individual sounds in a drum kit by pressing the *Mute* button (*M*) to the right of the name.

Solo

You can listen to sounds in isolation by pressing the *Solo* button (*S*), found beside the Mute button.

Pan

The rotary knob to the right of the *Mute* and *Solo* buttons controls the placement of the signal in the stereo field (*Panorama*).

Individual Outputs

Ultrabeat features eight separate stereo outputs, and can be inserted as a multi-channel instrument. In this situation, each drum voice can be independently routed to individual outputs (or output pairs) by using the *Out(put Selection)* pull-down menu, found beside the *Panorama* knob. The following routing assignments are possible: *Main*, 3–4, 5–6, and so on through to 15–16. Drum voices that are routed to an output pair other than *Main* are automatically removed from the main output(s).

Master Volume



The slider above the assignment section controls the levels of all drum voices in the kit. Put another way, the overall mix level of all drum voices.

The Synthesizer

The majority of Ultrabeat's user interface is dedicated to creating and shaping individual drum sounds; in short, Ultrabeat's synthesizer. The parameters of the drum sound selected in the assignment section are displayed in this synthesizer section.

Note: Despite its vast feature set, Ultrabeat's user interface only requires a single Plug-in window. Depending on the function selected, only a few parts of the synthesizer section will change to display different parameters and operational elements, as necessary.

The Signal Flow

Ultrabeat's synthesis engine is based on classic subtractive synthesis principles.

If you look at the synthesizer section from left to right, you'll recognize the classical structure and signal flow of a subtractive synthesizer. First, the basic tonal material is created by the oscillators and noise generator. A filter then takes away certain frequencies from the raw sound, followed by volume shaping (envelopes). The details of Ultrabeat's functions and their importance become more apparent when you look at the three dimensional interface, and recognize the different levels from front to back:

The elevated *Filter* section is in the middle. It's a large, round control object. Its placement and design are symbolic, as the filter section plays a central role in Ultrabeat.

The filter receives its signal from the following sound sources: *Oscillator 1*, *Oscillator 2*, the *Noise Generator* and the *Ring Modulator*. Their output sections are displayed by four objects that sit adjacent to the filter (three round objects and the smaller, rectangular ring modulator to the right of the filter). One level down, you'll find the control elements for these sound sources.

On each of the objects that adjoins the filter you'll find a small, red signal flow button which indicates whether the signals should proceed through the filter or bypass it on their way to the output section of the synthesizer. Along the output path to the right, the signals pass through two equalizers and a stage for stereo expansion or panoramic modulation.

The output of the drum voice is then passed along to the mixer that is integrated into the assignment section (see "The Drum Mixer" section, on page 305).

You can find a detailed description of all synthesis parameters in "The Synthesizer Parameters" section, on page 309.

Modulation

Ultrabeat was developed with special attention paid to dynamic sound shaping possibilities. To this end, almost every sound parameter can be modulated. Ultrabeat provides two powerful LFOs, four new types of envelope generators (*Env 1–4*), velocity, and four freely-definable MIDI controllers as modulation sources.

The settings for the LFOs and envelope generators are represented graphically. They are located above and below Ultrabeat’s output section.

Each modulation routing is set at the desired destination (at the sound parameter itself). Ultrabeat distinguishes between two types of modulation: *Mod* (standard modulation) and *Via* (indirect modulation).

Mod and Via Modulations

You can modulate a sound parameter using an adjustable modulation value (called modulation depth) with *Mod*. You can choose between two LFOs, four envelope generators and the *Max* source, as sources for this modulation.

Via allows you to further tailor the modulation effect. To explain, the modulation depth of the first modulation (*Mod*) can be modulated by a separate, independent source. The intensity of this effect is set with the *Via* parameter. The sources for *Via* modulations include velocity and four freely definable MIDI controllers.

A typical application for *Via* function usage is; increasing a pitch sweep as you play at higher velocities, for example. To this end, an envelope (*Env*) is chosen as the *Mod* source for the *Pitch* of an oscillator, and velocity (*Vel*) is chosen as the *Via* source. The firmer the key is played, the higher (in pitch) it will sound—this is typical of synthesizer tom tom sounds.

Exceptional Modulation Features

The design of the *Mod* and *Via* modulation options afford substantial differences between Ultrabeat and other, more traditional, synthesizer designs. Ultrabeat’s *Mod* and *Via* are given a target value that can be reached by modulation of the respective target parameters, rather than indicating a modulation or effect intensity as a percentage. The result of such modulation routing—its minimum and maximum effect on the modulated parameter—can be set simply, and grasped at a glance, making the complex subject of primary and secondary modulations an intuitive task.

As this method differs greatly from earlier approaches used in synthesis, we recommend that you carefully read the “Modulation” section, on page 325, to fully benefit from this innovative new functionality.

Control Elements

All parameters can be set by clicking or grabbing them, and moving in an upward or downward direction.

Note: If you hold Shift before clicking and moving a control, its value can be fine-tuned.

Repeated clicks on buttons steps through different operating states.

Move the mouse vertically while holding down the mouse button to adjust values in number fields.

You can adjust envelopes graphically by grabbing the envelope handles (or the outer edge of the envelope) with the mouse, and moving it/them.

The equalizer can also be adjusted graphically by dragging on the EQ graph.

The effect of modulations are determined by the small modulation controls found at the control elements of the target parameter. If no source is selected in the *Mod* and *Via* menus (they're set to *Off*), the modulation controls will remain hidden.

Note: All parameters can be reset to their default values by Option-clicking on the respective numerical labels, rotary knobs or sliders.

The Step Sequencer

In the step sequencer section, you can see two rows—each consisting of 32 steps. In the upper row, trigger information (note on) is either inserted or removed, the lower row is used for controlling note lengths and velocity. The rows always correspond to the sound currently selected in the assignment section (synthesizer parameters can be seen in the main working area of the plug-in). Choosing another sound (via the onscreen keyboard or MIDI) updates the sequencer display to show the rows that correspond to the newly-selected sound.

The step sequencer is described more fully in “The Step Sequencer” section, on page 332.

The Synthesizer Parameters

In this section, you'll find a description of the individual parameters found in Ultrabeat's synthesizer section. A discussion of the signal flow can be found in "The Signal Flow" section, on page 306.

The Sound Sources

A drum voice in Ultrabeat has four sound sources: two multi-synthesis capable oscillators, a noise generator and a ring modulator.

Oscillator 1

To use Oscillator 1, you need to first turn it on. This can be done with the *On/Off* button in the upper left corner of the Oscillator 1 section. When in an active state, the button is red.



Note: When you program a drum sound, you can turn the individual sound sources on or off with the corresponding *On/Off* buttons. You can also listen to the individual components of the sound separately this way, and remove them from the patch if necessary.

The volume of Oscillator 1 is controlled by the Volume knob on the right edge of the Oscillator 1 section.



Volume can be modulated by the sources found in the *Mod* and *Via* menus. If a *Mod* source is activated, the effect it has on *Volume* is set by the (Mod) ring that surrounds the knob. If a *Via* source is activated, its effect can be set by moving the slider that appears on the Mod ring. Colored areas between the Volume knob and its surrounding ring clearly show the values of the *Mod* modulation (blue) and the *Via* modulation (green), compared with the mean *Volume* value (red).

If neither a *Mod* or *Via* source is selected (both set to *Off*), the Mod ring, its slider and the colored areas remain hidden.

The curved slider to the left of the *Volume* knob controls the *pitch* of the oscillator in half step intervals. If you press Shift, you can adjust the pitch of Oscillator 1 in cent intervals.



The pitch value is displayed numerically to the left of the slider. You can change the displayed value by click-holding directly on the value field, and moving the mouse vertically.

Pitch can be modulated by the sources found in the *Mod* and *Via* menus. If sources are selected, small blue (*Mod*) and green (*Via*) sliders appear beside the *pitch* control. These allow control over the effect of the *Mod* and *Via* modulation routing. The range affected by parameter modulation is colored blue and green, and runs alongside the *pitch* slider.

Oscillator 1 can be switched between two different types of synthesis engines: *Phase Oscillator* and *FM*. This can be done by clicking the appropriately labeled buttons at the upper edge of the Oscillator 1 section.



Phase Oscillator

The waveform of the Phase Oscillator can be “twisted” with the *Slope*, *Saturation*, and *Asymmetry* parameters, and shaped into almost any basic synthesizer waveform. The effects of these three parameters are graphically illustrated in the waveform display within the oscillator section. Setting all three parameters to zero values will cause the oscillator to produce a sine wave.

The *Slope* parameter determines the slope or steepness of the waveform. The higher the *Slope* value, the steeper the waveform. The resulting sound takes on an increasingly nasal character as steepness is increased.

Increasing *Saturation* values clip the waveform, gradually molding its shape towards a rectangular waveform. This results in a corresponding increase in odd numbered overtones.

Sine and rectangular waves (and all variations in-between) are achieved with the *Slope* and *Saturation* parameters. *Asym* (Asymmetry) “tilts” the waveform in the direction of a sawtooth wave, making the sound more edgy.

The *Asym* parameter can be modulated by the sources found in the *Mod* and *Via* menus. This allows you to create dynamic sound changes at the oscillator level. The effect of the *Mod* and *Via* modulations are adjusted with the small sliders to the left and right of the *Asym* slider. The range affected by the modulations is colored blue (*Mod*) and green (*Via*). If no source is selected in the *Mod* and *Via* menus (set to *Off*), the *Mod* and *Via* sliders remain hidden.

Note: The classic basic waveforms of analog synthesizers can be easily reproduced with the Phase Oscillator: sine, rectangular, and sawtooth waves are each the result of setting the *Slope*, *Saturation*, and *Asym* parameters to their minimum or maximum values, in different combinations.

FM (Frequency Modulation)

In FM mode, Oscillator 1 generates a sine wave. Its frequency is modulated by the waveform of Oscillator 2. Please remember that Oscillator 2 must be switched on to do this. The more complex the Oscillator 2 waveform, the more partials will be created (by increasing the *FM Amount*) during the FM process. You can watch the display to see how the sine wave takes on an increasingly complex shape.

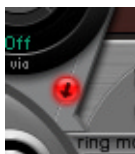
The *FM Amount* parameter can be modulated by the sources found in the *Mod* and *Via* menus. If a *Mod* source is activated, the effect it has on *FM Amount* is altered by moving the ring that surrounds the rotary knob. If a *Via* source is activated, its effect can be set by the moveable slider that appears on the *Mod* ring. Between the rotary knob and its surrounding Mod ring, colored areas clearly show the values of the *Mod* modulation (blue) and the *Via* modulation (green), compared with the mean *FM Amount* value (red).

If neither a *Mod* or *Via* source is selected (both set to *Off*), the Mod ring and slider remain hidden.

Note: While the Phase Oscillator is well-suited for simulating analog waveforms and analog-style sounds, FM mode offers bell-like digital tones and metallic sounds.

Filter Bypass Button

Between Oscillator 1 and the filter section you'll find a signal flow switch that controls the routing (Filter Bypass button). Repeated mouse clicks will send the signal to the filter (Filter Bypass button turns red), or bypass the filter and send it directly to the EQ section (*Filter Bypass button* remains gray).



The direction of the arrow on the Filter Bypass button illustrates the routing.

Note: The Filter Bypass button simply determines the signal flow. It doesn't turn the oscillator on or off. Use the oscillator *On/Off* button for this (see above).

Oscillator 2

To use Oscillator 2, you first need to turn it on. This can be done with the *On/Off* button in the lower left corner of the Oscillator 2 section. When active, the button is red.

Note: When you program a drum sound, you can turn the individual sound sources on or off with the corresponding *On/Off* buttons. You can also listen to the individual components of the sound separately this way, and remove them from the patch if desired (or necessary).

The volume of Oscillator 2 is controlled by the Volume knob on the right edge of the Oscillator 2 section.

Volume can be modulated by the sources found in the *Mod* and *Via* menus. If a *Mod* source is activated, the effect it has on *Volume* is set by the (Mod) ring that surrounds the knob. If a *Via* source is activated, its effect can be set by moving the slider that appears on the Mod ring. Colored areas between the Volume knob and its surrounding ring clearly show the values of the *Mod* modulation (blue) and the *Via* modulation (green), compared with the mean *Volume* value (red).

If neither a *Mod* or *Via* source is selected (both set to *Off*), the Mod ring, its slider and the colored areas remain hidden.

The curved slider to the left of the *Volume* knob controls the *pitch* of the oscillator in half step intervals. If you press Shift, you can adjust the pitch of Oscillator 1 in cent intervals.

The pitch value is displayed numerically to the left of the slider. You can change the displayed value by click-holding directly on the value field, and moving the mouse vertically.

Pitch can be modulated by the sources found in the *Mod* and *Via* menus. If sources are selected, small blue (*Mod*) and green (*Via*) sliders appear beside the *pitch* control. These allow control over the effect of the *Mod* and *Via* modulation routing. The range affected by parameter modulation is colored blue and green, and runs alongside the *pitch* slider.

Oscillator 2 can be switched between three different types of synthesis engines: *Phase Oscillator*, *Sample*, and *Model*. This can be done by clicking the appropriately labeled buttons at the lower edge of the Oscillator 2 section.

Phase Oscillator

The waveform of the Phase Oscillator can be “twisted” with the *Slope*, *Saturation*, and *Asymmetry* parameters, and shaped into almost any basic synthesizer waveform. The effects of these three parameters are graphically illustrated in the waveform display within the oscillator section. Setting all three parameters to zero values will cause the oscillator to produce a sine wave.

The *Slope* parameter determines the slope or steepness of the waveform. The higher the *Slope* value, the steeper the waveform. The resulting sound takes on an increasingly nasal character as steepness is increased.

Increasing *Saturation* values clip the waveform, gradually molding its shape towards a rectangular waveform. This results in a corresponding increase in odd numbered overtones.

Sine and rectangular waves (and all variations in-between) are achieved with the *Slope* and *Saturation* parameters. *Asym* (Asymmetry) “tilts” the waveform in the direction of a sawtooth wave, making the sound more edgy.

The *Saturation* parameter can be modulated by the sources found in the *Mod* and *Via* menus. This allows you to create dynamic sound changes at the oscillator level. The effect of the *Mod* and *Via* modulations are adjusted with the small sliders to the left and right of the *Saturation* slider. The range affected by the modulations is colored blue (*Mod*) and green (*Via*). If no source is selected in the *Mod* and *Via* menus (set to *Off*), the *Mod* and *Via* sliders remain hidden.

Note: Oscillator 1 differs from Oscillator 2 as it is *Saturation* not *Asymmetry* that can be modulated. This difference means that when both oscillators are in Phase Oscillator mode they can produce different types of sounds.

Sample

A selection of multi-layer drum and percussion samples that were specially created for Ultrabeat and its function set are included with Ultrabeat. You can load these via the sample function in Oscillator 2. You can also load your own samples in AIFF, WAV or SDII stereo interleaved format.

To accomplish this, activate the button labeled *Sample* in Oscillator 2. You can see how the control elements of Oscillator 2 change—among other things, a waveform display appears.



Samples are selected in a dialog box, which can be reached by clicking on the arrow (or *no sample loaded* text) in the upper left corner (or top) of the waveform display. In addition to the supplied Ultrabeat multi-layer samples, it is also possible to use this dialog to select and load an audio file of your own choice. It should be noted, however, that the velocity layering function is not available for such samples.

Note: When saving a drum kit using the Settings menu, the location of the sample is saved with the setting. The Ultrabeat setting doesn't actually save the audio files themselves—only a reference to their location. If you load a setting that contains a reference to a sample that has been moved or erased, Ultrabeat will present you with a dialog box that requests you to find it. To avoid this problem, it is highly recommended that you use a dedicated Ultrabeat sample folder.

The *Reverse* arrow changes the playback direction of the sample (forwards/backwards).

The two *Min/Max (Velocity)* horizontal sliders below the waveform display determine the start point of the sample—dependent on the dynamics of the performance. *Min* determines the start point of the sample at the minimum velocity level (velocity = 1), *Max* at the maximum level (velocity = 127). If *Min* and *Max* are set to the same value, this corresponds to a static setting of the sample start point.

Every internal Ultrabeat sample consists of different layers that are velocity switched—dependent on the dynamics of your performance. The layers that are switched to, in accordance with incoming velocity values, is determined by the small *Vel Layer* slider on the right. This slider determines which layer is triggered at the minimal level (velocity = 1). The second small slider on the left determines which layer sounds at the maximum level (velocity = 127).

Note: User-supplied samples loaded into Ultrabeat cannot be separated into velocity-dependent layers, and therefore the *Vel Layer* slider has no effect on samples you have created and imported.

Model

This oscillator type offers a physical model of a string instrument for the creation of percussive sounds. The parameters at your disposal are based on the physical properties of a real string.



Two contrasting exciters, each with different sound characteristics, are available. You can toggle between them with the corresponding buttons (*Type 1* and *Type 2*).

Note: In Ultrabeat's oscillator 2 *Model*, an exciter is the agent or triggering device used to initiate the vibration of the string. It should not be confused with the effect plug-in of the same name.

In the Material Pad, you can set the *Inner Loss* and *Stiffness* string parameters. This parameter actually determines the material qualities of the physical model.

Inner Loss determines the damping of the string which, in the real world, depends on the material the string is made of (for example steel, glass, nylon, or wood). Damping primarily affects high frequencies, and forces the sound to become more muffled and smooth during the decay phase.

Stiffness controls the stiffness or rigidity of the string. In the real world, this depends on the material the strings are made of and their diameter (or, more precisely: their sluggishness). Rigid strings create an inharmonic vibration where the overtones do not represent whole number multiples of the fundamental frequency. These overtones are, in fact, slightly higher. Marked increases in rigidity (stiffness) ultimately transforms the string into a metal rod.

Along the x-axis of the Material Pad you'll find the value range for the *Stiffness* parameter, and the value range for the *Inner Loss* parameter along the y-axis. To adjust the parameters, click-hold on the dot in the Material Pad and move it.

Note: Click on the dot in the Material Pad while holding down Option if you want to return the string parameters to their default values.

To the right of the Material Pad you'll find the *Resolution* parameter. You can also use this parameter to influence the overtone structure of the sound.

In contrast to the other parameters of the Model oscillator, *Resolution* does not reproduce a pre-defined real-world property of the physical model, but affects the modelling process itself: higher values lead to an improved calculation resolution which results in more overtones. Lower values reduce the precision of the calculations, leading to fewer overtones and often to inharmonic spectra.

Filter Bypass Button

Between Oscillator 2 and the filter section you'll find a signal flow switch that controls the routing (Filter Bypass button). Repeated mouse clicks will send the signal to the filter (Filter Bypass button turns red), or bypass the filter and send it directly to the EQ section (*Filter Bypass Switch* remains gray). The direction of the arrow on the Filter Bypass button illustrates the routing.

Note: The Filter Bypass button simply determines the signal flow. It doesn't turn the oscillator on or off. Use the oscillator *On/Off* button for this (see above).

The Ring Modulator

The ring modulator functions as its own sound source; its signal can bypass or be sent into the filter, independent of Oscillators 1 and 2. Its volume can also be regulated. Please note that both oscillators need to be switched on to use it.

The sound of the ring modulator is largely dependent on both of the oscillators, as it modulates the output signals of both. Parameter changes, especially the tuning relationships of each oscillator, have a direct effect on the sound of the ring modulator.

Note: The individual volumes of the oscillators have no effect on the process of ring modulation.

The ring modulator doesn't have an *On/Off* button like the oscillators. It is activated by clicking directly on the *Ring Mod* label itself. When switched on, the label is red and when off, it's gray.



Note: As the ring modulator needs the signals of both oscillators to produce its output, the ring modulator is muted when one of the oscillators is switched off. If you want to hear the ring modulator's signal in isolation (in order to better judge your settings), temporarily set the volume of both oscillators to 0.

The slider adjusts the output volume of the ring modulator. *Volume* can be modulated by the sources found in the *Mod* and *Via* menus. If sources are selected, small blue (*Mod*) and green (*Via*) sliders appear beside the *volume* control. These allow control over the effect of the *Mod* and *Via* modulation routing. The range affected by parameter modulation is colored blue and green, and runs alongside the *volume* slider.

Between the ring modulator and the filter section you'll find a signal flow switch that controls the routing (Filter Bypass button). Repeated mouse clicks will send the signal to the filter (Filter Bypass button turns red), or bypass the filter and send it directly to the EQ section (*Filter Bypass Switch* remains gray). The direction of the arrow on the Filter Bypass button illustrates the routing.

Note: The Filter Bypass button determines the signal flow. It doesn't turn the ring modulator on or off. Use the *Ring Mod* field for this (see above).

The Noise Generator

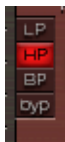
The fourth synth engine is the noise generator. Noise contains—in a technical sense—all tonal frequencies; that's why our ears can't recognize any tonality in a noise signal. Despite this (or as a direct result of it), noise is an indispensable ingredient when creating drum sounds. For this reason, Ultrabeat's noise generator is outfitted with extensive features.

To use the noise generator, you first need to turn it on. This can be done with the *On/Off* button. When in an active state, the button is red.

Volume can be modulated by the sources found in the *Mod* and *Via* menus. If a *Mod* source is activated, the effect it has on *Volume* is set by the (Mod) ring that surrounds the knob. If a *Via* source is activated, its effect can be set by moving the slider that appears on the Mod ring. Colored areas between the Volume knob and its surrounding ring clearly show the values of the *Mod* modulation (blue) and the *Via* modulation (green), compared with the mean *Volume* value (red).

If neither a *Mod* or *Via* source is selected (both set to *Off*), the Mod ring, its slider and the colored areas remain hidden.

The noise generator has its own filter which functions independently of Ultrabeat's multimode filter. The four *Type* buttons *LP*, *HP*, *BP*, and *Byp* allow you to switch the filter between lowpass, highpass or bandpass modes, or deactivate it (*Byp*).



The names of the filter types illustrate how they work: A lowpass (LP) filter allows frequencies that are lower than the Cutoff frequency (see below) to pass. This filter type dampens higher frequencies, and makes the sound less sharp and bright.

A highpass (HP) filter has exactly the opposite effect. It filters out the lower frequencies while leaving the higher frequencies untouched.

The bandpass (BP) filter only allows a certain frequency range (a frequency band) centered around the *Cutoff* frequency to pass. It can be used in the upper, as well as at the lower, end of the frequency spectrum to reduce the highs and lows of a sound.

The *Cut* knob determines the Cutoff frequency, and defines the point in the frequency spectrum where reduction begins. Depending on the type of filter you select, you can make a sound darker (*LP*), thinner (*HP*) or more nasal (*BP*) by adjusting the *Cut* value.

Cutoff can be modulated by the sources found in the *Mod* and *Via* menus. If a *Mod* source is activated, the effect it has on *Cutoff* is set by the (Mod) ring that surrounds the knob. If a *Via* source is activated, its effect can be set by moving the slider that appears on the Mod ring. Colored areas between the Cutoff knob and its surrounding ring clearly show the values of the *Mod* modulation (blue) and the *Via* modulation (green), compared with the mean value (red).

If neither a *Mod* or *Via* source is selected (set to *Off*), the ring and slider remain hidden.

Increasing *Resonance* boosts frequencies that surround the Cutoff frequency. Values range from 0 (no increase) to self-oscillation of the filter at high *Resonance* values.

Note: Self-oscillation is typical of analog filter circuits. It occurs when the filter feeds back into itself and begins to oscillate at its natural frequency, when high resonance values are used.

Dirt is a parameter developed especially for the noise generator. Turning up the *Dirt* knob roughens up the pure, white noise appreciably, making it more grainy.

Note: The *Dirt* parameter is especially effective at high *Resonance* values.

Between the noise generator and the filter section you'll find a signal flow switch that controls the routing (Filter Bypass button). Repeated mouse clicks will send the signal to the filter (Filter Bypass button turns red), or bypass the filter and send it directly to the EQ section (*Filter Bypass Switch* remains gray). The direction of the arrow on the Filter Bypass button illustrates the routing.

Note: The Filter Bypass button determines the signal flow. It doesn't turn the noise generator on or off. Use the *On/Off* button for this (see above).

In the noise generator, the Filter Bypass button operates as per the oscillators and ring modulator: it determines whether the signal is sent to Ultrabeat's main filter or bypasses it. It has no effect on the independent filter contained in the noise generator. This is deactivated with the *Byp* button in the noise generator filter section.

It is therefore possible to filter the noise generator signal twice. In many instances, you may want the noise generator signal to bypass the main filter, freeing the main filter for other duties—an important element when programming drum sounds.

The Filter Section

The output signals of both oscillators, the ring modulator and the noise generator are passed on to Ultrabeat's central filter section (if they haven't bypassed it through use of the various Filter Bypass buttons). The filter section offers a multimode filter and a distortion unit.

The order that sounds are passed through the filter and distortion unit is determined by the red arrow found at the "equator" of the filter section. Repeated clicking on the arrow switches between the ▲ (distortion, then filter) and ▼ (filter, then distortion) settings.



Note: A description of Ultrabeat's filter parameters and a discussion on the basic concepts of subtractive synthesis and analog filters follows. If you're new to synthesizers, please read the "Synthesizer Basics" chapter for more information.

The Multimode Filter

A click on the word *Filter* in the upper middle section activates or deactivates the multimode filter. In a deactivated state (the word *Filter* is gray: red when active), all synthesis engine signals pass the filter unprocessed and are forwarded to the distortion unit.

The multimode filter offers the following filter types: lowpass (*LP*), highpass (*HP*), bandpass (*BP*), and band rejection (*BR*).

You can switch between the filter types by pressing the corresponding button directly beneath the word *Filter*.



The names of the individual filters illustrate their function: A lowpass (*LP*) filter allows frequencies lower than the *Cutoff* frequency to pass. It removes (cuts) the highs of a sound, making it darker and less bright.

A highpass (*HP*) filter allows frequencies higher than the *Cutoff* frequency to pass. The lows of the sound are cut.

A bandpass (*BP*) filter allows a frequency band centered around the *Cutoff* frequency to pass. Frequencies that lie further away (the lows and highs outside the band) are filtered out. A sound with a lot of mid frequency content results.

The abbreviation *BR* stands for Band Rejection filter. In this mode, the area (the frequency “band”, to be more exact) around the Cutoff frequency is filtered out while frequencies that lie further away (from the Cutoff frequency) are allowed to pass. The mid frequencies become softer and the low and high frequencies remain unchanged.

Below the filter type buttons, you’ll find two buttons labeled 12 and 24. These allow you to select the *slope* of a filter. All of Ultrabeat’s filter types offer a filter *slope* of either 12 or 24 dB/octave.

Note: Filters don’t completely remove the parts of the signal that you wish to filter out and always work with limited precision in the selected “band”. The steepness or slope is measured in decibels of damping per octave (dB/oct). Frequencies that are located close to the Cutoff frequency are generally reduced less than those that are farther away. The higher the *slope* value, the more apparent the level difference is between frequencies that are nearer the Cutoff frequency and those that are further away from it.

The *Cut* knob determines the Cutoff frequency of the filter.

Note: Adjusting the Cutoff frequency can make a sound darker (*LP*), thinner (*HP*), more nasal (*BP*) or more transparent (*BR*), dependent on the type of filter chosen.

Cutoff can be modulated by the sources found in the *Mod* and *Via* menus. If a *Mod* source is activated, the effect it has on *Cutoff* is set by the (Mod) ring that surrounds the knob. If a *Via* source is activated, its effect can be set by moving the slider that appears on the Mod ring. Colored areas between the Cutoff knob and its surrounding ring clearly show the values of the *Mod* modulation (blue) and the *Via* modulation (green), compared with the mean value (red).

If neither a *Mod* or *Via* source is selected (set to *Off*), the ring and slider remain hidden.

Increasing *Resonance* boosts frequencies that surround the Cutoff frequency. Values range from 0 (no increase) to self-oscillation of the filter at high *Resonance* values.

Note: Self-oscillation is typical of analog filter circuits. It occurs when the filter feeds back into itself and begins to oscillate at its natural frequency, when high resonance values are used.

If a *Mod* source is activated, the effect it has on *Resonance* is set by the (Mod) ring that surrounds the knob. If a *Via* source is activated, its effect can be set by moving the slider that appears on the Mod ring. Colored areas between the Resonance knob and its surrounding ring clearly show the values of the *Mod* modulation (blue) and the *Via* modulation (green), compared with the mean value (red).

If neither a *Mod* or *Via* source is selected (set to *Off*), the ring and slider remain hidden.

The Distortion Unit

Depending on the order determined by the red arrow in the filter section, the distortion unit is inserted either before or after the multimode filter. It provides either a bit crusher or distortion effect.

The desired mode is activated by clicking on the *Crush* or *Distort* button. The active effect is indicated in red. If neither button is red the distortion unit is bypassed

Note: The distortion effect is modeled on an analog distortion unit, which distorts the sound by overdriving the level. The bit crusher uses a digital process that deliberately reduces the digital resolution of the sound (measured in bits), achieving an intentional digital coloration of the sound. Both methods lead to distortions that are as tonally divergent as the two approaches. Distortion offers a more “analog” feel while the bit crusher can’t hide its digital origins (nor is it supposed to!).

The bit crusher and distortion effect are adjusted with the same three dials:

- *Drive*: Turning this parameter up increases the degree of distortion.
- *Color*: This parameter determines the basic sound of the distortion. Higher values help you achieve a brighter sound and lower values lead to a darker, warmer tone.
- *Level/Clip*: The output volume is set here (*Level*) when in distortion mode. In bit crusher mode, this dial determines the level required before distortion (*Clip*) begins.

Output Section

Depending on the status of each Filter Bypass button, the output signals of both oscillators, the ring modulator and the noise generator are routed either; directly or via the filter section to the output section of Ultrabeat. The output section passes signals through both equalizers (EQ) and the pan/stereo spread section (in a pre-configured order) before the final level is set, and the trigger behavior (of the signals) is adjusted.

2 Band EQ

Both equalizer bands have almost identical features. Their parameters are explained jointly, but you can, of course, adjust band 1 (the upper EQ in the output section) and band 2 separately.

Clicking on the *Band 1* and *Band 2* labels turns the individual band on or off. When active, the field is red. If neither EQ is activated, the signal passes through unaffected.

The EQ Type buttons switch between two different types of EQs: shelving and peak.



In shelving mode, all frequencies above or below the set frequency are either increased or reduced. In peak mode, only frequencies located near the set frequency are affected.

The shelving EQ is activated by pressing the upper of the two EQ Type buttons. The peak EQ is activated by pressing the lower of the two EQ Type buttons.

Note: The shelving filter in band 1 offers a *low shelving* EQ while the shelving filter in band 2 features a *high shelving* EQ. Low shelving means that the frequencies below the set frequency are affected. High shelving affects frequencies above the set frequency.

Note: Shelving EQs function similarly to synthesizer lowpass and highpass filters. The fundamental difference: while lowpass and highpass filters merely dampen certain frequencies (filter them out), shelving EQs also allow these frequencies to be boosted.

The *EQ Gain* knob is bipolar. Positive values (turned to the right) boost a certain frequency range as determined by the *EQ Type* and *Hz* settings. Negative gain values (to the left) lower the gain of the frequency range. If the Gain knob set to the mean value of 0, the EQ has no effect.

Note: You can also return this knob to its neutral position by Option-clicking on it. Alternately, you can click on the tiny 0 above the *EQ Gain* knob.

The frequency (measured in scale units called Hertz) is set by click-dragging vertically on the *Hz* parameter field. This determines the frequency range to be boosted or reduced.

Note: Option-clicking the *Hz* parameter returns its value to a neutral position. This is 200 Hz for the first band and 2000 Hz for the second. The selection of these default frequencies was made in accordance with the different shelving characteristics of each frequency band. Band 1 is designed to filter low frequencies and band 2, high frequencies.

The *Q* factor is regulated by click-dragging vertically on the *Q* parameter field. The effect of *Q* on the sound is heavily dependent on the selected EQ Type:

- With shelving filters, as the *Q* value goes up, the area around the threshold frequency becomes more pronounced.
- With the peak EQ, *Q* determines the width of the frequency band selection: low *Q* values select a broad band while high *Q* values select a very narrow band to be boosted or reduced with the Gain control.

Editing the EQ Bands Graphically

The EQ bands each have their own display which shows changes on a frequency response curve. The display provides immediate access to the *Gain*, *Hz*, and *Q* EQ parameters. Just grab the graphical frequency response curve with the mouse, and alter it by moving the mouse vertically and/or horizontally! Horizontal moves change the EQ frequency, and vertical moves influence the *Gain*. At the peak (maximum point) of the EQ, a handle can be dragged vertically to change the *Q* factor. This procedure is much like that used in Logic's Channel EQ.

Pan Modulation/Stereo Spread

The EQ's output signal is passed along to the pan/spread section. In the pan/spread section, the placement of the sound in the stereo field (set in the assignment section's mixer) can be modulated (Pan Modulation mode), or the stereo basis of the sound can be broadened (Stereo Spread mode). Activate the desired mode by clicking on the appropriate button (*Pan Mod* or *Spread*). If neither mode is activated, the signal passes through unaffected.

Pan Modulation

Pan Modulation varies the panorama position of a drum sound dependent on a *Mod* and *Via* source. The modulation set here is relative to the panorama position set in Ultrabeat's mixer.

The panorama position set in the mixer is represented here by a thin, red line. To the left and right of the line, small sliders (and corresponding menus) allow the adjustment of the *Mod* and *Via* modulation routings.

Note: You cannot directly grab and move the red line (that represents panorama position) shown in this section. In order to move the line, rotate the pan knob in the mixer section (see the GUI detail below).



Stereo Spread

Stereo Spread broadens the stereo image, making it wider and more spacious.

Low Frequency applies the (spreading) effect to the bass frequencies: the higher the value, the more prominent the effect becomes. *Hi Frequency* allows you to apply the effect to the high frequencies.

Voice Volume

This rotary knob adjusts the output volume of the individual drum sounds. To be more exact, you are controlling the *Voice Volume* with *Env 4*, thereby adjusting the maximum volume level attained after the attack phase of *Env 4*.

The effect the envelope has on *Voice Volume* can also be modulated by a *Via* source.

Note: The leveling stage for *Voice Volume* precedes the sliders in the mixer. This approach allows the starting volume of the individual drum voices to be set independently of their relative levels in the drum kit mix.

Trigger and Group Menus

The manner in which Ultrabeat reacts to a succession of incoming notes is individually defined for each sound. These parameters are found in the output section, below the *Voice Volume* knob.



Clicking the button below the Trigger label opens the Trigger menu, allowing you to choose between *Single* and *Multi* trigger modes.

- **Single:** A new trigger note cuts off the (same) note that is currently playing.
- **Multi:** When a new note is played, preceding (currently playing) notes continue to decay in accordance with their respective amplitude envelope settings (*Env 4*).

Clicking the button below the Group label opens the Group menu, allowing a choice between the *Off* and group 1 to 8 settings. If two different sounds are assigned to the same group, they will cut each other off. A typical use of this facility is when you're programming hi-hat sounds: when playing a real hi-hat, the closed hi-hat note cuts off and mutes the ringing of the open hi-hat. This function is often referred to as "hi-hat group" mode.

Note: While in Single Trigger mode, only the currently sounding note of the same sound is cut off. A sound that is assigned to a group cuts off all other sounds (regardless of note) in the group.

Clicking on the *Gate* button turns the *Gate* function on and off. If active, the sound is immediately cut off as soon as the MIDI note is released (MIDI note off), regardless of envelope settings.

Note: The *Gate* function ensures that a specific sound does not play beyond a note off event, as defined in the sequencer. A corresponding rhythmic definition of the exact note off time is achieved with the *Gate Length* parameter in Ultrabeat's step sequencer. Logic's sequencer allows you to quantize note off events, or precisely edit them manually. Note length can be an important creative element when programming rhythm tracks.

Modulation

Numerous sound parameters can be controlled dynamically (modulated) in Ultrabeat. The setting of modulation routings follows a universal principle that is explained in this chapter.

The Principle of the Modulation Routings

Ultrabeat's modulation routings feature three key players:

- The synthesizer parameter that you want to modulate (the *modulation target*)
- The source of the modulation (the *modulation source*)
- A second modulation source that can influence the intensity of the first modulation (we call this a *Via* modulation)

Let's look at an example to better understand how this works:

The *Cut* (*Cutoff*) parameter has a mean (default) value of 0.50. It's not being modulated yet as no modulation source has been selected in either the red *Mod* or blue *Via* menu (set to *Off*).



As soon as a modulation source is selected in the *Mod* menu (*Env 1* in this example), the ring around the rotary knob is activated. Grabbing and moving this ring with the mouse allows you to set the value that this parameter will be increased to by the *Mod* source (0.70 in the example).



As soon as a modulation source is selected in the *Via* menu (*Vel* in this example), a movable slider appears on the *Mod* ring. Grabbing and moving this slider with the mouse allows you to set the maximum modulation value that can be reached through use of the *Via source* (0.90 in this example).



So much for the settings. What do the markings found around the *Cut* knob mean, and what is happening to the sound?

The *Mod* and *Via* controls indicate the minimum and maximum values that the modulated parameter can attain (in comparison to the mean value). Put another way, the *result* of the modulation. These controls don't show us—as is customary with other synthesizers—a percentage value describing their intensity, but state very clearly: “These are the minimum and maximum possible values of the modulated parameter”.

Back to our example; the frequency of the filter is set to the mean value of 0.50. When the *Mod* source *Env 1* enters the equation, the *Env 1* envelope generator drives the *Cut* value up from 0.50 to 0.70 (during the attack phase) and back down to 0.50 (during the decay phase).

Note: You can view the exact values in the Help Tags that appear when you grab the individual handles of various parameters.

If the *Via* source *Ctrl A* is introduced, the following interplay occurs: when *Ctrl A* remains at its minimum value, nothing changes (yet); *Cutoff* continues to be modulated between values of 0.50 and 0.70 by the envelope. A maximum value for *Ctrl A* causes the envelope generator to vary the parameter between the values of 0.50 (the mean value) and 0.90 (the *Via* amount).

You can see, at a glance, the degree of maximum influence on basic parameters by the *Mod* and *Via* modulation sources: the area between the *Mod* and *Via* points shows the amount that the modulation depth can be (further) altered by the *Via* modulation source. In our example, the *Cutoff* can reach values between 0.70 and 0.90 depending on the value being sent by *Ctrl A*.

Here's another example:



Cutoff is again set to 0.50, *Env 1* now drives the value down to 0.25, and a maximum *Ctrl/A* value reduces the *Cutoff* frequency down to 0.

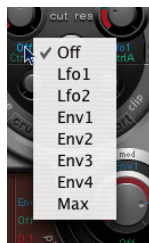
Here is another example that illustrates the simplicity and speed of Ultrabeat's modulation options:



In this example, you won't just be changing the modulation intensity of *Env 1* (which affects *Cutoff*) with the dynamics of your performance (*Vel*), but you'll also control its direction as well. Try this setting in Ultrabeat to create some extremely interesting sounds.

Setting the Modulation Routing

Clicking on the *Mod* label opens the *Mod* menu. This is where you can choose one of the *LFOs* or envelope generators (*Env*) as a modulation source.

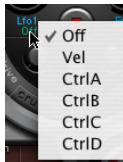


The *Off* setting deactivates the *Mod* routing, and the *Mod* ring can no longer be adjusted. In this situation, no *Via* modulation can occur either (this is because *Via* no longer has a modulation target) and the *Via* slider disappears.

Note: The *Max* setting produces a static modulation at maximum level. When the *Mod* value is set to *Max*, the *Via* parameter is routed directly to the modulation target. This way, velocity can be used as a direct modulation source, even though *Vel* is not available as a source in the *Mod* menu.

Another example would be to set up an external MIDI fader unit with *Ctrl A*, *B*, *C*, or *D* (see below). You could then use the *Max* item in the *Mod* menu to route the respective *Via source*—*Ctrl A*, *B*, *C*, or *D*—to the parameter you'd like to control with one of the faders on your MIDI fader box.

Clicking on the *Via* label opens the *Via* menu. This is where you choose the *Vel* or *Ctrl A* to *Ctrl D* parameters.



Vel represents velocity.

Ctrl A to *Ctrl D* are four continuous controllers that can be assigned to four external MIDI controllers. These assignments are made in the *MIDI Controller Assignments* area at the upper right edge of the Plug-in window (see below). The assignments apply to all sounds in the current Ultrabeat plug-in instance.



MIDI Controllers A–D

In the *MIDI Controller Assignments* area at the upper edge of the Plug-in window you can assign a standard MIDI controller to each of the four controller slots: *Ctrl A*, *B*, *C*, or *D*. *Ctrl A*, *B*, *C*, and *D* can be used as *Via* modulation sources within Ultrabeat. Use these assignments to set up your external MIDI controller hardware to operate with Ultrabeat. As examples; aftertouch or the modulation wheel of your MIDI keyboard.

Note: All MIDI Controller Assignment menus feature a *Learn* option. If this is selected, the parameter will automatically be assigned to the first appropriate incoming MIDI data message.

LFO 1/2

Among other items, two LFOs are available as modulation sources in the *Mod* menu.

LFO is the abbreviation for Low Frequency Oscillator. The LFO signal is used as a modulation source. In an analog synthesizer, LFO frequency generally ranges between 0.1 and 20 Hz, which is outside our audible frequency spectrum.

Note: The speed of the LFO in Ultrabeat can reach up to 100 Hz which, when compared to analog synthesizers, offers a number of far-reaching possibilities.

Ultrabeat has two LFOs that offer identical feature sets. The parameters for each are described jointly; you can, of course, adjust LFO 1 and LFO 2 completely independently of each other.

The buttons labeled 1 and 2 select the corresponding LFO, allowing adjustments of each LFO's parameters. The blue *On/Off* button activates and deactivates the selected LFO.



The LFO section display shows the LFO waveform, the shape of which is governed by the *Shape* slider located underneath it. Dragging the slider from left to right causes the waveform to fluidly morph from a sine to a triangle, and then finally to a square wave (with variable pulse width), including all variations in-between. At the far right hand position of the *Shape* slider, the LFO produces random waveforms.

The LFO speed (*Rate*) can be set independently (*Free*) or synchronised (*Sync*) to Logic's song tempo. Clicking either button once activates the corresponding mode.

The *Rate* knob determines the speed of the LFO. Depending on the *Free/Sync* setting, *Rate* is displayed in either Hz or musical (measure) units.

The *Ramp* knob determines whether the output signal of the LFO is faded in or out. *Ramp* works in a bipolar fashion: turning it to the right increases the *Attack* time of the LFO, turning it to the left decreases the *Decay* time. In its middle position, *Ramp* has no effect on the LFO. The *Ramp* value is displayed in milliseconds in its parameter field.

An LFO normally oscillates constantly. On percussive signals it can, however, be interesting to limit the LFO cycles to a defined number. Ultrabeat allows you to set the number of LFO cycles with the *Cycle* parameter. After completing the defined number of cycles, the LFO stops oscillating.

Note: Try small *Cycle* values, and route the LFO to *Osc Volume* to create typical drum flams or hand claps.

Note: The *Cycle* = 1 setting allows the LFO to function as an additional (albeit simple) envelope generator.

The range of *Cycle* values extends from 1 to 100. Turning the knob to its maximum value (all the way to the right) results in a permanent oscillation (an infinite number of cycles).

The *Cycle* parameter can also determine whether the LFO (waveform) is started from the beginning (at a zero-crossing point) with each note trigger, or whether it simply continues oscillating. A *Cycle* value of *Inf* (*Infinity*) forces the LFO to run freely. It is not reset by incoming MIDI note on messages. When *Cycle* is set to values under 100, the LFO will be reset by each new MIDI note on message (Note On Reset).

It's really a question of taste as to whether or not you choose to trigger an LFO cycle from the same spot or just allow it to oscillate freely, regardless of phase. The random element of free-running LFOs can make many sounds fatter. This, however, can come at the expense of a percussive attack—an often undesirable quality in a drum synthesizer.

Note: You can, of course, use minor shifting of the LFO phase—with the *Cycle* value set to *Infinity*—to your advantage, adding an “analog” character to a drum sound, for example.

Env 1 to 4

Further modulation sources available to you in the *Mod* menu include four identically specified envelope generators. Envelope parameters are described in this section.

Note: In addition to its potential use in the *Mod* menus of various sound parameters, Env 4 is permanently connected to the *Voice Volume*. In other words, Ultrabeat has a “hard-wired” volume envelope generator.

Structure of the Envelope Generators

The default setting of the envelope generators is known as the *one shot* envelope mode: after a key is pressed (note on message), the envelopes run their course, regardless of how long the note is held. This setting is ideal for percussive signals because it allows simple emulations of the behavior of natural percussion sounds. For special cases such as sustained pad or cymbal sounds, you can activate a sustain mode where the envelopes take the lengths of the played notes into account.

Editing the Envelopes Graphically

Before delving into the individual parameters, please take the time to familiarize yourself with the graphical depiction of an envelope, shown below.

Ultrabeat's envelope window provides a new type of envelope design, consisting of Bezier curves in which two segments—attack and decay—make up the entire envelope.



In the envelope graphic, you can see various junction points of two different sizes. Both of the larger handles on the x-axis (the horizontal, or time axis) control the attack and decay times, respectively. A vertical line extends up from the first of the two handles, and divides the envelope into an attack and decay phase. Both segments each have two small curved junction points. You can move these in any direction to deform the contour of the envelope, and freely shape its amplitude.

In order to move the curved junction points, simply grab them and drag them to a new position. Experiment with the different junction points and you'll quickly realize that handling is very intuitive. You can also directly click-hold and drag any curve edge on the envelope itself.

Envelope Parameters

In order to edit the envelope parameters, first select an envelope by clicking on the desired 1 to 4 buttons. The parameters of the corresponding envelope can now be directly changed in the envelope display window.

Attack Time

Attack time defines the period of time the envelope needs to reach its maximum value. This is measured from the instant you press a key (note on). This period is called the attack phase.

Grab the attack junction point (the left-most of the two handles found on the x-axis) with the mouse, and move it to shorten or lengthen the attack time.

Note: To change the shape of the envelope in the attack phase, you can edit both junction points found in this segment. It is also possible to directly grab the curve with the mouse and alter it.

Decay Time

Decay time defines the period of time the envelope needs to fall back to a zero amplitude, after it has reached its maximum value (defined in the Attack phase). Moving the second junction point on the x-axis shortens or lengthens the decay phase.

Note: To change the shape of the envelope in the decay phase, you can edit both junction points found in this segment. It is also possible to directly grab the curve with the mouse and alter it.

Envelope Modulation

The time and shape of the envelopes can be modulated by velocity. Clicking in the menu field below the envelope 1 to 4 buttons opens the *Env Mod* menu. Choose either *Time* or *Shape* of the (A)ttack or (D)ecay phase as the modulation target.

The intensity of modulation is adjusted with the *mod* slider below the envelope display.

Note: When you modulate *Shape*, low velocity values lead to a “sagging” envelope shape, while higher values cause the selected envelope segment to “bulge”.

Note: When you modulate *Time*, increasing velocity values lead to a reduction in length of the envelope segment. Lower velocity values increase the length of the envelope segment.

Sustain

Activation of the *Sustain* button causes a red handle (and vertical line) to appear on the x-axis. This can be moved horizontally—but only within the decay segment area. The amplitude that the envelope reaches at the *Sustain* junction point is retained until the MIDI note is released. After receiving the MIDI note off command, the envelope continues for the remaining decay time.

Note: If the *Sustain* button is not activated, the envelope functions in “one shot” mode, and the note length (MIDI note off command) is disregarded.

Zoom (to fit)

When you select the *Zoom* button, the envelope is enlarged to fill the entire width of the display, making it easier to adjust junction points and curves. The graphic display is quickly redrawn after any change is made to the *Attack* or *Decay* values.

Note: When the *Zoom* function is selected, the decay junction point can be dragged beyond the right-hand edge of the display area, in order to lengthen the decay time. After you release the mouse button, the envelope graphic is automatically resized to fit the display area.

Zoom A/D

The *Zoom A* button only shows the attack phase across the entire width of the display, and the *Zoom D* button only shows the decay phase. This allows easier and more accurate edits to envelope shapes, even down to millisecond values.

The Step Sequencer

The integrated step sequencer allows all Ultrabeat sounds to be combined in sequences, based on patterns. Its design and use (step programming input) are based on analog predecessors.

Dependent on your personal taste and favored musical style, you'll want to control Ultrabeat from either the integrated step sequencer or from Logic, when programming rhythms. Combining both sequencers is also possible; they both can be active at the same time, and are automatically synchronized with each other. Logic's song tempo stipulates the tempo of Ultrabeat's internal step sequencer.

Before we turn our attention to Ultrabeat's step sequencer, let's take a brief look back at the early days of sequencing.

The Step Sequencer Principle

The basic idea behind analog step sequencers was to set up a progression of control voltages, and output these step by step. In early analog sequencers, three control voltages were usually created per step, in order to drive different parameters. The most common usage was control of a sound's pitch, amplitude, and timbre (*Cutoff*) per step.

The control surface of analog sequencers often contained three rows of knobs or switches on top of (or beside) each other, each row with 16 steps. Each row had its own control voltage output, and the parameter it controlled was determined by the control input (on the synthesizer) it was connected to. A trigger pulse determined the step tempo. A “running light” (an LED) indicated the current step. This principle helped to create an electronic music style whose appeal stems from the mesmerizing effect that repeating patterns can have.

The “running light” programming concept also appeared in later drum computers. The most well-known representatives of this category being the very popular Roland TR series drum machines.

The introduction of the MIDI standard and increasing use of personal computers in music led to a rapid decline in the step sequencer and related technology. More modern concepts that didn't adhere to the step and pattern principle came into vogue.

Despite this, step sequencers haven't disappeared completely. Hardware “groove boxes” have experienced a renaissance over the last couple of years. Their intuitive nature has made them a favorite tool for rhythm programming.

Ultrabeat provides an integrated step sequencer of the newest design, that time-warps the advantages of its analog predecessors to the present day. As part of the “dynamic duo” with Logic, it raises modern rhythm programming to a new level.

Step Sequencing with Ultrabeat

Ultrabeat's step sequencer contains 24 patterns—each consisting of 32 steps. You can play 25 Ultrabeat sounds per step, each sound completely independent of the others.

The Control Surface of the Sequencer

The sequencer is divided into three sections.

- On the left you'll find the parameters that globally control the pattern and sounds, independent of the individual steps and patterns. These parameters are known as the *global parameters*.
- At the bottom you'll find the parameters of the currently selected pattern (*pattern parameters*).

- The actual sequencing takes place in the *step grid* above. In this section, a pattern of 32 steps is shown for each sound. The pattern grid of the sound that is currently selected in the *assignment* section is shown. You can add or remove events to the grid by simply clicking at the desired step position. Parameter values in the individual steps are altered by grabbing and moving them with the mouse.

Global Parameters

A description of the parameters that apply globally to all internal sounds follows.

On/Off button

This button turns the step sequencer on or off.

Transport Button

The Transport button starts and stops the sequencer pattern. This allows you to inspect the step sequencer pattern while the Logic song is stopped, for example. The Transport button changes appearance, dependent on its current operating status.

The Transport button changes color when the step sequencer is, or is not, waiting for incoming MIDI notes:

- Gray: The sequencer functions independently of incoming MIDI information.
- Blue: The sequencer interprets incoming MIDI notes between C-1 and B0 as performance information. In this mode, MIDI messages can modify numerous functions that control how the patterns are played.

Note: The step sequencer is always synchronized to Logic's song tempo.

Swing

This rotary knob globally determines the swing intensity for all sounds that have the Swing function activated (see "Swing Enable" on page 337).

Swing changes the distance between notes: notes on odd-numbered steps remain unchanged, while even-numbered notes are slightly shifted. At a 0 setting (the knob all the way to the left), the Swing function is not active. Turning the Swing knob up shifts the affected notes towards the following note.

Note: Swing is only active at grid resolutions of 1/8 and 1/16.

Pattern Parameters

A pattern has a maximum number of 32 steps and contains the total events of all 25 sounds. At the bottom edge of the Plug-in window you can select from one of 24 patterns, and set global parameters (for each pattern) that apply to all sounds.

Pattern # (Pattern Number)

Click the field next to the Pattern # label to open the Pattern menu, which allows you to choose one of the 24 patterns.

If you've activated the function that allows you to switch between patterns via MIDI notes (the Transport button turns blue), *Pattern #* indicates which pattern is currently active (being played).

Control-clicking in the Pattern menu field opens a context menu where you can access further pattern-specific commands (*Copy, Paste, Clear*).

Resolution

This parameter field displays the resolution of the pattern. It defines the metric unit of a measure that is represented by the individual steps. Clicking the *Resolution* button opens a pop-up menu that allows you to switch between different settings.

The 1/8 setting means that each step of the grid represents an eighth note. Given a pattern length of 32 steps, the pattern would run for 4 measures ($32 \div 8$).

The *Resolution* parameter applies to the entire grid, and therefore, equally to all sounds.

Length (Number of Steps, Pattern Length)

This parameter defines the length of the pattern. The length of the grid can be adjusted by dragging the value in the *Length* parameter field or the bar beneath the swing buttons.

Note: The interplay between the *Length* and *Resolution* values allows the creation of different kinds of time signatures. Here are a few examples: The values *Length* = 14 and *Resolution* = 1/16 result in 7/8 time, *Length* = 12 and *Resolution* = 1/16 in 3/4 time, and *Length* = 20 and *Resolution* = 1/16 in 5/4 time.

Accent

Individual steps can be strongly emphasized, or accentuated.



In order to program an accent, click on the blue LED above the desired step. The accent applies to all sounds that fall on this beat, and the step in question is simply played louder. The *Accent* slider to the left of the chain of blue LEDs globally determines the volume of the programmed accents.

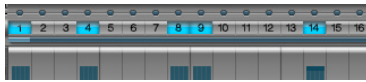
Step Grid (The Heart of the Sequencer)

In the *step grid* area, the pattern is rasterized and displayed in numerous rows and steps. The displayed grid applies to the sound that is currently selected in the assignment area. Choosing a different sound (either via the onscreen keyboard/assignment section or via MIDI) switches the grid display to match the newly chosen sound.

The step grid area contains two rows—each with 32 fields:

Trigger On/Off (The Trigger Row)

In this row, consisting of the 1 to 32 buttons, triggers are placed on corresponding steps. In other words: this is where you designate when (on which beat) the selected sound plays.



A click on one of the 1 to 32 buttons activates or deactivates the sound on that corresponding beat. In the example shown above, these are steps 1, 4, 8, 9, and 14.

Note: Dragging the mouse across the buttons allows corresponding triggers to be quickly turned on or off.

Trigger Context Menu

Control-clicking on one of the Trigger buttons opens the context sensitive Trigger menu, which offers the following options:

- *Alter*: Randomly reorders the sequencer steps while retaining the number of activated triggers.
- *Randomize*: Randomly reorders the sequencer steps; the trigger row is also altered by randomly erased and added events. In other words, a brand new sequence is created.
- *Shift Left*: Shifts the sequencer data one step to the left.
- *Shift Right*: Shifts the sequencer data one step to the right.
- *Copy*: Copies all activated triggers to the Clipboard.
- *Paste*: Pastes all activated triggers from the Clipboard.
- *Clear*: Turns off all activated triggers.

Swing Enable

Activation of the blue *Swing* button to the left of the trigger row stipulates that the grid of the currently selected sound will be played in accordance with the *Swing* knob setting.



Only even-numbered steps are affected by the *Swing* parameter; exactly which beats this corresponds to depends on the selected *Resolution* parameter setting, as demonstrated by the following example.

At a *Resolution* of 1/8 and a *Length* of 8, the notes on steps 1, 3, 5, and 7 represent quarter notes in the measure. These remain unchanged. Only the eighth notes found between them (steps 2, 4, and so on) are shifted by the *Swing* function. The amount of shift is equal to the swing intensity (see “*Swing*” on page 334).

Note: Swing is only active for grid resolutions of 1/8 and 1/16.

Velocity/Gate Row (The Row for Velocity and Note Length)

In this row, you set the length (*Gate Time*) and the *velocity* of the notes entered in the trigger row. Both parameters are displayed as a single graphical bar. The bar’s height represents the velocity, it’s length depicts the note length (*Gate Time*).



Click-dragging on the desired bar allows you to change the length and velocity values for each step.

Note: The *Gate time* is divided into four equal sections, making it easy to set rhythmically accurate note lengths. In order for the “one shot” envelope to react to *gate time*, it is necessary to either; activate the *Gate* function in the sound itself (see “Trigger and Group Menus” on page 324) or use envelopes in sustain mode (see “Sustain” on page 332), in conjunction with rhythmically useful (short) decay times.

Reset

The *Reset* button located to the left of the Velocity/Gate row returns all velocity and gate values to their default settings.

Velocity/Gate Context Menu

Control-clicking on steps in the Velocity/Gate row opens a context-sensitive menu which offers the following options:

- *Alter Vel(ocities)*: Randomly changes the velocity values of all steps while retaining the selected beats (the trigger row remains unchanged).

- *Alter Gate (Time)*: Randomly changes the note lengths of all steps while retaining the selected beats (the trigger row remains unchanged).
- *Randomize Vel(ocities)*: Same as *Alter Velocities*, but random parameter alteration is more pronounced.
- *Randomize Gate (Time)*: Same as *Alter Gate*, but random parameter alteration is more pronounced.

Using MIDI to Control the Sequencer



As mentioned earlier, pattern performance can be influenced by incoming MIDI notes. This allows you to spontaneously interact with the step sequencer, making Ultrabeat an excellent live performance instrument. The manner in which Ultrabeat reacts to MIDI control is determined by the *Pattern* and *Mute Mode* buttons and the *Playback* menu.

Pattern Mode

The *On* setting allows patterns to be switched or started via incoming MIDI note on/off commands.



In this mode, the Transport button turns blue to indicate that it is ready to receive incoming control commands.

MIDI notes C-1 to B0 switch between patterns: C-1 selects pattern 1, C#-1 pattern 2 and so on up to pattern 24, selected when MIDI note B0 is received.

Playback Mode

Pattern reactions to incoming MIDI notes is set in the *Playback Mode* menu.



The menu is opened by clicking on the field to the right of the *Pattern Mode* label on the bottom-right edge of the Plug-in window. You will find the following options here:

- *One Shot Trigger*—The reception of a MIDI note starts the pattern, which plays once through its cycle, then stops. If the next note is received before the pattern has reached its final step, the new note stops playback of the first pattern and the next pattern begins playing immediately (this can be a different pattern or the same pattern, depending on the MIDI note received).

- **Sustain**—The reception of a MIDI note starts the pattern and it continues playing in an infinite loop until the corresponding MIDI note is released.
- **Toggle**—The reception of a MIDI note starts the pattern and it continues playing in an infinite loop until the next note is received. If it is the same note, the pattern stops immediately. If it is a different note, the sequencer immediately switches to the new pattern.

Note: In *Toggle* mode, you can confidently switch between patterns in the middle of a bar—the sequencer will stay in time and will automatically jump to the corresponding beat of the new pattern. This isn't the case in *One Shot Trigger* mode, however, because the moment you switch in this mode, the new pattern is started from the beginning.

- **Toggle On 1**—The behavior is as per *Toggled* mode except that the pattern change or stop occurs the next time beat 1 is reached—at the beginning of the next pattern cycle.

Voice Mute Mode



- **Off**—This is the default setting. It means nothing more than the fact that Ultrabeat's sounds can be triggered by MIDI notes. The sounds can be played from a MIDI keyboard, starting with note number C1, upwards.
- **On**—Playing a MIDI note starting at C1 and upwards mutes the corresponding sound in Ultrabeat's mixer. A subsequent MIDI note of the same pitch un-mutes it.

This setting is ideal for spontaneous arranging of pre-programmed patterns, and muting single elements of a pattern without deleting them. This is especially useful in a live performance—but not only there. Triggering the step sequencer via MIDI notes opens up a number of remixing possibilities. Ultimately, all of the creative pattern switching options discussed in this section is achieved through the use of MIDI note messages and can therefore be simply recorded, edited, arranged, and automated in Logic.

Importing Sounds

Ultrabeat allows you to import individual sounds.

Please proceed as follows:

- 1 Click on the *Import* field to the right of the *Voice Auto Select* function.



- 2 Navigate the dialog box as you would normally until you find the setting that you want to import sounds from.
- 3 After selecting the setting, a list of all the sounds found in this setting will open up in a “drawer” next to the mixer section.



- 4 Select a sound in the list and click on its name while holding down the Control key.
- 5 A context menu opens. Use the *Copy* command to copy the selected sound to the Clipboard.
- 6 Now select the sound you want to replace in the current drum kit in the assignment section to the left.
- 7 Click on its name while holding down the Control key. A context menu opens. Use the *Paste* command to replace the sound you selected (also see “Naming and Organizing the Sounds” on page 304). This facility allows you to create a new drum kit that contains sounds from several other drum kits.

Note: In the sound import process described above, existing step sequencer data is neither copied nor replaced.

Previewing Sounds

Before importing a sound, you can preview it by clicking on its name in the import list.

Tutorial: Creating Drum Sounds in Ultrabeat

Now that you're acquainted with all of Ultrabeat's features, we'd like to offer you a few specific sound creation tips in the following section. Please take the time to explore the vast and complex possibilities available to you in Ultrabeat, using the following programming tips as a starting point. You'll discover that there is hardly a category of electronic drum sound that Ultrabeat can't create easily.

Before we jump into sound programming, we'll briefly discuss how sounds are selected.

To select sounds, proceed as follows:

- The 25 sounds of an Ultrabeat drum kit are mapped to the onscreen keyboard found on the left hand side of the Plug-in window. The order of sounds on the keyboard corresponds to notes on a connected MIDI keyboard, starting with C1 for the first (bottom) sound. Clicking on the name of a sound selects it for editing. The sound's parameters are displayed to the right of the keyboard, and can be edited

Note: Make sure that the sound you're playing via MIDI is also the one you're editing: you can recognize the selected sound by the frame that appears around its name in assignment area. The corresponding key on the onscreen keyboard to the left of the sound name turns blue when it receives appropriate MIDI information. Clicking on these keys with the mouse will play the sounds directly in the Plug-in window.



In this example, drum voice 2 is being played (the blue key) while drum voice 4 is selected (the red frame).

When the *Voice Auto Select* function is switched on, the sound played via MIDI is selected for editing.

Time to move on to sound programming! First, we'll analyze several classic electronic drum sounds, and show you how you to accurately recreate them using the many features available to you in Ultrabeat. We'll also take a look at additional refinements you can make.

Note: In Ultrabeat's Settings folder, you will find a drum kit called *Tutorial Kit*. This drum kit contains all drum sounds discussed in this tutorial. It also includes a drum sound called "Default Tut(orial)", which is a default set of "neutral" parameters that provide an excellent starting point for many of the following examples.

Kick Drum

Electronically produced kick drum sounds are primarily based on the sound of a deeply tuned sine wave.

To program this type of sound in Ultrabeat, please proceed as follows:

- 1 Load the Default Tutorial setting. Note that Oscillator 1 is in Phase Oscillator mode.
- 2 Find a suitably tuned pitch in the lower octaves by soloing the bass drum along with other important tonal elements of the song (a bass or pad sound, for example). Use the *Osc 1 Pitch* slider to adjust the pitch until appropriate.
- 3 Use Env 4 to shape the volume of the bass drum. For slower beats you'll want a longer *Decay* phase, while at faster tempos you'll choose a shorter *Decay* time. The Attack time of Env 4 should be very short in any case (zero, in most cases) or the sound will lose its percussiveness, and its ability to be clearly heard in the mix.

Our kick drum still sounds very soft and is somewhat reminiscent of the famous TR 808 bass drum. It's still missing a clearly defined attack. In order to give the bass drum more "kick," we'll control the pitch of the oscillator with an envelope.

To accomplish this, please proceed as follows:

- 1 Ensure that *Env 1* is chosen in the *Mod* menu of Oscillator 1's *Pitch* parameter.
- 2 Set the degree of modulation by moving the blue Mod slider approximately 3–4 octaves above the original pitch.



- 3 Set the *Attack* time in Env 1 to zero by sliding the leftmost of the two junction points that sit on the x-axis all the way to the left.



- 4 Now experiment with the *Decay* time by moving the rightmost of the two junction points that sit on the x-axis; you'll discover that higher *Decay* values (shifting the Bezier handle to the right) result in sounds similar to synth toms, while shorter *Decay* values (shifting to the left) provide the "kick" character that we're after.

- 5 Change the *Mod* amount (the blue control) of *Osc 1 Pitch* again (see step 1). The interaction of this parameter with the envelope's Decay time provides numerous possibilities for shaping the “kick” or “punch” of the bass drum sound.

Note: This simple bass drum sound is called “Kick 1” in the tutorial set, at a pitch of C1.

Removing Tonality

One advantage of bass drums based on sine waves is that their sound can be precisely tuned to match the song. The disadvantage: a recognizable pitch is not always desirable. Ultrabeat offers several methods to reduce the tonality of the sound. A very effective tool is the 2 Band EQ.

Try the following setting:

- 1 Band 1 is used in shelving mode with negative Gain, high Q value and a frequency of about 80 Hz.
- 2 For band 2, choose the Peak mode at a frequency of around 180 Hz, a medium Q value and also a negative gain value.
- 3 On the EQ graph, you can see how the frequencies around 80 Hz are boosted, while the surrounding frequencies are reduced.



- 4 Vary the frequency of Band 2 now (easily recognizable in the blue part of the EQ graph) and you can influence the extent of bass drum tonality.

A further method for reducing the tonality of a drum sound that is rich with overtones is to use a lowpass filter. Control the cutoff frequency of the filter with an envelope.

Please proceed as follows:

- 1 Reload the Default Tutorial sound, choose A#0 as the basic pitch in Oscillator 1 and modulate it (as shown in the example on page 342) using Env 1.
- 2 An increase in the Saturation parameter value enhances the overtones of the drum sound.
- 3 Note, that the output of Osc 1 is directed to the filter, as the Filter Bypass button (the arrow between Osc 1 and the filter) is activated.

- 4 Make the settings shown in the following graphic to the filter section: Filter type = *LP 24*, Cutoff value = *0.10*, *Mod* Source for Cut = *Env 3*, *Mod* Amount for Cut = *0.60*, Resonance = *0.30*.



- 5 Set the Attack time of Env 3 to zero. Use the Decay time of Env 3 to shape the sound of the filtered bass drum.
- 6 You may also choose to control the filter resonance with an envelope. Make sure you dedicate a single envelope to this function (in this case, use Env 2 as a *Mod* source for *Res*). Choose a *Mod* amount for *Res* of about 0.80. Select a longer Decay time in Env 2 than in Env 3 and listen carefully to the fatter and more atonal bass drum sound achieved through this *Res* modulation (due to the higher filter resonance).

Note: The bass drum described in the above example is called “Kick 2” in the tutorial set, at a pitch of C#1. It also features an interesting EQ setting (see paragraph below).

More Bass...

Use the “Kick 2” filtered bass drum sound as a starting point, and try out the remaining parameters in the Phase Oscillator. You will discover that high saturation values make the sound rounder and add more bass, for example. The character of our example is beginning to head in the direction of a TR-909.

More Kick...

To get even closer to the 909, we recommend an EQ setting as shown in the following graphic. Note the low frequency pressure point around 60 Hz (which can be seen in the red area on the EQ graph) as well as the assertive punch or kick (the blue area starting at 460 Hz and up) of a 909 bass drum are strengthened. (This EQ setting is already part of the “Kick 2” setting.)



More Contour...

In our example, all four envelopes are being used. Take some time to play with the shapes of the envelopes, while maintaining the Attack and Decay settings. Experiment with the junction points of the Decay phase in the different envelopes to familiarize yourself with the sound shaping options available. Start with the Decay phase of Env 4 (which controls the volume of Oscillator 1 as well as filter resonance) and observe how reshaping the “belly” of the envelope can change the character of the sound from crisp and short to round and voluminous.

The Ultrabeat Kick

Let’s move on to bass drums that are uniquely “Ultrabeat”, rather than analog models. Try modulating pitch with an LFO, rather than an envelope, for example.

An appropriate LFO setting could look something like this:

- 1 Start with the Default Tutorial sound at a pitch of A#0 (*Osc 1 Pitch*), and choose LFO 1 as the Mod source in the *Osc 1 Pitch* section.
- 2 Set the degree of modulation by moving the blue Mod control to a value of A3.
- 3 LFO 1 should be set to a low number of *Cycles* (25 to 35), a high *Rate* (starting with 70 Hz and higher) and a medium value for *Decay* (*Ramp* rotary knob at about –190).
- 4 Experiment with the LFO waveform and you’ll discover that you can attain different nuances in the character of the bass drum attack.
- 5 Now modulate the *Asym* (Asymmetry) parameter with the same LFO, and also vary the *Slope* and *Saturation* values. This method enables you to create very different bass drum sounds with a single oscillator, one LFO and one envelope (for volume). The character of the sounds can range from soft to punchy and the degree of tonality in the sound can be adjusted to taste.

Note: The bass drum sound described is called “Kick 3” in the tutorial set at a pitch of D1.

Use the second oscillator (with similar settings or with a sample) or use the filter and the ring modulator—the sky’s the limit as far as your imagination is concerned, so get on with it, and create that next “gotta have it” drum sound.

Note: You can find an “emulation” of the legendary 808 bass drum under the name “Kick 4” in the tutorial set, at a pitch of D#1.

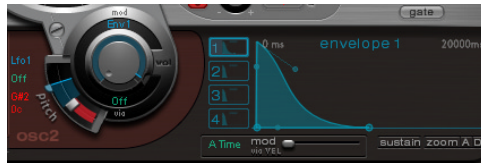
Snare Drum

The sound of an acoustic snare drum primarily consists of two sound components: the sound of the drum itself and the buzzing of the snare springs. Try to approximate this combination in Ultrabeat with a single oscillator and the noise generator.

Start programming your snare drum as follows:

- 1 Begin again with the Default Tutorial setting. Deactivate Oscillator 1, and switch Oscillator 2 on (in Phase Oscillator mode).
- 2 To get rid of the sine wave (which is not especially desirable for a snare sound, in contrast to the bass drum), modulate *Osc 2 Pitch* with a rapidly vibrating LFO with a medium *Ramp Decay* value. To accomplish this, select LFO 1 in the *Mod* menu of *Osc 2 Pitch*. The pitch value for *Osc 2 Pitch* should be around G#2 and the *Mod* amount (the blue Mod control) should be about 3–4 octaves higher.
- 3 LFO 1 should be set to a high Rate. Choose a value of 20 for *Cycles* and –20 for *Ramp*. The LFO *waveform* parameter should be set to a value of about 0.58, which is a square wave.
- 4 Use Env 1 to control the volume of Oscillator 2 by setting *Vol* to the lowest possible value (–60 dB), selecting *Env 1* in the *Mod* menu and adjusting the modulation intensity to a point just below its maximum value.

The GUI detail shows the settings of Oscillator 2 and Env 1 described in steps 2 and 3.



- 5 Experiment with different *Slope* and *Asym* values to impart a more or less “electronic” character on the sound.
- 6 Now turn on the noise generator and control its volume with the same quick envelope used in *Osc 2 Volume*.

Use the filter parameters of the noise generator to either roughen up, refine or add bright frequencies to the noise component of the snare drum sound. Select a LP filter type, and try a filter frequency between 0.60 and 0.90. Modulate it with LFO 1 that you’re already using to control the pitch of Oscillator 2.

Note: The snare drum sound is called “Snare 1” in the Tutorial Kit, at a pitch of E1.

Now refine the snare drum sound using FM synthesis:

- 1 Turn on Oscillator 1 in *FM* mode. Use Env 1 to control the volume of Osc 1 as well.
- 2 Choose a pitch for Oscillator 1 that's about an octave lower than Oscillator 2. Consciously avoid even intervals between the oscillators and detune them slightly from each other. As an example, try a pitch setting of F#2 in Osc 2 and E1 in Osc 1, then fine tune Osc 1 a few cents higher by holding Shift while adjusting its Pitch slider.
- 3 Now experiment with *FM Amount*, and add more tone (low *FM Amount* value) or noise (more *FM Amount*) as desired. Also try modulating the *FM Amount* with a fast LFO.

Note: An exemplary snare drum sound that makes use of FM can be found in the tutorial set at a pitch of F1. It is called "Snare 2".

Higher *FM Amount* values lead to considerably more overtones and a very electronic sound character. If you want to make the sound more acoustic, we recommend feeding Oscillator 1 (and possibly Oscillator 2 as well) into the main filter. Use these settings to start: *LP 24* mode, a *Cutoff* value of about 0.60.

The 808 Snare

The famous 808 snare is based on two resonating filters and a noise generator, fed through a highpass filter. The mix ratio of the two filters and the volume of the noise generator can be adjusted. This structure cannot be 100% replicated in Ultrabeat.

To "clone" the 808 snare sound, proceed as follows:

- 1 Begin again with the Default Tutorial setting.
- 2 Both of the resonating filters of the 808 snare can be replicated by two cleverly programmed Phase Oscillators. They are given slightly different *Slope* values and are detuned by almost an octave. The tonal relationship between the oscillators should also be uneven here—from E3 to F2, for example.
- 3 The volume of each oscillator should be controlled by a different envelope. The envelope for the lower tuned oscillator should have a longer decay time than the very "snappy" envelope setting for the higher oscillator.
- 4 Feed the output of both oscillators into Ultrabeat's main filter, and "hollow out" the sound with a highpass filter. Activate the Filter Bypass button in both oscillators (the arrow between the oscillator and the filter). Choose the *HP 12* setting in the filter, a *Cutoff* value around 0.40 and a *Resonance* value of about 0.70.

You have just very cleverly emulated both of the 808's resonating filters. Shifting the pitch of both oscillators simulates the behavior of the 808's *Tone* control by the way.

To complete the 808 emulation, add some noise:

- 1 Switch the noise generator on, and activate the highpass mode in its filter (*HP*). Set the *Cutoff* value to about 0.65, *Resonance* to 0.35 and add a little *Dirt* (around 0.06).
- 2 The noise generator provides the sustained snare sound. It should be shaped by its own envelope, independent of the decay phases of both oscillators, in order to get 808-like results. Changing the volume of the noise generator simulates the “snap” parameter of the 808.

Note: The 808 snare drum described is called “Snare 3-808” in the Tutorial Kit, at a pitch of F#1. It also features an interesting EQ setting.

Dynamics through Velocity

Using the 808 snare drum sounds in our tutorial kit, we’d like to explore the possibilities Ultrabeat offers for implementing velocity.

To implement velocity, proceed as follows:

- 1 Select the sound “Snare 3-808”.
- 2 Click on the word *Off* below the Volume knob in Oscillator 1.



- 3 In the Via menu that appears, choose *Vel*.
- 4 A slider appears on the ring around the knob. Move it in a clockwise direction. When you grab the marking, a help tag displays the value. Set it to –0 dB.



- 5 Repeat steps 1–4 in both Oscillator 2 and the noise generator. You can now dynamically play the sound using velocity.

You can increase the performance dynamics with the following steps:

- 1 First, reduce the values of the individual volumes by turning down the Volume knobs in both oscillators and the noise generator. Note how the Mod ring and its Via sliders also move back. Change the Via slider positions until all three Volume knobs look like this:



- 2 If you use differing intensities for each Volume knob when completing step 1, you'll have the potential of individual velocity reactions for each sound component.
- 3 The dynamics of the sound as a whole can be further increased by the following setting to the Voice Volume knob:



You now have an 808 snare that is exceptionally responsive to velocity. As you may know, this wasn't possible with the original—not even an 808 sample could offer the dynamic volume control of individual sound components that's demonstrated here. A sample only offers you the whole sound, not its constituent parts.

In the next step, you'll use velocity to control the character of the sound—individually for each component—plus volume, of course:

- 4 In the Saturation Mod menu of Oscillator 2, choose *Max* and then velocity in the corresponding Via menu.



- 5 Set the additional control that appears (as shown), to control the character of the sound with velocity:



- 6 Repeat this with the other parameters of Oscillator 2, as well as pitch:



- 7 Within the noise generator, you'll be working with negative modulation (the position of the marking is below that of the base parameter value), and will modulate one parameter directly (*Max* setting in the *Cut Mod* menu), and another indirectly (the *LFO 2* setting in the *Dirt Mod* menu):



Our sound is now nothing like an 808 snare—and that’s exactly what we wanted to achieve. Keep experimenting with velocity and figure out when it makes sense to use it as a direct or indirect modulation source, in either its positive or negative form.

Note: The dynamic 808 snare drum described above is called “Snare 4 - vel” in the tutorial set, at a pitch of G1.

The Kraftwerk Snare

A further classic electronic snare drum sound is the highly resonant lowpass filter of an analog synthesizer that quickly closes with a “snap”. This sound was used extensively by Kraftwerk.

Proceed as follows to recreate this sound with Ultrabeat:

- 1 Choose the “Snare 1” sound to start with.
- 2 Direct the signals of both oscillators and the noise generator to the main filter.
- 3 Modulate *Cutoff* with Env 1 (this is already modulating the volume of the noise generator).
- 4 Modulate the filter resonance with Env 2.
- 5 Now begin to experiment with the parameters described in steps 1–5 (especially the envelopes), introduce EQ into the sound, and discover how much “playing room” these basic settings allow you.

Note: An exemplary sound called “Snare 5 - KW” is in the tutorial set, at a pitch of G#1. Analyze, and compare it to your own creation.

Toms and Tonal Percussion

Tonal percussion sounds such as toms or congas are relatively easy to emulate electronically with sine or triangular wave oscillators. Ultrabeat’s Phase Oscillator offers you a broad spectrum of suitable basic sounds with which to start. Control the pitch of the oscillators with envelopes, and use the programming techniques discussed in the bass and snare drum sections to alter tonality. You should find it easy to create a broad range of toms and similar sounds.

Note: At the pitches A1 to B0 in the tutorial set you’ll find typical 808 toms. Analyze these sounds and modify them as you see fit.

At this point, spend some time experimenting with the *Model* mode of Oscillator 2. Try to familiarize yourself with the effect of each parameter, and create some of your own tonal percussion sounds, ranging from small tabla drums to glass bowls.

Note: The “Tabla” and “Glass” sounds (at pitches C2 and C#2) of the tutorial set combine both Osc 2 Model and FM. They are also good examples of the complex use of velocity as a modulation source.

Hi-Hats and Cymbals

Electronic hi-hat sounds are very easy to create in Ultrabeat.

Please proceed as follows:

- 1 Load the Default Tutorial sound.
- 2 Switch off Oscillator 1 and turn on the noise generator. Choose the following settings for the noise generator:



- 3 In the GUI detail above, you can see, that the Cutoff parameter is modulated by Env 1. The modulation is negative, the position of the Mod slider is below that of the base parameter value.
- 4 Use rather short Decay values for Env 1 and Env 4 ().
- 5 Set the Attack time of Env 4 to a value of 0. The Attack time of Env 1 should also be rather short, but not equal to zero.

Note: You'll find a similarly constructed sound called "HiHat 1" at a pitch of F2 in the tutorial set. Also analyze the hi-hat sound Hihat 2 at a pitch F#2.

It's not far from the hi-hat to the crash cymbal: the main difference between a hihat and crash cymbal sound is the length of the decay time. Correct assignment of the envelopes is the key to producing different cymbal sounds.

Select the "Cym 1" and "Cym 2" sounds in the tutorial kit and try different envelope assignments and settings for *Cutoff* and *Volume* in the noise generator, *Cutoff* and *Volume* in the main filter and so on.

Metallic Sounds

If you want to create metallic sounds with Ultrabeat, the ring modulator and the Model oscillator are the ideal tools.

To use the ring modulator, proceed as follows:

- 1 Load the Default Tutorial sound.
- 2 Activate a Phase Oscillator and the Model oscillator. Choose a pitch for each oscillator above C3 so that a slightly detuned interval is created.
- 3 In the Material Pad of the Model oscillator, choose a setting with plenty of overtones as in the graphic below.



- 4 Set the volume of each oscillator to a value of –60 dB and turn the ring modulator on by clicking on its name.

You’ve just created a bell-like sound that you can filter (with a high resonance value) if required.

Note: You can find a similar sound called “Ring Bell” at a pitch of A2 in the tutorial set.

Clicks and Cuts

Ultrabeat features extremely fast envelopes and uncommonly powerful LFOs. Use these modulation sources to perform extreme modulations of the oscillator and filter parameters. The key to creating “out of the ordinary” sounds is to try modulating as many targets as possible, and not to be afraid of using extreme settings: use a quick envelope to drive the filter to self oscillation for a fraction of a second, use a few LFO cycles at a much higher rate than other cycles, or experiment with the Dirt parameter or the bitcrusher.

Programming in Building Blocks

As you become familiar with drum sound programming, you may begin thinking in “building blocks”. By this we mean that you might realize that drum sounds usually consist of different components. Once you’ve mentally, or physically, written down your “list” of components, you should try to emulate each *component* that contributes to the sound’s character—making use of the different sound generators available in Ultrabeat. Assigning dedicated (amplitude) envelopes to the different components allows you to control their temporal behaviour individually. As an example: You can emulate the body of a drum with Oscillator 1, the sound of the stick hitting the skin (or first transient) with the noise source, additional overtones and harmonics can be provided by Oscillator 2 and/or the ring modulator. Once you begin thinking that drum sounds consist of several building blocks or layers, the design of the Volume controls in the individual sound generators might make more sense to you, as this is the place where the blocks are combined, balanced, and controlled.

We hope to have given you a few insights into the interplay of Ultrabeat’s functions and parameters. We also hope that you’ve gained some inspiration and insights for programming your own sounds. We would also ask that you further explore all of Ultrabeat’s functions at your leisure. The included settings will afford you some interesting insights and hopefully, further stimulation. Spend some time analyzing Ultrabeat’s presets and ask yourself “How did they do that?”—then figure it out.

Have fun creating your own Ultrabeat sounds and sequences!

Sculpture is a synthesizer plug-in that generates its sound based on a simulated vibrating string or bar.

To keep things clear, we will always refer to the “String” throughout this chapter, even though many of the possible sounds that you can create with Sculpture have nothing in common with what you’d expect from a stringed instrument!



The Graphical User Interface (GUI) of Sculpture is broken down into three main areas. The silver section at the top contains the sound engine. The blue/gray area below houses the polyphonic modulation sources and Envelopes. Global control sources, incorporating the Morph Pad, are located on the bottom, gray ledge.

Most software synthesizers emulate real-world hardware, and generate sound based on samples or emulations of the signals generated by analog oscillators.

Sculpture uses a method of synthesis called component modelling. This approach to tone generation shares some aspects and parameters with other synthesis techniques, such as those found in additive and subtractive synthesizers. As such, many of the parameters used by Sculpture will be immediately familiar to you, such as LFO's, Vibrato, Envelopes, and so on. Many others, however, will be very new.

Fortunately, the GUI of Sculpture is incredibly elegant and intuitive to use, so you should find yourself creating amazing sounds in next to no time.

Sculpture's recordable envelopes provide massive scope for sonic animation, with the payoff being some just plain astonishing sounds! Sculpture's flexibility in this area provides a new level of control that we're sure will amaze and delight you.

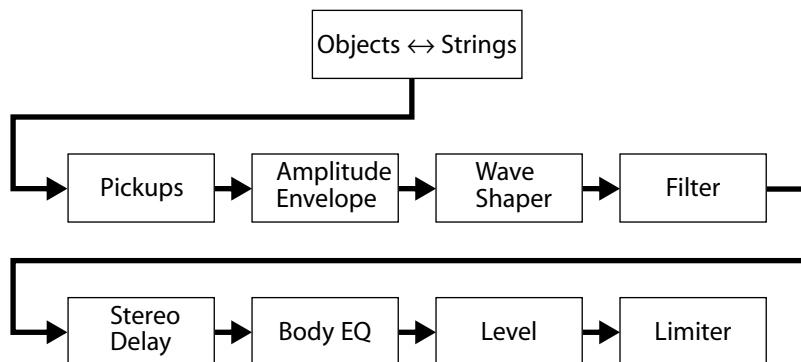
We encourage you to make full use of every control and parameter that is available to you—both when initially auditioning some of the supplied factory sounds, and when creating new ones of your own. Don't be afraid to experiment with sound design—that's what Sculpture was created to do!

Before taking a look at the product features, we'd like to briefly touch on the synthesis core of Sculpture. We will also take a look at the creation of particular types of sounds in "Programming: Quick Start Guide" on page 409.

The Synthesis Core of Sculpture

We have included this section at the top of this chapter to give you a feel for the way Sculpture works. Component modelling, as you'll discover, is quite different to "traditional" synthesis methods—and so are the results!

We have followed the signal path (shown in the diagram) of the core synthesis engine in the layout of the manual's parameter descriptions. We invite you to check out each parameter's options as you read about them. This will give you a better feel for where things are, and what's available. So let's get to it ...



The *String* is the central synthesis element. It offers a range of parameters that allow you to adjust its material—what it’s made of, in other words.

Up to three *Objects* of different types are used to excite or disturb the vibration of the String. These *Objects* can be positioned anywhere along the String, and offer multiple parameters for adjustments to their properties.

The String itself doesn’t make a sound unless it is stimulated by the *Objects*.

The vibration of the String is passed on to the ensuing processing section—via two movable *Pickups* (you can view these as being similar to the electromagnetic pickups found on guitars or electric pianos and clavichords).

The processing section consists of the ADSR-equipped *Amplitude* stage, a *Waveshaper* (with selectable types) and a multi-mode *Filter*.

Note: All elements described above exist on a per voice basis.

The sum of all voice signals is then processed by an integrated *Stereo Delay* effect.

From there, the signal is sent to an EQ-like module (*Body EQ*), which globally simulates the spectral shape/body response of your instrument. There are several body types to choose from.

The resulting signal is then fed to a *Level Limiter* section.

A vast number of modulation options are also available, from tempo-synced *LFOs* to *Jitter* generators and newly-developed recordable *Envelopes*.

A (recordable) *Morph* function also allows for smooth or abrupt transitions between (up to) five *Morph* snapshots.

A String as a Synthesis Element

As you can see, *Sculpture*, as a component modelling synthesizer, provides you with a String—rather than the oscillators found in traditional synthesizers.

The String is considerably more sophisticated in concept than simple oscillators. Basically, you are actually creating the waveform, or base timbre from nothing. This is achieved by (mathematically) describing the string’s properties, and that of its environment. These include, amongst others: the material the string is made of, the thickness and tension of the string, its characteristics over time, the atmosphere (water, air, and so on) it is being played in, and the way it is being played—struck, bowed, and so on.

In *Sculpture*, the general timbre/sound sculpting is set by the String parameters. Modules such as the *Waveshaper*, *Filter*, and *Body EQ* are useful additions.

Sculpture goes far beyond the mere creation of an infinite number of base timbres, however. One of the key differences between Sculpture's *String* and a traditional synthesizer's *waveform* is that the base timbre (provided by the String) is in a constant state of flux.

By this, we mean that if Sculpture's *String* is still vibrating for a specific note, retriggering that same note will interact with the ongoing vibration. This is not dissimilar to the effect of repeated plucking of a guitar string, where the string is still vibrating when the next note is played. This will alter the harmonic spectrum each time—which is why acoustic guitars sound organic when a note is played repeatedly, and sampled guitars don't.

As you can see, this is quite different to other synthesis methods where the base timbre (waveform), even if modulated, does not harmonically interact when retriggered. What usually happens in traditional synthesizers is that the waveform is restarted—from mid cycle, or from the beginning—with the result being an increase in volume, or a slight cyclical wave shift.

Beyond this aspect of base timbre control and interaction, Sculpture also provides you with the means to model (emulate) a number of other properties of acoustic instruments. Among these, you'll be able to emulate a bow dragging across a string, a piano hammer strike, or even dropping a coin onto the bridge of a guitar.

There would be no point in bowing or plucking a string, however, if you didn't have a neck or a body to your violin, cello or guitar. You'd probably want some way to change the tension of the strings, or try steel, rather than nylon or catgut. In other words, to control the physical properties of the string, and its excitation.

All of this is possible with Sculpture.

It is an instrument that will require some investment of your time, but it will reward you with beautifully warm organic sounds, evolving soundscapes or a harsh and metallic "Hell's Bells" patch—if you're after that sort of thing.

Think of it as your own personal sound design and sculpting toolkit.

Sculpture's Parameters



The following sections discuss the parameters of Sculpture.

Before touching on them, however, a key factor with component modelling is the interaction between various sections of the synthesis engine. This can lead to some truly unique sounds, but can also lead to unexpected results.

Sculpture is very different to traditional synthesizers, and requires a more measured approach to achieve a particular end result. We have compiled a collection of tips in “Programming: Quick Start Guide” on page 409, that will assist you in the creation of particular types of sounds. Please ensure that you read this section.

Note: Keep the Sculpture flowchart diagram (on page 356) handy while familiarizing yourself with the interface/programming. If your approach is methodical—and you follow the flowchart, you shouldn’t encounter too many surprise results.

Parameter Use

Some additional parameters are accessible via Control-clicking (or clicking with the right mouse button, if you have a two button mouse) on particular areas of Sculpture’s front panel. This will launch a contextual menu, where you can make your choice by highlighting the desired entry and releasing the mouse button.

Sculpture’s envelopes (and other fields, such as the Material or Morph Pads) can be adjusted graphically, by click-holding on the nodes (shown as hollow or filled dots, balls or diamonds), and dragging to the desired position in the respective panels on Sculpture’s interface.

Most parameters can be reset to their default value by clicking on the desired control, while holding down Option.

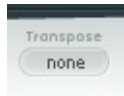
Note: When adjusting most parameters, keep an eye on the small help tag that pops up while the mouse button is depressed. This will provide you with an accurate indication of the current parameter value, allowing you to make precise changes.

Global Parameters

These are found across the top of the Sculpture GUI, unless otherwise specified.

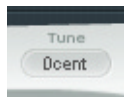
Transpose

Transpose is used for coarse tuning of the entire instrument (range: ± 2 octaves). Given the ability of component modelling to radically alter pitch with certain settings, coarse tuning is limited to octave increments.



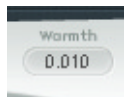
Tune

Tune is used for fine tuning of the entire instrument (range: ± 50 cents). A cent is 1/100th of a semitone.



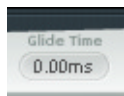
Warmth

Warmth is used to slightly detune the different voices, much like the random fluctuations caused by the components and circuitry of analog synthesizers. As the parameter name suggests, this “warms up” or “thickens” the sound.



Glide Time

Values above 0.0 lead to a softer pitch change between played notes (range: 0.00 to 5000 milliseconds).



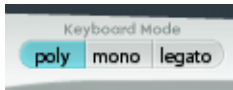
Voices

When *Keyboard Mode* is set to *Poly*, this parameter limits the number of simultaneously sounding voices to the set value. A value of 16 voices is the maximum polyphony of Sculpture.



Keyboard Mode

Here, you can select between the *Mono*, *Legato* or *Poly* Keyboard Modes.



You can switch between mono and poly modes by clicking on the *Poly* and *Mono* buttons. The *Portamento*, the duration of which is set by the *Glide Time* parameter, affects legato performances. As long as you don't release the key of the previously struck note before pressing a new key, you'll hear the glide or portamento. If you release every note before striking another key (portato or even staccato style), the glide effect will not be audible.

When the *Legato* button is activated, a further side-effect occurs. The *Amplitude Envelope* is not retriggered—the sound is sustained and the *Attack* phase of newly played notes is not retriggered. These behaviors are sometimes referred to as *Single Trigger* and *Multi Trigger* modes.

All modes simply retrigger a (potentially sounding) voice with the same pitch, instead of allocating a new one. As such, multiple triggering of a given note results in slight timbral variations, depending on the current “state” of the model at note-on time.

If Sculpture's String is still vibrating for a specific note, retriggering that same note will interact with the ongoing vibration, or current “state” of the string.

Important: A “true” retrigger of the vibrating string will only happen if both *Attack* sliders of the *Amplitude Envelope* are set to zero. If either slider is set to any other value, a new voice will be allocated with each retriggered note.

Bender Range Up/Down

These parameters are found below *Object 3*, on the left-hand side of the Sculpture GUI.



Separate settings are available for upwards and downwards pitch bends—using your MIDI keyboard’s pitch bend controller.

When *Bender Range Down* is set to *Linked*, the *Bender Range Up* value is used for both (up/down) directions.

The range is set in semitone increments: *+12* and *–36* respectively.

Note: Bending the string will, just like the real deal on a guitar, alter the shape of the model, rather than merely act as a simple pitch bend.

String and Object Parameters

The string and object parameters discussed in this section apply on a per-voice basis.

You will note a number of parameter names followed by “(morphable)”. This indicates that the parameters can be morphed between up to five snapshots, called Morph points. More detail on morphing can be found in “Morph” on page 400.

Hide/Keyscale/Release Buttons

Before beginning, we’d like to briefly cover some common parameters that you’ll encounter when manipulating the String. The String’s default note position is C3 (middle C).



Three buttons are available for your use that will activate/deactivate and hide the various Keyscaling and Release parameters. Simply click on the *Keyscale*, *Release* or *Hide* button, dependent on the adjustments you would like to make. The corresponding parameters will become visible in the ring surrounding the Material Pad.

Keyscale parameters can be set for notes that fall below C3, or notes that are positioned above it. In simple terms, the impact of these parameters can be controlled across the keyboard range. As an example: a parameter such as string stiffness (which we’ll discuss shortly) could be more intense for high notes, and less intense for low notes. In practical terms, this would result in more harmonic (“sweeter”) sounding bass notes, and inharmonic overtones in treble notes (notes above C3).

The string *Release* parameters impact on the vibrations of the string once the key has been released.

The *Hide* button is handy for avoiding accidental parameter changes, and simplifies the interface.

The Material Pad

The following two string material parameters determine the general timbre, and are controlled by the “ball” (which correlates to the X and Y co-ordinates) within the *Material Pad*.



The “crosshair” is a handle for the *Key Scale/Release Scale* diamonds in cases where these are hidden by the ball. It also allows you to independently change the keyscaling for one of the two axes (X/Y positions—Inner Loss or Stiffness).

In general synthesizer terms, this could be viewed as a base timbre generator in the Oscillator section.

Note: Option-clicking resets all string parameters to their default values.

InnerLoss (morphable)

This parameter is used for damping of the string, as caused by the string material—steel, glass, nylon, or wood. These are frequency-dependent losses which cause the sound to become more mellow during the decay phase. Simply click-hold, and drag, the *Inner Loss* and *Stiffness* “ball” to the desired position.

Inner Loss Scale Low/High

Allows independent adjustments to the key tracking of inner “losses” for notes above and below C3.

To adjust Inner Loss keyscaling, first select the *Keyscale* button, then click-hold on the *green* horizontal line for low notes or the *blue* horizontal line for high notes, and drag up/down to the desired position.

A diamond indicates the intersection between the *Inner Loss* and *Stiffness Low/High Scaling* positions. You can click-hold and drag this diamond directly, to adjust both parameters simultaneously.

Inner Loss Scale Release

Values above *1.0* cause the inner losses to increase when the key is released. This is quite unnatural, as this would mean that the string material would change after the note was released.

In practice, however, the use of this parameter in combination with *Media Loss Scale Release* allows a natural simulation of strings that are dampened at note-off time.

To adjust, first select the *Release* button, then click-hold on the *blue Release* line, and drag up/down to the desired position.

Stiffness (morphable)

Controls the stiffness (rigidity) of the string. In reality, this is determined by the string material and diameter (or to be more precise: by its geometrical moment of inertia). Stiff strings exhibit an inharmonic vibration, where overtones are not integer multiples of the base frequency. Rather, they have higher frequencies.

In effect, an increase in *Stiffness* turns the string into a solid metal bar.

Simply click-hold, and drag, the *Inner Loss* and *Stiffness* “ball” to the desired position. Note that as you do so, the thickness of the “string”—the green horizontal line in the *Pickup* window—will change.

Stiffness Scale Low/High

Allows independent adjustments to the key tracking of the stiffness parameter, for notes above and below C3.

To adjust Stiffness keyscaling, first select the *Keyscale* button, then click-hold the *green* vertical line for low notes or the *blue* vertical line for high notes, and drag left/right to the desired position.

You can also simultaneously adjust both *Stiffness* and *Inner Loss* keyscaling by dragging the diamond that intersects the green lines.

Low *Stiffness* values combined with low *Inner Loss* values lead to “metallic” sounds. Increasing the *Stiffness* makes the sound become more “bell” or “glass”-like.

Increasing the *Inner Loss* value, while maintaining a low *Stiffness* level, corresponds to nylon or catgut strings.

High *Stiffness* values combined with high *Inner Loss* values simulate “wood”-like materials.

The String Parameters Around the Material Pad

Resolution (Harmonics)

This parameter determines the maximum number of harmonics contained in (and spatial resolution of) the sound at C3. This is roughly proportional to the required CPU power, so the more harmonically-rich/the higher the *Resolution* setting of the sound, the more processing muscle will be required.



Note: As you alter the *Resolution* value, you are changing the interaction of the string with the Objects. This also changes the frequency of the overtones: very low *Resolution* values result in inharmonic spectra—even with *Stiffness* set to zero.

Resolution Scale Low/High

Allows you to set the key tracking resolution (the accuracy of key tracking) separately for notes above and below middle C (C3). The green “low” slider (down to C0), inside the Material Pad ring, is visible when the *Keyscale* button is clicked. To adjust, click-hold, and drag. The blue “high” slider (up to C6) runs along the left-hand side of the outer ring. Click-hold, and drag, to adjust.

Media Loss (morphable)

Controls dampening of the string caused by the surrounding media, for example; air, water, pea, and ham soup and so on. These “losses” are independent of frequency. This approach allows control over the duration of the (exponential) amplitude decay, once the excitation of the string has stopped.



Media Loss Scale Low/High

Allows independent key tracking adjustments of media losses for notes above and below middle C (C3). The green *low* slider (down to C0), inside the Material Pad ring, is visible when the *Keyscale* button is clicked. To adjust, click-hold, and drag the green arrow. The blue *high* slider (up to C6) runs along the left-hand side of the outer ring. Click-hold, and drag the blue arrow, to adjust.

Media Loss Release

The blue slider (in the outer ring of the Material Pad) controls the Media Loss release time. To activate it, you must first click on the *Release* button, to the bottom right of the Material Pad. Values above 1.0 cause media losses to increase when the key is released. This parameter can be used to simulate a string that is dropped into a bucket of water after initially vibrating in air, for example. Obviously, this is not what the average violinist or pianist would do, but it can be useful for a number of interesting sound variations.

Tension Mod (morphable)

Strings, such as those of a guitar, exhibit a particularly prominent non-linear behavior: if the string excursion is large, the string is detuned upwards. As this detuning is caused by the momentary, rather than the average excursion of the string, the detuning occurs very quickly. This phenomenon is known technically as “tension modulation non-linearity”. Non-technically, setting or modulating the *Tension Mod* parameter to values above 0.0 emulates this momentary detuning effect in Sculpture.



Please note that this non-linear effect can produce some surprising results, and can also make the entire model unstable, especially when combined with low *Media Loss* and *Inner Loss* values. So, if you have a sound that “spikes” or “drops out” during the decay phase, try reducing *Tension Mod* (and perhaps *Resolution*).

Tension Mod Scale Low/High

Allows independent adjustments to tension modulation key tracking, for notes above and below middle C (C3). The green “low” slider (down to C0), inside the Material Pad ring, is visible when the *Keyscale* button is clicked. To adjust, click-hold, and drag. The blue “high” slider (up to C6) runs along the left-hand side of the outer ring. Click-hold, and drag, to adjust.

Note: If you find that your instrument seems slightly sharp or flat as you play up/down the keyboard, look at making adjustments to the *Tension Mod* and perhaps *Media Loss* keyscaling parameters.

Excite/Disturb Object Parameters



The following parameters are used to excite, disturb or dampen the String.

Important: At least one *Object must* be used, as the String itself does not make any sound!

As you'll discover shortly, there are a number of different string *excite* "models" such as blow, pluck, bow, and so on. Needless to say, these quite radically alter the general timbre of the String's attack phase, resulting in bowed or plucked "flute" and "bell" sounds, or guitars with a flute-like "blown" sound.

Judicious use of the Object parameters can deliver very accurate emulations of real-world instruments, or sounds that are altogether more "other-worldly".

A particular aspect of component modelling to note is that each additional *disturb/damp* Object that is activated will impact on the String. This will, in turn, alter the interaction of *any* other active Object with the String, often resulting in a completely different character to your sound.

Obviously, changing the sonic character is the reason why you would use a new Object, but the *pluck* and *blow* combination you selected may sound like fingernails on a blackboard—rather than the "plucked pan flute" you were going for—depending on other String settings.

As such, you need to pay special attention to the *Type* and *Strength* of Objects. You may find that the "flavor" of the *excite* Object (1 or 2), for example, has changed significantly—and you may need to adjust or change the parameters of *all* Objects (and perhaps several String parameter values) after introducing a new *disturb/damp* Object (2 or 3). Similarly, the selection of a different *Type* of *excite* Object will impact on the *disturb/damp* Objects (and the string, obviously), and therefore the character of your sound.

The repositioning of Objects changes the timbre of the String. If emulating, say a guitar, changing an Object position could be viewed as similar to picking or bowing a string at various spots along a fretboard.

About Objects and Velocity Sensitivity

It is important to note that:

- *Object 1* is velocity sensitive.
- *Object 2* is *only* velocity sensitive when a *Type* that actively excites the string is selected.
 - When damping Objects are used, Object 2 is *not* velocity sensitive.
- *Object 3* is *not* velocity sensitive. This is because Object 3 can only be used as a disturber of the string and not as an active excitation element.

On/Off (1, 2, and 3 buttons)

Enables/disables the respective Object. When active, each button is lit in a fetching shade of aqua blue.

Type

The following three tables list all types available for Sculptures Objects.

Note: Object 1 can only make use of the excite Types found in the first table. Object 2 can make use of any of the Types available in either table. Object 3 can only make use of the disturb/damp Types found in the second table.

The following table lists all *excite* Types available for *Objects 1* and *2*, and information on the “controls” available for each. Click-hold on the *Type* panel for these objects, and select from the list.

Name	Description	Strength controls	Timbre controls:	Variation controls:
Impulse	a short impulse excitation	impulse amplitude	width	velocity dependency of width
Strike	short excitation like piano hammer or mallet	hammer start speed (velocity dependent)	hammer mass	felt stiffness
Grav Strike	like hammer but with gravitation towards the string, leading to multiple hammer-string interactions and disturbed string vibrations	hammer start speed	felt stiffness	gravitation
Pick	finger or plectrum picking	pickup force and speed	force/speed ratio	plectrum stiffness
Bow	bowing of the string	bow speed	bow pressure	slip stick characteristics
Bow Wide	same as bow, but wider, resulting in a more mellow tone, esp suited for smooth bow position changes	bow speed	bow pressure	slip stick characteristics
Noise	noise injected into the string	noise level	noise bandwidth/cutoff frequency	noise resonance
Blow	blow into one end of the “string” (an air column, or tube). At various positions, starting from 0.0 (far left): move the blowing direction and position from “along the string”, towards one end. The string is “blown sideways” at the chosen position.	lip clearance	blow pressure	noisiness
External (Only avail. for Object 2)	feeds sidechain signal into string.	level	cutoff frequency of lowpass filter being used to process sidechain signal	width (size) of the string area being affected by the sidechained signal

The following chart lists all *Disturb/Damp* Types available for *Objects 2* and *3*. Click-hold on the *Type* panel of these Objects, and select from the list.

Name	Description	Strength controls	Timbre controls:	Variation controls:
Disturb	A disturb object that is placed at a fixed distance from the string's "resting" position	the hardness of the object	the distance from the resting position. Negative values: push the string away from the resting position Positive values: string is unaffected when in the resting position	controls width. Negative values: only a small section of the string is affected Positive values: a broader section of the string is affected
Disturb 2-Sided	This parameter is somewhat akin to a "ring" placed around the string, that limits the string's vibration in all directions	the hardness of the ring	the "clearance" of the ring (the distance between the ring and string). (Negative values: the sides of the damping ring overlap, influencing the string if any movement occurs Positive values: there is an amount of clearance inside the ring. The string will only be influenced if moved sufficiently to actually touch the ring	no impact
Bouncing	Emulates a loose object laying or bouncing on, and interacting with, the vibrating string. This is very random, by nature, and can't be sync-ed.	controls the gravity constant for the object laying/bouncing on the string	the stiffness of the object	the damping of the object
Bound	A boundary that limits and reflects string movement. This is much like a fingerboard that limits string movement when the string is plucked very firmly	the distance from the boundary "center" position to the string's "resting" position	the slope (steepness) of the boundary. A value of 0.0 places the boundary parallel to the string. Other values will position the boundary closer to the string on one end, and further away on the other.	the amount of reflection at the boundary limits
Mass	Used to model an additional mass attached to the string. This can lead to inharmonic sounds, and very interesting results, if the position of this mass is modulated "along" the string	the mass size/weight	no effect	no effect
Damp	Localized damper, which is useful for soft damping	the intensity of the damping	the damping characteristics	the width of the damped string section

Gate

Determines when the Object is active—in other words, when it disturbs/excites the String. Options are:

- *KeyOn*—between note on and note off.
- *Always*—between note on and the end of the release phase.
- *KeyOff*—triggered at note off, and remaining active until the voice is released.



Note: If using an Object *Type* such as *Gravity Strike*, the note may retrigger when you release the key. To avoid this artefact, set *Gate* mode to *Always*.

Strength (morphable)

The central dial adjusts the intensity of the excitation/disturbance, dependent on *Type*. See table above. A value of *0.0* means no excitation/disturbance at all. In contrast to the *On/Off* switch, however, it is possible to fade in the *Strength* (intensity of excitation/disturbance) via modulation and/or morphing.

Timbre (morphable)

The fader to the left of each Object's controls is responsible for the timbre (tonal color) of the excitation/disturbance, which is type dependent. See the table above. *0.0* is the "normal" value for the Object. Positive values make the sound brighter, while negative values lead to a more "mellow" sound.

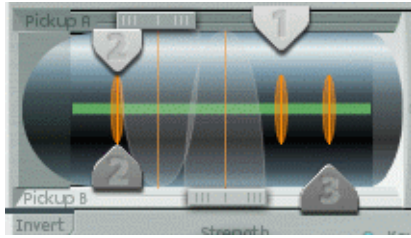
Variation (morphable)

The Variation slider to the right of each Object's controls is an additional timbre parameter, which is again type dependent. Please see the table above for details of its impact on the sound.

VeloSens (Objects 1+2 only)

The excite/disturb Objects are velocity sensitive, but this may not be appropriate for all sounds. This parameter, found at the bottom of Objects 1 and 2, allows you to reduce the velocity sensitivity to zero.

Position (morphable)



Determines the position of each Object along the String (A value of *0.0* means one end, and a value of *1.0*, the other end of the String).

To adjust, simply click-hold and drag the corresponding numerical slider handle (the *1*, *2* or *3* “arrows”) for each Object. Adjustment of these Object pickup positions will disturb/excite a given portion of the string.

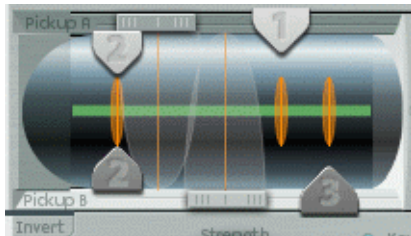
Object 1 can be an exciter. Object 3 can be a damper. You’ll note that Object 2 has two arrows. This indicates that this Object can be used as either an exciter or damper.

- As you move the Object pickups through the Pickup A and B ranges (shown as transparent bell curves), you’ll find that the intensity of the Object disturbance increases significantly. This gives rise to a number of changes which can completely alter the general timbre of your sound.
- The green horizontal line within the Pickup window represents the string. As the *Stiffness* of the string is increased, the line will become thicker.
- You can Control-click on the green horizontal line (the String) to activate/deactivate string animation. When active, this graphical string will vibrate, making it easier to visualize the impact of the Objects and Pickups. Note that string animation increases the CPU overhead, so disable it if your computer is struggling to process all data in realtime.
- The vertical orange lines represent the positions of disturb/excite/damp Objects 1, 2, and 3. The thickness and brightness of these lines indicate the *Strength* of the Object(s).

Processing

The “processing” tools covered in this section act on a per-voice basis, as do the String parameters discussed in the previous chapter.

Pickup Parameters



- The transparent bell curves represent the position and widths of Pickups A and B.
- The green horizontal line within the Pickups window represents the String. As the *Stiffness* of the String is increased, the line will become thicker.
- The vertical orange lines represent the positions of disturb/excite Objects 1, 2, and 3. The thickness and brightness of these lines increases as the *Strength* level of each Object is raised.

Note: View these pickups as being like the electromagnetic pickups found on an electric guitar. Obviously, changing their positions will alter the tone of your axe, and they’ll do the same in Sculpture.

Pickup A Position (morphable)

The *Pickup A* slider at the top of the Pickup window determines the position of Pickup A along the string. Just click-hold, and drag, the slider “cap” to adjust.

Values of 0.0 and 1.0 determine the two ends of the string.

Pickup B Position (morphable)

The *Pickup B* slider at the bottom of the Pickup window determines the position of Pickup B along the string. Just click-hold, and drag, the slider “cap” to adjust.

Values of 0.0 and 1.0 determine the two ends of the string.

Invert (Pickup B Phase)

The *Invert* button is found at the bottom left of the Pickup window. Options are: *normal* or *invert(ed)*.

Note: If the phase of *Pickup B* is inverted, the sound becomes thinner because portions of the *Pickup A* and *Pickup B* signals cancel each other out.

While not found in the actual Pickup window, two further pickup parameters are available ...

Stereo (Key)

Panning position (left/right) is determined by MIDI note number. Dependent on settings, the further up/down the keyboard you play, the more the voice signal is panned left/right.



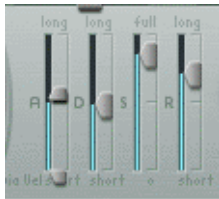
Stereo (Pickup)

Spreads the two pickups across the stereo base. In other words, the pickup position, combined with this parameter, will be spread further towards/from the left/right stereo channels.

Note: You can create animated “width” and “chorus” effects by modulating the Pickup Position parameters with an LFO or other modulator.

Amplitude Envelope Parameters

This is a “classic” analog synthesizer ADSR envelope that scales the *Pickup* signals before passing them on to the *Waveshaper* and *Filter*. This approach/positioning of the Amplitude Envelope at this point in the signal path produces more natural sounding results when the *Waveshaper* is used.



Note: Even with long decay/release times, the sound may decay quickly. This can be caused by high *Inner* or *Media Loss* values (in the String material section) or by Objects (2 or 3) that are used to “damp” the string.

Attack—Soft

The Attack parameter features two sliders. The lower slider determines the Attack time, when the keyboard/incoming note data is played at minimum velocity.

Attack—Hard

The upper Attack slider determines the Attack time when the keyboard/incoming note data is played at maximum velocity.

You can adjust both slider halves simultaneously by dragging in the space between them.

Important: The Attack time parameters of the Amplitude Envelope have a major impact on the way a single note is retriggered. When both *Attack Soft* and *Hard* are set to value of *zero*, the vibrating string is retriggered. If either of these parameters is set to a value *above zero*, a new note will be triggered.

Decay

Defines the Decay time. The Decay time is the amount of time that it takes for the signal to fall to the Sustain level, following the initial strike/Attack time.

Sustain

This parameter sets the Sustain level. The Sustain level is held until the key is released.

Release

This parameter determines the length of time that it takes for the signal to fall from the Sustain level to a level of zero. Short Release values help to reduce CPU load, as the voice is no longer processed once the Release phase has completed.

Waveshaper Parameters

These parameters control and determine any non-linear processing/polyphonic distortion for the stereo signal, resulting from the panning and level scaling of the two Pickup signals.

The *Waveshaper* provides a non-linear shaping curve (per voice) that shapes the signal coming from the *Pickups* (and *Amplitude Envelope*) and passes this reshaped signal on to the *Filter*. This is quite similar to the waveshaping of oscillators in synthesizers, such as Korg's O1/W.



Waveshaper On/Off Button

The *Waveshaper* button enables/disables the Waveshaper.

Type

The *Type* pull-down menu above the Waveshaper button allows you to select different types of waveshaping curves from the list below.

- *Soft Saturation*
- *Vari Drive*
- *Tube-like distortion*
- *Scream*

Input Scale (morphable)

This is a bipolar parameter. Negative values attenuate, and positive values amplify, the input signal prior to processing by the Waveshaper. When set positively, this results in a richer harmonic spectrum. The level increase introduced by the parameter is automatically compensated for by the Waveshaper.

Given its impact on the harmonic spectrum, *Input Scale* should be viewed/used as a timbral control, rather than a level control.

Note: At extreme *Input Scale* values, processing noise can be introduced at the Waveshaper output.

Variation (morphable)

The *Variation* parameter is a bipolar control. Its impact is dependent on the type of Waveshaper selected.

Type	Variation controls:	Value of 0.0	Negative Values:	Positive Values:
VariDrive	wet/dry ratio	provides shaped signal only	reduce shaped signal and add dry signal	raise shaped signal and add phase inverted dry signal, making sound sharper
SoftSat Tube Dist. Scream	Bias—which alters the symmetry of the shaping curve	results in symmetrical shaping	alter symmetry	alter symmetry

Filter Parameters

These parameters offer further timbral/spectral control over your sound. They should be pretty familiar to you if you have any experience with synthesizers.

On/Off

The *Filter* button activates/deactivates the filter.



Filter Type Buttons

The five buttons at the bottom of the filter section determine the filter mode. Choices are:

- *HiPass*—Allows frequencies above the Cutoff Frequency to “pass”. As frequencies below the Cutoff Frequency are suppressed, it’s also known as a Low Cut Filter. The slope of the filter is 12 dB/octave in *Highpass* mode.
- *Lowpass*—Allows frequencies which fall below the Cutoff Frequency to “pass”. As frequencies above the Cutoff Frequency are suppressed, it’s also known as a High Cut Filter. The slope of the filter is 12 dB/octave in *Lowpass* mode.
- *Peak*—This mode allows for an increase in level of a frequency band, the width of which is controlled by the *Resonance* parameter.
- *Bandpass*—In this mode, only the frequency band directly surrounding the Cutoff Frequency can pass. All other frequencies are cut. The *Resonance* parameter controls the width of the frequency band that can pass. The band pass filter is a two-pole filter with a slope of 6 dB/octave on each side of the band.
- *Notch*—In this mode, the frequency band directly surrounding the Cutoff Frequency is cut. All other frequencies are allowed to pass. The *Resonance* parameter controls the width of the frequency band that is cut.

Cutoff (morphable)

Determines the filter cutoff frequency. As an example of its use: In a *low pass* filter, all frequency portions *above* the *Cutoff Frequency* are suppressed, or “cut off”; hence the name. The *Cutoff Frequency* controls the brilliance of the signal. The higher the Cutoff Frequency is set, the higher the frequencies of signals that are allowed to “pass” through the low pass filter.



Resonance (morphable)

Sets the filter resonance value.

For *highpass* and *lowpass* modes, the *Resonance* parameter emphasizes the portions of the signal which surround the frequency—as defined by the *Cutoff* value.

In *Peak*, *Notch*, and *Bandpass* modes, *Resonance* controls the bandwidth.

Key

This knob adjusts the key tracking of the *Cutoff Frequency*. Put plainly, the further up/down the keyboard you play, the more bright/mellow the sound becomes. Put more technically, the Cutoff Frequency is modulated by keyboard position.

- A value of *0.0* disables key tracking.
- A value of *1.0* allows the Cutoff Frequency to proportionately follow the fundamental of the note across the entire keyboard range.

Velo Sens

Determines the velocity sensitivity of the *Cutoff Frequency*. The harder you strike the keyboard you're playing, the higher the cutoff frequency (and generally, the brightness of the sound) becomes.

- A value of *0.0* disables velocity sensitivity.
- A value of *1.0* results in maximum velocity sensitivity.

Post Processing

The “post processing” tools covered in this section impact on the summed stereo signal of *all* voices, rather than on a per-voice basis.

Stereo Delay

This is a (song) tempo-syncable stereo delay. It may also be set to run freely (not synchronized).

On/Off

The *Stereo Delay* button enables/disables the Delay section.

Wet Level

The *Wet Level* knob sets the level of the Delay output (wet signal). The parameter value is expressed as a percentage (%). Dry level is 100%.



Feedback

Defines the amount of delay signal that is fed from the left out (of the delay unit) to the left in (of the delay unit), and from right out to the right in. Negative values result in phase-inverted feedback.

Crossfeed

As above, but this parameter defines the amount of delay signal that is fed from the left out to right in and right out to left in (of the delay unit). Negative values result in phase-inverted feedback of the crossfed signal.

LoCut

Determines the cutoff frequency of the highpass filter at the delay line output/feedback loop.

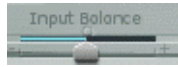


HiCut

Determines the cutoff frequency of the lowpass filter at the delay line output/feedback loop.

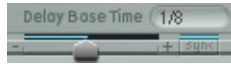
Input Balance

This parameter provides true stereo panning. Adjustments allow you to move the (stereo) center of the Delay input to the left or right, without the loss of any signal components. This makes it ideal for “ping-pong” delays.



Delay Base Time

This parameter, coupled with the *Sync* setting, enables you to set the delay base time. This can be in either: musical note values— $1/4$, $1/4t$ (t = triplet) and so on—or in *milliseconds*.

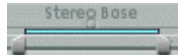


Sync

The Sync button allows you to select either tempo-synced, or tempo-independent Delay modes.

Stereo Base

This parameter allows a reduction of the stereo base of the wet signal.



- A value of *0.0* = mono output
- A value of *1.0* results in full stereo output (the left delay line output is panned hard left, and the right delay line output is panned hard right).

This parameter is used to achieve “pure” delay grooves, without hard left/right “ping-pong” panning.

The following two parameters are combined in the two-dimensional *Delay Time Pad*. Drag the diamond in the center of the crosshair to adjust.

Note: You can independently adjust the *Spread* and *Groove* parameter values by directly dragging the lines that intersect the “diamond”.

Spread

Positive values on the “y” axis (above the default, centered position) increase the delay time of the right delay line and decrease the delay time of the left delay line—in effect, “smearing” the delay times of the left and right channels. Negative values reverse this. *Spread* is useful for “wide” stereo delay effects.



Groove

This parameter (on the “x” axis) allows you to reduce the delay time of one delay line by a given percentage, while keeping the other delay line constant. Basically, *Groove* distributes the delay taps to the left/right channels, rather than “smearing” them, like *Spread*. Keep an eye on the small Help Tag while adjusting.

As an example, a value of +50% reduces the right delay time by half. If a value of 1/4 was used as the *Delay Base Time*, the right delay would equal 1/8th of a note and the left delay would remain at 1/4 of a note. Needless to say, this parameter is perfect for the creation of interesting rhythmic delays—in stereo.

Note: You can create some truly wide “chorus” and “modulated delay” effects by modulating the Pickup Position and Pickup Stereo parameters (with an LFO or other modulator), and then feeding this into the Delay unit.

Body EQ/Spectral Shaping

Model-based spectral shaping. It can work as a simple EQ, as a complex spectral shaper or as body response simulator. In effect, the Body EQ can emulate the resonant characteristics of a wooden or metallic “body”—such as that of a guitar or violin.

The various models are derived from Impulse Response recordings of instrument bodies. These recordings have been separated into their general formant structure and fine structure, allowing you to alter these properties separately.

Body EQ On/Off Button

This button (to the left of the image) enables/disables the spectral shaping section.



Model

The *Model* pull-down menu allows you to select the model from the following list. Any selection will be reflected in the graphic window to the right:

- *Low Mid Hi*—A broadband equalizer module, with individual controls (via the knobs) of Low, Mid, and High frequency ranges.
- *Guitar 1*—Emulates the body of an acoustic guitar.
- *Guitar 2*—Emulates the body of another acoustic guitar.
- *Violin 1*—Emulates the body of a violin.
- *Violin 2*—Emulates the body of another violin.

Body EQ Controls

Please note that the following parameters are labeled differently for the Basic EQ (*Low Mid Hi* Model). All other model parameters are discussed below.

- *Low*—gain of a low shelving filter—set at 80 Hz
- *Mid*—gain of a peak filter—set at 2.5 kHz (*but is sweepable—see below*)
- *High*—gain of a high shelving filter—set at 12 kHz

In addition, when the *Lo Mid Hi* model is selected, the slider above the response graph is labeled *Mid Frequency*, and allows you to sweep the center frequency of the mid band between 100 Hz and 10 kHz.

You can directly control the *Lo Mid Hi* model by click-dragging in the Body EQ graph:

- Click-dragging vertically on the left third of the graph allows you to control the *Low* parameter.
- Click-dragging vertically on the center third of the graph allows you to control the *Mid* parameter.
- Click-dragging horizontally on the center third of the graph allows you to control the *Mid Frequency* parameter.
- Click-dragging vertically on the right third of the graph allows you to control the *Hi* parameter.

For all other Models:

For all other BodyEQ models you have the following parameters:

Formant—Intensity

Scales the intensity of the model's formants. In other words, any formants (harmonics) in the model will become louder, or will be inverted, dependent on how this parameter is used.

- A value of 0.0 results in a flat response.
- A value of 1.0 results in strong formants.
- Negative values invert the formants.



Formant—Shift

This parameter shifts the formants logarithmically. A value of -0.3 , for example, shifts all formants one octave downwards, and a value of $+0.3$ shifts the formants up one octave. A value of $+1.0$ shifts up by a factor of 10—from 500 Hz to 5000 Hz, for example.

Formant—Stretch

Stretches the formant frequencies, relative to each other. In other words, this parameter alters the width of all bands being processed by the *Body EQ*, extending or narrowing the frequency range.

Low *Formant Stretch* values move the formants closer together (centered around 1 kHz) while high values move the formants further apart from each other. The control range is expressed as a ratio of the overall bandwidth.

When combined, *Formant Stretch* and *Formant Shift* alter the formant structure of the sound, and can result in some interesting timbral changes.

Fine Structure

This parameter enhances the spectral (harmonic) fine structure, making the overall harmonic makeup of the sound more “precise.” This results in a more detailed sound that is harmonically richer and—dependent on the model selected—more guitar or violin-like, for example. Put another way, the resonant cavities of the instrument become more resonant—somewhat like the increased “depth” of tone provided by a large-bodied guitar.



- A value of 0.0 denotes no fine structure.
- A value of 1.0 results in enhanced/full fine structure of the selected model.

Please note that heavy use of *Fine Structure* may be quite CPU intensive.

You should also note that the use of *Fine Structure* may not actually result in too much of a difference to your sound. This is dependent on the *Waveshaper* and *Body EQ* modes, plus other *String* parameter settings. As always, use your ears!

Adjusting Model

You can directly control the model by click-dragging in the *Body EQ* graph:

- Click-dragging vertically on the graph allows you to control the *Formant Intensity* parameter.

- Click-dragging horizontally on the graph allows you to control the *Formant Shift* parameter.

Level/Limiter

Level

Controls the overall output level for the instrument. Drag to adjust.



Level Limiter Mode

Clicking on the desired button activates/deactivates the integrated limiter. Options are:

- *off*—disables the limiter.
- *mono*—a monophonic limiter on the summed signal of all voices.
- *poly*—a polyphonic limiter, that processes each voice independently.
- *both*—a combination of both limiter types.

This function is very useful for “domesticating” some of the more “aggressive” aspects of Sculpture’s component modelling synthesis engine.

Modulation Generators

Sculpture features a comprehensive collection of modulation generators. There are:

- Two assignable *LFOs*, with (song) tempo-syncable rates.
- An additional LFO that is hard wired to *Vibrato*.
- Random variations can be created via two *Jitter* generators (with adjustable bandwidth).
- Two *Randomizers* that only change values at note start/on.
- Two *Control Envelopes* can either be used as “standard” envelopes or as MIDI controlled modulators—with the ability to record, polyphonically play back (on a per-voice basis) and modify incoming MIDI controller movements.

All modulation assignments take place within the generators.

“*First order*” modulations are where the modulation generator modulates core synthesis parameters.

“*Second order*” modulations are where one modulation generator modulates the parameters of the other modulation generator. These include: *LFO Rate Modulation*, *VariMod*, *Morph Envelope Modulation*, and *A Time Velosens*.

Each modulation generator allows you to select one (or in most cases, two) of the core synthesis parameters as a modulation “*target*.” The same target can be selected for all modulators, if desired.

Modules such as the LFOs and Control Envelopes also offer “*via*” modulations (a freely modulatable amount/intensity—the scaling factor—of the modulation source output level). These are “*sidechain*” modulations.

Some modulation generator parameters—for example LFO rates, can be modulated by selecting the desired modulation “*source*” and amount.

To access the desired modulation generator, click on the corresponding button in this section of the Sculpture GUI. Once a modulation source has been activated, the corresponding button label will be lit.



LFO 1 and 2

The two full-featured LFOs 1+2 offer several possibilities that go beyond the Vibrato LFO and Jitter generators described below.



Waveform

This pull-down menu allows you to select the waveform used for LFO modulation. Options are: *Sine*, *Triangle*, *Sawtooth*, *Rectangle Unipolar*, *Rectangle Bipolar*, *Sample&Hold*, *Sample&Hold with Lag*, *Filtered Noise*.

The triangular wave is well suited for vibrato effects.

The sawtooth is well suited for helicopter and “space gun” sounds. Intense modulations of incoming frequencies leads to “bubbling and boiling, underwater” sounds. Intense sawtooth modulations of lowpass filters create rhythmic effects.

The rectangular waves make the LFO periodically switch between two values (e. g.—a positive value and zero—Unipolar). The Bipolar Rectangular wave switches between a positive and a negative value set to the same amount above/below zero.

The two Sample&Hold waveform settings output *random* values. A random value is selected at regular intervals, defined by the LFO rate. A modulation of *Pitch* leads to the effect commonly referred to as a “random pitch pattern generator” or “sample and hold”. Check out very high notes, at very high rates and high intensities—you’ll recognize this well-known effect from hundreds of science fiction movies! If the Sample&Hold with Lag setting is used, the random wave is delayed, resulting in a “fluid” changing of values.

The term “*Sample & Hold*” (abbreviation—*S & H*) refers to the procedure of taking “samples” from a noise signal at regular intervals. The voltage values of these “samples” are then “held” until the next *sample* is taken. When converting analog audio signals into digital signals, a similar procedure takes place: Samples of the voltage of the analog audio signal are taken at the rate of the sampling frequency.

Rate

The *Rate* knob determines the rate of LFO modulation, which can be either synced to the current song tempo or set independently in Hz (Hertz) values.

- In “Hz” mode, the rates range from DC (Direct Current) to 100 Hz.
- In “sync” mode, rates range from a periodic duration of 32 bars through to speeds of 1/64-triplets. Triolic and punctuated values are also possible.

The LFO’s are ideally suited for rhythmic effects which retain perfect synchronicity even during tempo changes to the song.

Sync/Free Buttons

Part of the *Rate* parameter, these buttons allow you to select either synchronized or free-running LFO rates. When toggling between modes the value is derived from the song tempo and meter.

Curve

Allows you to define a freely-variable number of waveform variations, resulting in subtle/drastic changes to your modulation waveforms.

The Curve parameter can even influence the Sine waveform type.

- Curve value of 0.0—pure sine wave
- Curve values above 0.0—wave is smoothly changed into a nearly rectangular wave.
- Curve values below 0.0—the slope at the zero crossing is reduced, resulting in shorter soft pulses to +1 and –1.

Note: The waveform displayed between the *Curve* knob and the *Waveform* pull-down menu shows the results of these two parameter settings.

Envelope

A simple LFO envelope control that allows either;

- constant modulation
- a ramping up to the full amount

- a decay to a zero amplitude level.

Phase



Allows the choice between strictly monophonic or polyphonic LFO modulations with either; similar phases, completely random phase relationships, key-synced phase ... or anything in-between.

- If used polyphonically for modulation of multiple voices, the modulations will *not* be phase-locked. To explain, when used on polyphonic input (a chord played on the keyboard) the modulation is independent for each voice (note). Where the pitch (for example) of one voice may rise, the pitch of another voice might fall and the pitch of a third voice may reach its minimum value.
- When used monophonically, the pitch of all voices will rise and fall synchronously.
- If used randomly, some notes will be modulated synchronously, and others won't.

Note: If you move the *Phase* knob slightly away from the *mono* position, you'll get non-locked modulations for all voices running at similar, but not identical, phases. This is ideal for string-section vibratos!

RateMod Source and Amount

The LFO rate can be modulated by the *Source* and amount that you determine with this pull-down menu and vertical slider (found to the right of the abovementioned LFO controls).



Target1/2

These two modulation destinations can be assigned per LFO, with an optional, additional “via” modulation. These parameters determine modulation destinations 1 and 2. To activate, click on either the 1 or 2 buttons (which will highlight the Target and via text/pull-down menus), then click-hold on the *Target* pull-down menu(s). Select the desired target, and release the mouse button.



Via (source) 1/2

These parameters determine the source that controls the modulation scaling for each LFO.

Modulation amount sliders: Amt and Via (Amount)

In cases where the *Via* source is set to *off*, only one amount slider is visible (the *Via (Amount)* slider is hidden):

- Amt 1/2—The *Amt* slider determines the modulation amount.

In cases where any *Via* source other than *off* is selected, there are two sliders:

- Amt 1/2—The *Amt* slider determines the modulation amount in cases where the incoming *Via* signal is *zero*, e. g. a modulation wheel at its minimum position/value.
- Via (Amount) 1/2—The *Via (Amount)* slider determines the modulation amount in cases where the incoming *Via* signal is at *full level*, e. g. a modulation wheel at its maximum position/value.

Vibrato

One LFO is hard-wired to pitch, for vibrato effects. The strength of the vibrato effect can be adjusted via the MIDI controller assigned in the *VibDepth Ctrl* menu. This is set in the MIDI controller assignment section. More information can be found in the “MIDI Controller Assignments” on page 408.

Waveform

Allows you to select the waveform used for vibrato, for example *sine*, *triangle*, *sawtooth*, and so on.

There are two “special” rectangular waves: *rect01* and *rect1*—the former switching between values of *0.0* and *1.0* (unipolar), and the latter between values of *−1.0* and *+1.0* (bipolar, like the other waveforms). See the “LFO 1 and 2” section, on page 385.



Curve

Allows you to define a freely-variable number of waveform variations, resulting in subtle/drastic changes to your modulation waveforms.

The Curve parameter can even influence the Sine waveform type.

- Curve value of 0.0: pure sine wave
- Curve values above 0.0: wave is smoothly changed into a nearly rectangular wave.
- Curve values below 0.0: the slope at the zero crossing is reduced, resulting in shorter soft pulses to +1 and −1.

Note: The waveform displayed between the *Curve* knob and the *Waveform* pull-down menu shows the results of these two parameter settings.

Phase

Allows the choice between strictly monophonic or polyphonic vibrato with either; similar phases, completely random phase relationships, key-synced phase—or any value in-between. See the “LFO 1 and 2” section, on page 385, for more details.

Rate

Determines the rate of vibrato, which can be either synced to the current song tempo or set independently in Hz (Hertz) values. See the “LFO 1 and 2” section, on page 385, for more details.

Depth via Vib Ctrl—Max

This slider determines the maximum possible modulation amount.

Depth via Vib Ctrl—Min

This slider determines the minimum possible modulation amount.

Random Variations

Jitter 1+2

The two jitter generators are special LFOs, designed to produce continuous, random variations—such as those of smooth bow position changes. The jitter generators are equivalent to general purpose LFOs set to a “noise” waveform.

To activate the routing of the Jitter generators, click on the *1* or *2* buttons.



Note: Jitter modulation of pickup positions as the Target produces great chorus-like effects.

Rate/Bandwidth

These knobs set the bandwidth/speed of the modulation (jitter) signal for each respective Jitter generator.

Target 1/2

These pull-down menus define modulation destinations 1 and 2.

Amount 1/2

These sliders determine the amount of modulation for each Jitter generator.

Note On Random

The two note on random generators are intended for random variations between different notes/voices. Their values are randomized for each note, and remain constant until the voice is released. Such randomizations are useful for adding interest/thickening the sound when played polyphonically. It is also useful for emulating the random variations a player introduces when playing an instrument—even when repeating the same note.

To activate the routing of the Note On Random modulators, click on the *1* or *2* buttons.

Target

Determines the modulation destination—what parameter will be randomly modulated when a note is played.

Amount

Sets the modulation amount—the depth/strength of the modulation.

Velocity Modulations

The *Excite Objects* and the *Filter* have dedicated velocity sensitivity controls. Many other modulation routings also allow you to select *Velocity* as a *via* input source.

In some cases, it may be of use/interest to directly control other synthesis core parameters by velocity. This can be done in this section—where two independent destination/amount/velocity curve slots are available. To activate either, click on the *1* or *2* buttons.



Target 1/2

Click-hold on the respective *Target* pull-down menu to select the destination parameter that you wish to modulate by velocity.

Amount 1/2

The slider below each *Target* determines the amount of modulation.

Curve 1/2

Click on the appropriate radio button to select from *concave*, *linear*, and *convex* velocity curves.

Controller A and B



These parameters allow you to define two discrete modulation targets, and the strength/intensity of modulation for both Controller A and B.

To use, simply click on the *1* or *2* buttons, select the desired *Target* (and *Target mode*), and adjust the *Intensity* slider.

Each *Target* features a two-state button:

- *Cont*: continuous modulation
- *Note On*: modulation value is only updated when a note on message is received.

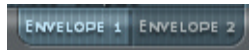
The Control Envelopes



The two Control Envelopes are somewhat special, as they can be used as:

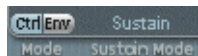
- “traditional” four segment envelopes.
- MIDI controller modulations.
- a combination of both: as MIDI controller movement *recorders* (with ADSR-like macro parameters), for polyphonic playback.

To select Envelope 1 or 2, click on one of these buttons.



Mode—Ctrl/Env Buttons

Selects either *controller* (“run” mode) or *envelope* functionality. If both are activated, the controller value is *added* to the envelope output, resulting in a modulation offset.



The Envelopes can act as polyphonic modulation recorders and playback units. Each voice is handled independently, with a separate Envelope being triggered/running as each note is played.

Modulation Routing

As in the LFO section, each envelope offers two modulation *Target* selectors with *amount* and *via amount* controls, and a separate *via* modulation option.

The following *target/amount/via* settings are available for all “run” modes.

Target 1 and 2

Two modulation targets can be assigned per Envelope, with an optional, additional *via* modulation. These parameters determine modulation destinations 1 and 2.

Either the 1 or 2 buttons must be clicked in order to activate these modulation options.

A large number of possible Targets are available, including; *string*, *object*, *pickup*, *waveshaper*, and *filter* parameters. To select, simply click-hold on the panel below the word Target, and make your choice from the list.



Via (source) 1/2

The *Via* parameter defines the modulation amount for Envelopes 1 and 2. The *Via* panels allow the selection of sources that are used to scale the modulation amount of the envelopes.

Modulation amount sliders: Amt and Via (Amount)

In cases where the *Via* source is set to *off*, only one amount slider is visible (the *Via (Amount)* slider is hidden):

Amt 1/2

The *Amt* slider determines the modulation amount.

In cases where any *Via* source other than *off* is selected, there are two sliders:

Amt 1/2

The *Amt* slider determines the modulation amount in cases where the incoming *Via* signal is *zero*, e. g. a modulation wheel at its minimum position/value.

Via (Amount) 1/2

The *Via (Amount)* slider determines the modulation amount in cases where the incoming *Via* signal is at *full level*, e. g. a modulation wheel at its maximum position/value.

Envelope Curve Window

The Envelope curve is displayed in the window to the lower right of the Sculpture GUI.

The envelope curve window is only active if the envelope functionality is engaged (*Mode* set to either *Env* or *Ctrl+Env*).



- The overall time/length of the Envelope is indicated by the numerical entry at the top right of the window (1790 ms in the graphic).
- The maximum time/length of the Envelope is 48 bars/40 seconds.

- The lines on the background grid are placed 100 milliseconds apart.
- The background lines are placed 1000 ms apart for very long displayed envelope times. In sync mode, this is displayed as 1 quarter.
- The envelope is *zoomed automatically* after releasing the mouse button. This allows the display of the entire envelope at the highest possible resolution for the graphic Envelope window.
- This behavior can be disabled/enabled by pressing the Autozoom button—the small magnifying glass.
- Autozoom is automatically disabled when you perform a manual zoom—by click-holding on the Envelope Curve window background, and dragging horizontally. As a reference, the current window width is displayed by the numerical entry at the top right of the window. You can re-engage autozoom by clicking on the Autozoom button.
- If you click on the handles (nodes) or lines between the nodes in the Envelope window, the current envelope segment will be highlighted. A small Help Tag also indicates the millisecond value of the current segment.

Envelope Handling

When first launched, a default Envelope “curve” is automatically created for each Envelope. To view either, press the *Env* button in the *Mode* section.

If a recorded Envelope exists, the parameters discussed in this section will allow you to fine tune positions and levels.

You will see a few handles (nodes) placed—from left to right—along a straight line within the Envelope. These are indicators of the following parameters.

- Node 1—Start level.
- Node 2—Attack time position/level.
- Node 3—Loop time position/level (can be freely positioned).
- Node 4—Sustain time position/level (can be freely positioned).
- Node 5—End time position/level.

As you move your mouse cursor along the line, or hover over the nodes directly, the current Envelope segment is highlighted.

You can create your own envelopes manually, by manipulating the nodes and lines, or you may record an envelope, as discussed in the “To record an Envelope...” section, on page 396.

To adjust the *time* between nodes, click on the desired handle, and drag it left or right. As you do so, the overall length of the Envelope will change—with all following nodes being moved.

You cannot move a node beyond the position of the preceding node. You can, however, move nodes beyond the position of the following node—even beyond the right-hand side of the Envelope window—effectively lengthening both the envelope segment and the overall envelope.

When you release the mouse button, the Envelope window will automatically zoom to show the entire envelope.

To adjust the *level* of each node, click on the desired handle, and drag it up or down.

To adjust the *shape of the curve* between nodes, click directly on the line that connects them, and drag it up or down.

In the case of recorded envelopes, you may have a more complex “curve” between nodes. To adjust, Control-click and drag the curve. An example of this is shown in the next section.

Experiment with each node/parameter to get a feel for its operation. You’ll find that the Envelopes are very intuitive to use.

Envelope Recording

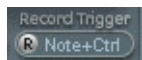
Before we begin...

It is important to note that you can only record MIDI controller movements of the assigned MIDI controller. MIDI controller assignments for the Envelopes must be set in this section of the Sculpture GUI:



R(ecord) Button

Enables the record functionality for the Envelope. This button works in a similar fashion to the record arm buttons in Logic. To stop recording, simply click on the *R* button a second time, or use the “trigger” mode functionality described below.



Record Trigger Mode

The pull-down menu to the right of the *R* button is used to select different record trigger modes to start recording (when *R(ecord)* is active):

- *NoteOn*: recording starts when a note is played
- *Note+Ctrl*: recording starts when MIDI control change messages (for the assigned controllers—see “MIDI Controller Assignments” on page 408) arrive while a note is held.
- *Note+Sus*: recording starts when the sustain pedal is depressed while a note is held.

To record an Envelope...

To provide you with an Envelope recording example:

- Set the *Record Trigger Mode* to *Note+Ctrl*.
- Enable record by clicking on the *R* button.
- Play, and hold, a key—and start moving the modwheel or whatever controllers are assigned to Envelope controls 1 and/or 2.

To end an Envelope recording...

An Envelope recording ends as soon as at least one of the following conditions is met:

- The record button (*R*) is disengaged manually by clicking on it.
- All *voices* are released.
- A new note is played after releasing all *keys*.

Following the recording of a controller movement, *R(ecord)* is automatically set to *off* and *Mode* is set to *Env*. This ensures that only the recorded movement will be active, regardless of the “stop” position of the recorded controller.

To play back a recorded Envelope...

Polyphonic playback of the recorded envelope occurs when you play a key. The *Mode* parameter must be set to *Env* and the *R(ecord)* parameter must be set to *off*.

You can also activate both the *Env* and *Ctrl* buttons of the *Mode* parameter, as this will allow you to use controllers assigned to *Ctrl Env1* or *Ctrl Env2* to manipulate the Envelope in realtime, alongside playback of the recorded Envelope.

Note that if both *Env* and *Ctrl* are activated, however, the controller value is *added* to the envelope output, resulting in a modulation offset.

Preparing the recorded envelope for editing

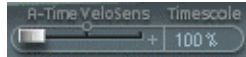
The envelope segments and handles are set automatically after recording. To change the interpretation of the envelope you can grab and drag the vertical lines that intersect the handles (also see the VariMod paragraph below). Note that this will not change the shape of the envelope.

Envelope Parameters

The following parameters are only active if the envelope functionality is engaged (*Mode* set to either *Env* or *Ctrl+Env*).

A-Time Velosens

This slider is used to set the velocity sensitivity for the Attack time of the envelope. Positive values will make the Attack time shorter at minimum velocities, and negative values will make the Attack time shorter at maximum velocities.



Timescale

This parameter allows you to scale the duration of the entire envelope between 10% (ten times faster) and 1000% (ten times slower). This will also visually impact on the appearance of the Envelope curve displayed as it is shortened (sped up) or lengthened (slowed down).

Sustain Mode

Allows you to select the behavior of the Envelope while a note is held. Choices are: the usual *sustain* mode, *finish* mode or one of three loop modes (*forward*, *backward*, *alternate*). When in any of the loop modes, the loop always cycles between user-defined Envelope “handles” (simply drag them to the desired position) that indicate the *Loop start* point, and the *Sustain* point.

The Envelope can—like any envelope—run in one shot (as long as the note is sustained). It can also run several times or in an infinite cycle, much like an LFO. You can achieve this through the use of *loops*.

- When set to *Finish*, the Envelope runs in “one shot mode” from its beginning to its end—even if the note is released before the envelope has come to its end. The other loop parameters are disabled.
- When set to *Loop Forward*, the Envelope runs to the sustain point and begins to repeat the section between the loop point and sustain point periodically—always in a forward direction.
- When set to *Loop Backward*, the Envelope runs to the sustain point and begins to repeat the section between the sustain point and loop point periodically—always in a backward direction.
- When set to *Loop Alternate*, the Envelope runs to the sustain point and returns to the loop point and back to the sustain point periodically, alternating in both a backward and forward direction.

Note: If the loop point lies behind the sustain point, the loop will start after the key has been released.

Loops can be synchronized to the song tempo automatically via the *sync/ms* buttons.

Compare

Following the initial (original) recording of an envelope, and any subsequent edits, this button allows you to toggle between the original recording and the edited version. Obviously, this is only available as an option if an envelope curve has actually been recorded.



Sync/ms Buttons

These parameters allow you to select between a free-running envelope (with segment times displayed in milliseconds) or a tempo-synced envelope with note value options, such as *1/8th* or *1/4*.

Switching between values forces a recalculation of times to the nearest note value or ms time, respectively, based on the current song tempo.

VariMod—Source and Amount

VariMod is only available for recorded envelopes.

“Variation” in the Envelopes means the deviation of a recorded envelope path from “straight” interconnecting lines between the points. After having recorded an envelope you can reduce or exaggerate the amplitude-jitter of the recording by Control-dragging the curves down (to reduce) or up (to exaggerate).

The variation parameter can be modulated by using these source and amount parameters. To adjust the amount/level of variation, simply move the *VariMod* slider up/down. Source options can be accessed by click-holding on the *Source* button, and making your selection from the pull-down menu. Choices include: *Off*, *Velocity Concave*, *Velocity*, *Velocity Convex*, *KeyScale*, *Ctrl A*, and *Ctrl B*.



VariMod allows you to select a modulation source, and amount, to control the strength of the “deviation”. As an example; if you recorded a tremolo, as shown.



You could trim the loop by dragging the envelope “handles” to fit the Attack peak, and the start and endpoints of a tremolo loop.

- To do so, grab and drag the vertical lines that intersect the handles. (These are also highlighted when the mouse pointer touches them). This will result in something like this:



- If you now reduce the *variation* depth by Control-dragging down the curve of the loop segment, you’ll end up with the following:



- Now, set up the *VariMod* source to *CtrlA* (which might be set to *Touch*, for example), and an amount above 0. You'll end up with an interesting tremolo (or whatever you're modulating with this envelope) with an arbitrary waveform, and a modulation depth that can be controlled by Key Pressure (*Touch*) or whatever controller is assigned to *CtrlA*.



Copying Envelopes

If you Control-click on the Envelopes, a context menu opens, offering Copy, Paste, and Clear options. These allow you to copy and paste (or clear) envelopes between Envelopes 1 and 2, between settings and/or multiple open Sculpture instances.

Morph

Sculpture includes “morph” functionality that allows you to smoothly change the sound—in a subtle or radical way—between up to five snapshots, called morph points.

Each morph point includes separate settings for more than 20 important synthesis core parameters, including; string material parameters, parameters and positions of the excite/disturb objects, pickup positions, filter, and waveshaper parameters.

All morphable parameters can be automated independently for each morph point.

The current morph point position within the *Morph Pad* can be controlled via MIDI controllers (for example by a vector stick). Such movements can be recorded and played back independently—each voice can be morphed differently.

The morph section consists of two parts:

- The *Morph Pad*, featuring 5 morph points (center and 4 corners), plus options for randomizing (via the *Randomize* parameters) and copy/pasting morph points and Morph Pad “states” via a contextual menu.

- The *Morph Envelope*, that can be edited either by segment (with the mouse), or recorded via MIDI controller movements. With a vector stick (*Morph X/Y* controllers) or mouse movements of the *Morph Cursor* (the “ball”), for example, on the Morph Pad.



Morph Point Selection and Randomization



There is always *one* of the five Morph Pad points (A/B/C/D/Center) that is selected for editing. This selected point is indicated by two concentric circles that surround it.

All morphable parameters (all parameters which have an orange value bar, rather than a blue or turquoise one) are shown, allowing you to edit the values of the *selected* Morph Pad point.

Auto Select

Movements of the morph cursor automatically select the nearest morph point when this parameter is active.



You can also click into the circles around A, B, C, D, or Option to select a Morph Pad point manually.

Randomizing

The randomize feature allows you to create random variations of selected morph points. When combined with the copy/paste functionality that's also available, randomizing lends itself to using the Morph Pad as a kind of sound cell culture device.

Use of the Morph Pad can yield an interesting/inspiring composite sound. You can copy this sound to a corner of the Morph Pad (or several corners) and randomize it by a definable amount.

The morphed sound then becomes a new timbral element, creating permutations that can, in turn, be moved to the corners, randomized, and so on ...

In effect what you are doing is "breeding" a sound, while maintaining some control by selecting parent and child sounds.

This approach can result in new, complex sounds without the need to be a programming guru.

Points

Selects the number of morph points that are to be used for randomization, and indicates which points will be randomized.



Randomize Button

The *Rnd* button generates a randomized collection of Morph points.



Int(ensity)

The *Int(ensity)* slider determines the grade of randomization from 1% (slight deviation) to 100% (completely random values).

A Brief Randomizing Tutorial

- Select the top button (5 points) in the Points section. Ensure that *Auto Select* is active.
- Select the *Int(ensity)* slider, and drag it to a value of say 25%.
- Now press the *Rnd* button. As you're doing so, take a look at the parameters in the core synthesis engine. You will see a number of them move after the mouse click.
- Now, click-hold on the Morph Cursor (the "ball"), and drag it to each of the corners in the Morph Pad. Do this along the edges, as well as through the center of the Morph Pad, and take a mental note of how this impacts on the "morph".
- Oh, and don't forget to strike a few notes on your MIDI keyboard while doing so.

As you're moving the morph cursor around, you'll see the realtime movement of the "ghost" controls in the Pickup window, the "ghost" ball in the Material Pad, and—if you look closely—you'll also notice a number of red dots moving about in the various string and object parameters.

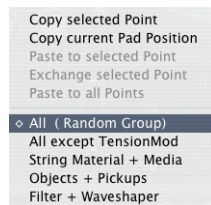
Each of these indicates the "current" Morph position. This is handy tool for seeing what parameters have changed, and how they have been changed.

You'll also note that positions on the Morph Pad that fall in-between the various morph points cause the randomized parameters to interpolate between values—which is what "morphing" is all about.

Check out the next section on the copy/paste options, as these will help you to make use of those "in-between" settings.

Morph Menu Parameters

You can access this contextual menu by Control-clicking on the Morph Pad.



Copy/Paste

These parameters allow you to copy:

- the current morph point (*Copy selected Point*) or the
- current morph state (*Copy current Pad Position*) into a local clipboard, and
- *Paste to a selected Point*
- *Exchange a selected Point* (basically, swaps previously copied data with another point of your choice) or
- *Paste to all Points*

The remaining entries in the Morph Pad contextual menu are to do with grouping of random parameters. Put another way, the following options allow you to determine which type of parameters you would like to randomize (via the *Rnd* button and *Int(ensity)* slider).

- *All (Random Group)*—this is your ticket to “wacky” sounds as all parameters in the following three groups are randomized. This can lead to some interesting results, but can be uncontrolled. This is less useful for the “sound cell culture” idea that we discussed in the Randomizing section above.
- *All except TensionMod*—basically the same as *All (Random Group)* but excluding the somewhat dangerous *TensionMod* parameter from randomization. This option is the default “random” group.
- *String* (parameters)—in isolation, including; Material Pad position, stiffness, inner loss, media loss, resolution, and Tension Modulation.
- *Objects + Pickups*—in isolation. Alters the positions of Objects and Pickups, plus the various Object parameters.
- *Filter + Waveshaper*—Randomizes the positions of all Waveshaper and Filter parameters.

Morph Envelope Window

The Morph Envelope offers 9 points/8 segments, and recording functionality that is much like that of the *Controller Envelopes*.

Selected Point

In the diagram below, you will see a selected point in the lower panel (the *Timeline*) and a corresponding selected point handle of the trajectory in the *Morph Pad*.

These are shown as an orange diamond in the *Timeline* and an orange diamond (or ball) in the *Morph Pad*.



- The overall time/length of the Morph Envelope is indicated by the numerical entry at the top right of the window.
- The maximum time/length of the Morph Envelope is 48 bars/40 seconds.
- The lines on the background grid are placed 100 milliseconds apart.
- If you click on the handles (nodes) or lines between the nodes in the Envelope window, the current envelope segment will be highlighted. A small Help Tag also indicates the millisecond value of the current segment.

- As you move your mouse cursor along the line, or hover over the nodes directly, the current Envelope segment is highlighted.
- You can create your own envelopes manually, by manipulating the nodes and lines, or you may record an envelope, as discussed in the “Morph Envelope/Record Path” section, on page 405.
- To adjust the *time* between nodes, click on the desired handle, and drag it left or right. As you do so, the overall length of the Morph Envelope will change—with all following nodes—being moved.

You cannot move a node beyond the position of the preceding node. You can, however, move nodes beyond the position of the following node—even beyond the right-hand side of the Envelope window—effectively lengthening both the envelope segment and the overall envelope.

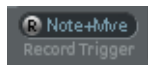
Morph Envelope/Record Path

R(ecord) Enable

The *R* button enables the Morph Envelope record functionality. This button works in a similar fashion to the record arm buttons in Logic.

To make a Morph Envelope recording, just press the *R* button, play a note, and start moving the silver ball in the Morph Pad with the mouse. You can also make use of an external controller (see the “Morph X/Morph Y” section, on page 409).

The *Mode* is automatically set to *Pad only* as soon as the *R* button is pressed (for more information on “Modes,” see the “Morph Envelope Parameters” section, on page 406).



Record Trigger Mode

The pull-down menu to the right of the *R* button is used to select different “trigger” modes to start recording (when *R(ecord) Enable* is active):

- *NoteOn*: recording starts when a note is played.
- *Note+Ctrl*: recording starts when MIDI control change messages (for the assigned *Morph X/Y* controllers) arrive while a note is held.
- *Note+Sus*: recording starts when the sustain pedal is depressed while a note is held.

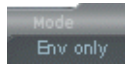
Recording is stopped by pressing the *R(ecord) Enable* button (or trigger) a second time. Once all keys are released, and all voices have completed their decay phase, the recording ends. You can stop recording early by releasing all keys, and then pressing a single key.

Following the recording of a controller movement, *R(ecord) Enable* is automatically set to *off* and *Mode* is set to *Env only*. This ensures that only the recorded movement will be active, regardless of the “stop” position of the recorded controller.

Morph Envelope Parameters

Mode

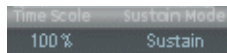
Activates the Morph Envelope, and allows you to select from between the following modes:



- *Off*—Morph functionality is disabled.
- *Pad only*—envelope is deactivated, and morph functionality is controlled by the morph cursor (the “ball”) and/or X/Y MIDI controllers only.
- *Env only*—envelope is running, but the morph cursor and X/Y MIDI controllers are deactivated.
- *Env+Pad*—envelope is running, and the position of the morph cursor and/or X/Y MIDI controllers is used as an offset for any envelope movements.
- *Env+Point Set*—envelope is running. The selected envelope point can be edited by moving the morph cursor in the pad or via MIDI (“MorphX” and “MorphY” Controller Assignments).
- *Point Solo*—envelope in a kind of “still picture” mode. the selected envelope point can be edited by moving the morph cursor in the pad.

Time Scale

This parameter scales the duration of the entire envelope between 10% and 1000%.



Sustain Mode

Allows you to select the behavior of the Morph Envelope while a note is held. Choices are: the usual *sustain* mode, *finish* mode or one of three loop modes (*forward*, *backward*, *alternate*).

When in any of the loop modes, the loop always cycles between the *loop* and *sustain* Envelope “handles” (the nodes indicated by the small “L” and “S” icons). The Morph Envelope can—like any envelope—run in one shot (as long as the note is sustained). It can also run several times or in an infinite cycle, much like an LFO. You can achieve this through the use of *loops*.

Note that if one of the three loop modes is selected, and the loop point is positioned before the sustain point, the loop will be active until the key is released. Following key release, the envelope will then continue beyond the sustain point, as per usual.

If the loop point is positioned after the sustain point, the loop will be entered as soon as the key is released, and will be cycled infinitely (until the complete voice is released by finishing the release phase of the amplitude envelope).

The *loop* and *sustain* point icons (the small “L” and “S”) can be directly grabbed and repositioned. Note that doing so can potentially alter the loop (and the overall Morph Envelope) length.

- When set to *finish*, the Morph Envelope runs in “one shot mode” from its beginning to its end—even if the note is released before the envelope has come to its end. The other loop parameters are disabled.
- When set to *forward*, the Envelope runs to the sustain point (indicated by the blue “S” icon) and begins to repeat the section between the loop point (indicated by the blue “L” icon) and sustain point periodically—always in a forward direction.
- When set to *backward*, the Envelope runs to the sustain point (“S” icon) and begins to repeat the section between the sustain point and loop start point (“L” icon) periodically—always in a backward direction.
- When set to *alternate*, the Envelope runs to the sustain point (“S” icon) and returns to the loop point (“L” icon) and back to the sustain point periodically, alternating in both a backward and forward direction.

Loops can be synchronized to the song tempo automatically via the *sync/ms* buttons.

Sync/ms

These parameters allow you to select between a free-running envelope (with segment times displayed in milliseconds) or a tempo-synced envelope with note value options, such as *1/8th* or *1/4*.



Switching between values forces a recalculation of times to the nearest note value or ms time, respectively, based on the current song tempo.

Depth

This parameter scales the amount of morph movement caused by the Morph Envelope. The effect of the *Depth* parameter is visually displayed in the Morph Pad. As you increase/decrease the value, the morph trajectory will also be scaled.



Modulation (Source and Depth)

These parameters allow you to select a modulation source and amount, which are used to scale the movement of the Morph Envelope.

Transition

Provides control over the “transitions” between the morph points. This can range from the original (possibly recorded) movement to linear connections, and beyond this to “stepped” transitions.



By the latter, we mean remaining at one morph state during the entire Morph Envelope segment, and then abruptly switching to the morph state set at the following envelope point.

This parameter (and the Morph Envelope itself) can lead to interesting evolving sounds or even rhythmic patches.

MIDI Controller Assignments

This section allows you to define the MIDI controllers you wish to use for vibrato depth control or morph pad movements, for example. These parameters are saved with each setting. They are only updated if the default setting (that is loaded on instantiating the plug-in) is used, or if the setting was saved with a song.

This approach helps you to adapt all MIDI controllers to the keyboard, without having to edit and save each setting separately.

MIDI Controller Assign	Vib Depth Ctrl Touch	Ctrl A 1 ModWheel	Ctrl B 4 Foot	CtrlEnv 1 1 ModWheel	CtrlEnv 2 4 Foot	Morph X 16 GP1	Morph Y 17 GP2	Mode use default / keep
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Learn

All parameters that allow you to select a MIDI controller (VibDepth Ctrl, CtrlA, CtrlB, CtrlEnv1, CtrlEnv2, MorphX, Morph Y) offer a *Learn* option. If selected, the parameter will automatically be set to use the first appropriate incoming MIDI message.

Vib Depth Ctrl

Defines the MIDI controller used for vibrato depth control.

Ctrl A/Ctrl B

Allows the assignment of two controllers that can be used as “via” modulation signals, for direct modulation routings found on the *CtrlA/CtrlB* tab or for side chain modulations.

CtrlEnv 1/CtrlEnv 2

Sets the controller assignments for the two Control Envelopes—used as a modulation signal or an offset—in cases where the Control Envelope is set to *Ctrl only* or *Ctrl+Env* modes. It also is used to define the source for recording controller movements.

Morph X/Morph Y

Determines the controller assignments for the X and Y co-ordinates of the Morph Pad. Once assigned, the controller can be used to: manually move the morph point, program single morph envelope points, shift the entire morph envelope and as source for recording morph movements.

Mode

The two *Mode* menu entries allow you to select whether the MIDI controller assignments shall be taken from a setting or left as they were, when loading a setting. Switching between modes toggles between the original assignments saved with the setting and the default assignments (taken from #default.pst setting—which is loaded on instantiation of Sculpture, if it exists).

Programming: Quick Start Guide

This section of the manual contains a collection of programming guidelines, tips, tricks, and information to assist you in creating particular types of sounds. A more involved look at programming can be found in “Programming: In Depth” section, on page 426.

Approaches to Programming

Given the flexibility of Sculpture’s synthesis core, you can take a number of different approaches to sound design.

By this, we mean that if you’re the type who prefers to sculpt a sound from scratch—parameter by parameter—you can.

If you prefer to make use of Sculpture’s morphing capabilities to create new sounds, you can also do this. This is discussed in the “A Brief Randomizing Tutorial” section, on page 403.

If you’re more of a “tweaker” of factory or user patches, then the parameters that affect the entire instrument may be more your style. These include, the Body EQ and Filter sections, plus the Modulators, for example.

Whatever camp you fall into, you’ll be able to achieve new (and hopefully interesting) results.

At the end of the day, however, we encourage you to experiment, and familiarize yourself, with each approach. You will find that each has its strengths and weaknesses, and that a combination of methods may strike the balance between satisfying sounds and a social life.

Basics

Throughout the manual, we have followed the signal flow of the core synthesis engine.

When programming “from scratch,” this is the approach you should also take, working on each component of the sound in isolation.

Obviously, when you're starting out with Sculpture, you won't be familiar with the impact of each parameter on your end results. Don't sweat it, we'll provide some pointers for particular types of sounds in a few moments ... but let's get back to it.

First up, you'll need a "plain vanilla" or "from scratch" patch. When you first launch Sculpture, this is exactly what you'll get—a default set of "neutral" parameters. Sonically, this patch won't set your heart racing, but it will provide you with a starting point for all of the examples in this chapter.

Note: Save this patch as a Settings file. Name it as desired—maybe “neutral,” “plain vanilla” or “from scratch”? This Setting can be reloaded as you work through the examples.

The Core Engine

We discussed the signal flow in the “The Synthesis Core of Sculpture” section, on page 356. To recap, and explain in a more “hands on” way, please follow these steps. This section is intentionally simplified, but we ask for your forbearance. Knowing the “mechanics” of Sculpture is essential to your success:

The String

The “string” is the central synthesis element. It offers a range of parameters that allow you to adjust its material—what it’s made of and what environment it’s being played in (water, air, and so on).

Each parameter can be explored further in the “String and Object Parameters” on page 362.

Before starting, we recommend that you Control-click on the “string” (the green horizontal line in the Pickup section) to activate/deactivate string animation. When active, the string will vibrate, making it easier to visualize the impact of the objects and pickups.



- Press the *Keyscale* button at the bottom of the *Material Pad* “ring”, as shown in the graphic.
- Strike and hold and/or repeatedly strike middle “C” on your keyboard.

Middle “C” is the default pitch/key of the string.

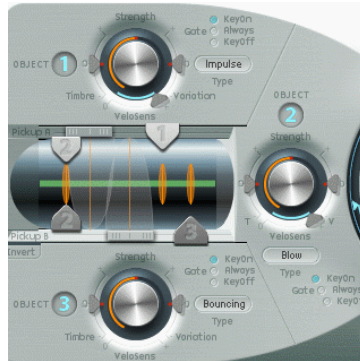
- While doing so, grab hold of the “ball” on the *Material Pad* by click-holding on it with your mouse, and move it around. Listen to the sonic changes as you move between the *Nylon*, *Wood*, *Steel*, and *Glass* materials. Keep an eye on the *String* (the green horizontal line in the *Pickup* section, to the left) as you’re doing so.
- Release the mouse button once you’ve found a basic tone that you like.
- Now, experiment with the values of each of the sliders that surround the *Material Pad*—namely the *Media Loss*, *Tension Mod* and *Resolution* parameters (while continuing to strike middle “C”). Take note of the changes that each makes to the sound, and also the string animation in the *Pickup* section. Play a few notes above and below middle “C”, again keeping your eye on the string.
- You probably noticed that moving the *Media Loss*, *Tension Mod* and *Resolution* sliders also had an impact on the green and blue *Keyscale* sliders inside and outside the “ring”. Grab, and drag each of these “keyscale” slider arrowheads to different positions—one by one—and play a few notes either side of middle “C” as you’re doing so. Note the changes that happen up/down the keyboard range.
- Once done, press the *Release* button at the bottom of the *Material Pad* “ring”, and adjust the blue *Media Loss Release* slider while striking notes.

The Objects

Up to three “objects” of different types are used to excite or disturb the vibration of the string. Each parameter can be explored further in the “String and Object Parameters” on page 362.

- Please reload your default or “plain vanilla” Setting.
- Now press the (Object) 1 button (so that it turns gray), while striking a key. You’ll note that you hear nothing. The string itself doesn’t make a sound unless it is “stimulated” by the objects. Reactivate the button by clicking on it again.
- Now, click-hold on *Object 1’s Type* pull-down menu, and select each entry in the list. Strike a note repeatedly while doing so to hear the impact of each object “type” on the string. As always, keep an eye on the *String* animation.

- Adjust the *Strength* knob, by click-holding it, and moving your mouse vertically for large changes, or horizontally for fine adjustments. Strike a note repeatedly while doing so.

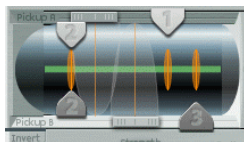


The three string Object dials/controls are shown, along with the Pickup section at the center left.

- Drag the *Timbre* and *VeloSens* arrowheads to different positions while striking a key to audition the changes that they bring. The impact of the *Variation* parameter is indicated on the charts found in the “Type” section, on page 368.
- Try out each of the *Gate* options.
- Once you’ve settled on a particular group of settings for *Object 1*, activate *Object 2* by pressing on the 2 button.
- Change parameters for this object as desired, and note the interaction of the two Objects with each other and the string.
- Do the same for *Object 3*.

The Pickups

The vibration of the string is captured by two movable “pickups.” The *Pickup* section also houses three sliders—one for each object.



- Reload the “plain vanilla” Setting.
- Click-hold on the *Object 1 Pickup* “handle”—the down arrowhead with the number “1” on it—and drag it left/right while striking a key. You’ll note that adjustments to the Object pickup position alters the tonal characteristics of the string.
- Adjust *Object 1*’s *Strength* control to hear things better, or adjust the tone, if desired. You may also wish to make use of the Object’s *Timbre* and *Variation* parameters to alter the tone. Use the table in the “Type” section, on page 368.

- Do the same with the *Pickup A* and *Pickup B* sliders. You'll note that changes to the Pickup positions result in quite different String vibrations (and tonal qualities). Adjust the *Level* control (directly opposite the *Pickup* section, on the right of Sculpture's GUI) to increase volume, if desired.
- Adjust the pickups of the other two Objects. Adjust each Object's *Strength*, *Timbre*, and *Variation* parameters to alter the tone. Make use of the tables in the "Type" section, on page 368.

Note that Object 1 can only make use of the excite Types found in the first table. Object 2 can make use of any of the Types available in either table. Object 3 can only make use of the disturb/damp Types found in the second table.

- You can disable any of the Objects at any time by pressing on their numerical buttons (1, 2, and 3).

About Parameter Interactions...

As you're probably discovering, each parameter has an impact on the overall tone of the string—and more often than not—an impact on the string interaction of other parameters.

In effect, each parameter that you introduce or make changes to, will affect the modelled string.

This will, in turn, affect the interaction of each parameter with the modelled string.

As such, parameter settings that you have already made for, say *Object 1*, may need to be adjusted when *Object 2* is activated.

Generally, such adjustments won't need to be radical, and may only involve a small tweak to the *Strength* parameters, or perhaps the *Pickup* positions of each Object, for example. These parameters have the greatest impact on the tone and level of the Objects, and should be the first things you look at if the introduction of *Object 2* results in an unwanted change to the color of your sound.

You may want to "fine tune" the Objects further through use of the *Timbre* and *Variation* controls.

Small changes—rather than radical ones—will retain the general tonal character of the string (and *Object 1*), while introducing the new flavor of *Object 2*.

Back to it ...

Processing

From the *Pickups*, the signal is sent to the processing section, which consists of the ADSR-equipped *Amplitude* stage, a *Waveshaper* (with selectable types of waveshaping curves) and a multi-mode *Filter*.

These processors and the Pickup parameters are covered in “Processing” on page 373. Feel free to experiment with these, while referring to the individual parameter descriptions.

All elements that we’ve covered thus far exist on a per-voice basis.

Additional Processing

All voice signals coming from the *Pickups* are summed, and then processed by an integrated *Stereo Delay* effect.

From there, the signal is sent to an EQ-like module (*Body EQ*), which globally simulates the spectral shape/body response of your “instrument.” There are several “body types” to choose from.

The resulting signal is then fed to a *Level/Limiter* section.

We invite you to explore all of the parameters available in these processing sections on your own—using the “plain vanilla” patch each time. This will give you a general “feel” for each parameter, and its impact on the sounds you hear.

All other parameters on the lower portions of the Sculpture GUI (Modulation, Morph, Envelope, and Controller Assignments) are not part of the *core synthesis engine*, although they can obviously impact upon it. We’ll discuss some tips and uses of these parameters shortly.

Creating Basic Sounds

This section covers the creation of basic types of sounds, such as organs, basses, guitars, and so on.

The idea here is to provide you with a starting point for your own experimentation, and to introduce you to different approaches for tone creation with Sculpture.

As you become familiar with the synthesizer, and component modelling, you’ll find that there are many ways to achieve an end result. By this, we mean that each “component” of the sound can be modelled using different techniques and tools available in Sculpture.

This flexible approach allows you to create, say a “brass” sound, in several ways—using the Waveshaper as a major tonal element in one patch. In another brass patch, the Filter and Body EQ can be used to emulate the same sonic component that the Waveshaper provided in the first patch.

A good understanding of the physical properties of the instrument that you are trying to emulate is obviously advantageous. This type of knowledge, however, is not common to most people, but it can be found online.

You can certainly do some detailed research, but for most sound creation tasks with Sculpture, you can follow this general breakdown formula when creating your “string”.

How is the sound of the instrument created?

By this, we mean:

- is it a string that is vibrating and resonating in a “box” (guitar or violin, as examples)?
- is it a column of air that is vibrating in a tube (flute, trumpet)?
- is it a solid object that is struck, causing vibration (woodblock)?
- is it a hollow object that is struck, causing vibration/resonance? (drum, bell)?

What are the physical properties of the instrument?

In other words, what is it made of?

When answering this question, don’t just consider the body of the instrument. Take into account the “string” material—nylon or steel on a guitar, or perhaps the thickness and material of the reed in a clarinet or oboe, or a mute in a trumpet.

Is it polyphonic or monophonic?

This is a pretty significant factor, that ties into the next question. Apart from the obvious things such as the inability to play chords on a flute, a modelled “string” will interact with any currently active string. This, of course, can’t happen in a flute. It’s strictly a one-note instrument.

How is it played?

Is it bowed, blown, struck, plucked?

Are there other characteristics that contribute to the sonic character of the instrument?

Examples of this are:

- changes to lip pressure and mouth position with brass and wind instruments.
- breath or mechanical noises.
- momentary pitch changes—as an example, when fingers are pressed into a fretboard, or when a string is plucked.
- momentary tonal or level changes—such as when brass players are running out of breath, or “fluttering” the valves.

Once you’ve mentally, or physically, written down your “list” of properties, try to emulate each *component* that contributes to the sound’s character. This is what component modelling is all about.

Important: Before beginning, we'd like to stress that the following examples are just that—examples. There are many ways to model each component of the sound. We encourage you to experiment with the settings that are suggested to create your own “versions” of patches. You'll note that specific parameter values are rarely given, and if they are, feel free to use another.

Just to balance the ledger a little. Subtle changes—particularly when it comes to Keyscaling parameters—result in more “controlled” sounds. Take your time, and try everything as you're following these examples.

You should make use of other users patches, and the factory Settings that ship with the synth. Close study of these will provide you with an insight into how the sound was created. Enable/disable different parts of Sculpture to see what each does, and then use this knowledge for your next sonic masterpiece.

Have fun and take risks—you can't break anything!

Basses

Creating bass sounds with Sculpture is pretty straightforward.

- Load up your “plain vanilla” Setting.
- Set the Transpose parameter to 1 Oct., and play a few notes around C2. You'll note that the general color of an acoustic bass is already there.
- You can certainly adjust the ball on the Material Pad towards the “Nylon” entry, but before doing so, we recommend that you change Object 1's Type to “Pick”.
- Have a play on the keyboard, and adjust the ball position while doing so.
- Now take a look at the *Strength*, *Variation*, *Timbre*, and *VeloSens* parameters of Object 1. Adjust each in turn, to taste.
- You may also wish to adjust the Amplitude Envelope *Release* parameter.
- To make your bass more “woody”, adjust Object 1's Pickup position towards the right. At extreme positions (left or right-hand end), the bottom-end of your bass will be gutted. Try it out!
- Now, adjust the position of Pickup A and Pickup B. As you'll hear, you can quickly recreate a picked acoustic or electric bass sound.
- To instantly make it a hybrid (or full-on) synthesizer bass, press the *WaveShaper* button, and select from the different Types.
- Save Setting as... with new names as you go. We're sure that you'll come up with several in just a few minutes. Once again, these can be used as templates for future bass sounds.

Bells

At a basic level, bell-like sounds are quite easy to produce with Sculpture. The creation of truly interesting bells involves a little more effort, but the harmonic richness and detuning during the decay/release phase makes all the difference ...

- Load your “plain vanilla” patch.
- Set Object 1’s *Type* to *Strike*.
- Move the Material Pad ball to the very bottom of the pad, and place it pretty much halfway between the *Steel* and *Glass* entries. If you play a few notes, you’ll hear that the sound is already more “bell” like.
- Now grab the *Media Loss* slider, and drag it nearly all the way down. Again, play a few notes, and you’ll hear that the release phase of the sound is considerably longer.
- Drag the *Resolution* slider all the way to the right.
- Drag Pickup A’s position to around halfway (0.48).
- Drag Object 1’s pickup position to a value of 0.10. We’re starting to get pretty bells now ... play a few notes.
- Now click on the Stereo Delay button to activate the Delay unit.
- Click the (Delay) *sync* button, and drag the *Delay Base Time* slider to a value of 20 ms.
- Increase the (Delay) *Wet Level* to 66%.
- Activate the Body EQ by pressing the Body EQ button. Ensure that the *Lo Mid Hi* model is selected.
- Adjust the *Low* level to 0.55, the *Mid* to 0.32, and the *Hi* to 0.20.
- At this point, you will have a working bell sound—BUT—you’ll probably find that there is a tuning issue below C3, in particular. We took this approach to programming as the harmonics of the sound are most noticeable after all other parameters have been set. The solution to the tuning issue primarily lies in the *Inner Loss* and *Stiffness KeyScaling* parameters. To adjust, first select the Keyscale button, then click-hold on the *green* horizontal line (within the Material Pad) for low notes, or the *blue* horizontal line for high notes—and drag up/down to the desired position.
- Save Setting as... with a new name, and use it as the basis for new bell sounds, or your next Christmas album.

Brass

Brass instruments are notoriously difficult to recreate with electronic instruments. Samplers do a reasonable job in the right hands, and with the right sample library, but they lack the “organic warmth” of a real live brass player. This is a simple and generic brass patch that can be played as a solo instrument or as a brass section.

- Load your “plain vanilla” patch.
- Set Object 1’s *Type* to *Blow*.
- Activate Object 2, and set its *Type* to *Noise*.
- Adjust the *Strength* of Object 1 to around 0.90.
- Set Object 1 *VeloSens* to around 0.30.
- Drag the Material Pad ball to a position that is diagonally between the “I” of *Inner Loss*, and the “I” of the word *Steel*, while playing middle “C”. The sound should be quite “brassy”.
- Now play the “E” above middle “C”, and you’ll hear a weird “mandolin meets a telephone” kind of sound.
- Drag the Resolution slider to the left/right while playing middle “C” and a few notes down an octave or so. You’ll discover a range of sounds that cover everything from sitars to flutes is possible, just through manipulation of this parameter.
- Now click on the Keyscale button and—while playing up/down the keyboard— independently adjust the Resolution slider, plus the Resolution Low/High Keyscale sliders until the range of the keyboard that you wish to play (say an octave or so around middle “C”) doesn’t suffer from those mandolin/phone artefacts. Oh, and make sure that your sound is still “brassy”.
- Move Pickup A’s position to around 77%.
- Turn on the Waveshaper and select *Scream* as your preferred *Type*. Adjust the *Input Scale* and *Variation* parameters to taste.
- Turn on the Filter. Select *HiPass* mode, and adjust the *Cutoff*, *Resonance*, and other filter parameters to taste. (As a suggestion, *Cutoff* at 0.30 and *Resonance* at 0.41).
- Save Setting as... with a new name.

Please explore this patch much further. There are a great number of directions that it could be taken in—as a muted trumpet, french horns and even sitars or flutes. The Waveshaper has a significant impact on this sound, and this is the first place you should look to radically alter it.

Use the Stereo Delay to create “space” and the Body EQ to cut the lows and boost the Mids and Hi’s.

Adjust the Material Pad ball position towards the *Nylon* entry, select *Blow* as Object 2’s *Type*, and then experiment with the Object 1 and 2 positions. This can also result in different “brass” sounds.

Flute

This can be used as the basis for most instruments in the “wind” family, including flutes, clarinets, shakuhachi, pan pipes, and so on.

- Load your “plain vanilla” patch.
- Keyboard Mode should, theoretically at least, be set to “mono” as flutes and other wind instruments are monophonic. After you’ve set up the patch, experiment with this parameter while playing, and make your choice.
- Set Object 1’s *Type* to *Blow*.
- Set Object 2’s *Type* to *Noise*.
- Set the *Gate* of both Objects to *Always*.
- Adjust Object 2’s *Strength* to a value of around 0.25.
- Adjust Object 1’s *Velosens* parameter to a value around 0.33.
- Move the Material Pad ball to a position that is pretty much at the end of the Inner Loss entry (below the word “Nylon”).
- Play the keyboard and you should hear a flute-like sound, but with a long release—which we obviously don’t want. Drag the Amplitude Envelope Release slider down to around 0.99 ms.
- Pickup A should be set to a value of 1.00 (far right).
- Set Object 1’s pickup position to around 0.27.
- Set Object 2’s pickup position to around 0.57.
- Now activate the *Waveshaper* by pressing on the *Waveshaper* button, and select the *Tube-like Distortion* Type.
- Play a few notes, and adjust the *Waveshaper Input Scale* and *Variation* parameters to taste. (*In. Scl* = 0.16/*Var* 0.55, as examples).
- As you play sustained notes, you’ll probably notice a distinct lack of interesting timbral shifts (typical of “real” flute sounds—due to changes in the player’s breath, lip position and so on) as the note is held.
- A number of approaches can be taken to add interest to the sustained sound. These include; using the *Vibrato* modulator (assigned to *aftertouch*, perhaps), or perhaps recording or drawing in an *Envelope*, and controlling the *Waveshape Inner Scale* via *Velocity* and/or *String Media Loss*. You could even use the *Loop Alternate Sustain Mode*. Feel free to experiment!
- Save Setting as... with a new name.

Guitar

Guitar, lute, mandolin, and other plucked type instruments, including harps, can be created from this basic patch.

- Load your “plain vanilla” patch.
- Set the *Voices* parameter to a value of 6—there’s only 6 strings on a guitar. Obviously, pick 7 for a banjo, as many as possible for a harp.
- Set Object 1’s *Type* to *Impulse*, if it’s not already there.
- Activate Object 2 and set its *Type* to *Pick*.
- Now move Pickup A’s position to the extreme right.
- Move Object 2’s Pickup position to a value of 0.14.
- Activate the *Body EQ*, and select one of the *Guitar* models.
- Adjust the various Body EQ parameters. These have a major impact on the overall brightness and tone of your guitar sound. (Suggestions—*Guitar 2*, *Int*—0.46, *Shift*—0.38 and *Stretch*—0.20).
- *Finestructure* should be at a value around 0.30 to 0.35—but use your ears to judge this. Remember that an increase in the *Finestructure* value results in a greater load on the computer CPU.
- Click-hold on the *Stereo Pickup* button, and drag your mouse upwards—to increase the perception of stereo width (a value around the 10 o’clock/2 o’clock mark is nice).
- Activate the Filter, and select *Lo Pass* mode.
- Adjust the *Cutoff* and *Resonance* parameters to taste. (suggestion—both at 0.81).
- Adjust the *Tension Mod* slider upwards, and play the keyboard to see how the momentary detuning effect caused by this parameter affects the sound. Set it to an appropriate amount.
- Set the *Level Limiter* to *Both*.
- Save Setting as... with a new name.

You may have noticed that we departed slightly from the signal path of the Core synthesis engine in the creation of this Setting. The reason for this is the major impact that the Body EQ model has on the sound.

In some cases, like this one, it may be better to work slightly out of sequence, rather than strictly follow the signal flow.

Obviously, this is but one “guitar.” You can make use of the Object Strength, Variation, and Timbre parameters, not to mention repositioning the Material Pad ball to create a completely different “tone” to your guitar.

For quick and easy mandolins, make use of the Stereo Delay (or Vibrato) to emulate the double-strike picking that is associated with the instrument.

Organ

Organ sounds are amongst the easiest and quickest sounds to emulate in Sculpture as they have no release phase. This simplifies programming as there is no real need to set up Keyscaling parameters to create the basic tone. You may, however, wish to do so at a later stage—for modulation routing or specific sound design purposes.

We encourage you to play notes/chords *while* you are adjusting parameters. This way, you can hear what each parameter is doing to the sound.

- Load your “plain vanilla” patch.
(*Object 1’s Type* should be set to *Impulse*. If it isn’t, change it now.)
- Increase the *Voices* parameter to a value of 8 (or higher if you wish).
- Move the Material Pad ball *into* the top left corner.
- Activate Object 2 by pressing the 2 button.
- Change Object 2’s *Type* to *Bow*.
- Set Object 2’s *Gate Time* to *Always*.
- Pull the *Release* slider of the Amplitude Envelope all the way down.
- Play a “C” chord, and you’ll hear a “flute-like” sound.
- Drag Pickup A to the extreme right.
- Play a “C” chord, and you’ll hear a cheesy “organ” sound. As you can see Pickup A’s position has a significant impact on the overall sonic character of the sound.
- Grab Object 2’s Pickup, and move it around while holding down the “C” chord. Once you’ve found a position that meets your “that sounds like an organ” criteria, release the Object pickup.
- Now, very slightly adjust Object 2’s *Timbre* parameter upwards.
- Adjust Object 2’s *Variation* parameter slightly downwards, and upwards, until you find a tone that you like.
- You may, at this point, want to move the Object 2 Pickup parameter to another position. Hold down a chord while doing so.
- Further “tweaks” can be made to the Variation and Timbre parameters of Object 2.
- To introduce a little “key click”, change Object 1’s *Type* to *Strike*, and adjust its *Strength* and *Timbre*.
- To add a little of that “detuned organ” vibe, set the *Warmth* parameter between 0.150 to 0.200.

At this point, you should have a basic organ tone. Save Setting as... with a new name. You can use this as the basis for your next organ patch.

You’ll probably notice some intermodulations that are introduced when you’re playing chords. Apart from the pitch differences between notes in the chord, this is a result of the interactions between each *voice* being produced by Sculpture. These slight variations between each voice (or string, if you will), and their harmonic interaction with each other are not dissimilar to the harmonic interactions of a violin section in an orchestra—even when playing identical lines.

Percussion

Percussive sounds, such as drums, tend to share a similar type of envelope. They contain a “strike” element, where most of the sonic character is exhibited, followed by a short decay phase. The release phase will vary—dependent on the instrument itself (a snare drum as opposed to a woodblock), and the ambient space it is placed in—a cavern, a bathroom and so on.

- Load your “plain vanilla” patch.
- Set Object 1’s *Type* to *Strike*.
- Activate Object 2, and set its *Type* to *Disturb 2*.
- Set Object 2’s *Gate* to *Always*.
- Object 1’s *Strength* should be about 0.84.
- Object 2’s *Strength* should be about 0.34.
- Adjust the *Media Loss* value upwards/downwards while playing to hear its impact. Find a suitable setting.
- Similarly, you can alter the Material Pad ball position—although its impact on the overall tone of the sound is heavily reliant on the *Media Loss* value.
- Activate the *Body EQ* and/or *Filter*, and adjust to your heart’s desire.
- Save Setting as... with a new name.

This sound can be used as the starting point for a vast range of percussive sounds—including drums, blocks, “industrial” percussion and even rhythmic sequenced synth sounds.

Just adjust the ball position in the Material Pad, and alter the Media Loss slider position.

Solo Strings—Cello Bowed

Solo stringed instruments, such as violins and cellos, that are played with a bow can be created in much the same way. This sound can also be played polyphonically.

- Load your “plain vanilla” patch.
- Set *Transpose* to *–1 Oct*.
- Set Object 1’s *Type* to *Bow*.
- Play the lower half of your MIDI keyboard, and you’ll hear a viola/cello like sound. This could obviously be improved.
- Set the Object 1 *Velosens* slider to match your playing style and that of the music, as you’re playing the keyboard. Adjust later, if desired.
- Grab the *Tension Mod* slider, and move it slightly upwards, so that the arrowhead covers the “D”. This emulates the momentary detuning effect of the bow stretching the string.
- Move Pickup A to a position around 0.90.
- Move Object 1’s Pickup position to a value around 0.48.
- Activate the *Body EQ*, and select the *Violin 1* model.
- Set the Body EQ parameters as follows: *Int*—0.73, *Shift*—+1.00, *Stretch*—+1.00.
- Adjust the *Fine Structure* to taste.

- Click-hold on the Stereo Pickup dial, and drag upwards until the 10.30/1.30 positions are reached.
- Set the *Level Limiter* to *both*.
- Save Setting as... with a new name.

We encourage you to set up your own modulations for this sound. The most common thing that springs to mind is the introduction of vibrato into the sound after a short period.

The creation of higher pitched solo string instruments is much the same as the example above but special attention must be given to ALL Keyscaling parameters. The Body EQ also has a large bearing on the upper octaves, so take care with its parameters.

Setting Bonus:

Simply change Object 1's Type to "Pick", and you'll have a round and rubbery synth bass sound in the lower octaves, and a passable "harp" across the rest of the keyboard.

Synthesizers

One of Sculpture's great strengths is the creation of endlessly evolving pad and atmospheric sounds. It can also easily do fat synth basses (which you hopefully discovered while following the "Basses" section, on page 416), powerful leads and other types of typical synthesizer sounds.

Sculpture has an advantage over traditional synthesizers as its core synthesis engine produces a wider variety of basic tones, and these tones have an "organic" quality and richness to them.

Basic Pad

- Load your "plain vanilla" patch.
- Set the *Voices* parameter to 16.
- Set Object 1's *Type* to *Bow*.
- Set Object 2's *Type* to *Bow Wide*.
- Grab the Material Pad ball, and position it at the extreme left of the Pad, exactly halfway between the top/bottom—on the "Material" line.
- Play a "C" chord (middle "C"), and you'll hear a pad sound.
- Move Pickup A to a position around 0.75, and the pad will become a little sweeter.
- Move Object 1's position to a value of 0.84.
- Move Object 2's position to a value of 0.34.
- As a final step, click on the Points icon that features five dots in the Morph Pad section.
- Set the "Int" slider in the Morph Pad Randomize section to a value of say 25%.
- Click once on the Morph "Rnd" button.
- Save Setting as... with a new name—say "vanilla_pad".

We'll be using this basic pad sound for the following examples. Don't be shy about doctoring the "vanilla pad"—anything goes, so make use of the Filter, the Delay, EQ, and Waveshaper to create new sounds.

Evolver

- Load your "vanilla_pad" patch.
- Click on the LFO 1 tab at the bottom of the GUI.
- Press the *1* button, and play the keyboard. Not much difference, there, huh?
- Now, click-hold on the Min/Max sliders, and drag left and right, while holding down a chord. Finally settle on a value of *0.15*.
- Click-hold on the *Target (Pitch)*, and select *Object 1 Strength*. You'll hear a "fluttering" sound.
- Now click on the "sync" button, and adjust the Rate knob to a value of *1/8t*.
- Activate the second LFO 1 object by pressing the *1* button, and select *Object 1 Position* as the Target.
- If you play the keyboard, there's not much that's different.
- Set the *via* option of the second LFO 1 object to *Velocity*.
- Play the keyboard at different velocities, and you'll hear some shifting of the Object 1 Pickup position ... and now, to make it interesting ...
- Change the Waveform to *Sample&Hold*, and play the keyboard at different velocities. If you've got a sustain pedal, use it. Listen to the endlessly evolving sound.
- You may like to experiment with the song tempo and LFO Rate.
- You may want to alter the Stereo Pickup value, and introduce LFO 2 or the other modulators.

Morpher

- Load your "vanilla_pad" patch.
- Click on the R(ecord) button in the Morph Trigger section.
- Play a chord on the keyboard, and drag the Morph Pad ball in a circle.
- Once you're done, press the R(ecord) button.
- Now change the *Morph Mode* to *Env only*, and you should see your Morph circle.
- Play the keyboard. There's your morphed pad!
- Feel free to adjust the Morph Envelope parameters.

Remember when you were asked to use the Morph Points, Intensity, and Rnd parameters while setting up the "vanilla_pad"?

This was to ensure that there were several Morph points already available for your morphing pleasure.

You can, if you wish, retain the path of your morphed pad, and continue to press the *RND* button and adjust the Intensity slider for an endless variety of sounds.

Modulations

The modulation options can be very important for the emulation of acoustic instruments. As a simple example, the introduction of vibrato into a trumpet sound over time.

Many “classic” synthesizer sounds also rely as much on modulation as they do on the basic sound source components—the VCO, VCF, and VCA.

Here’s a number of quick modulation tips ...

- Let’s say you want to modulate the timbre of Object 2, with the LFO, for example. To do so, click on the LFO 1 or 2 Tab, press the 1 or 2 button, select the desired Source/Target and value. That’s it!
- To control any modulation with an external controller, such as your keyboard’s modulation wheel, set the desired “Via” parameter to *CtrlA* or *CtrlB* respectively. By default, the Mod Wheel is set to *CtrlA*.
- The “Bouncing” damp Type available to Object 3 affects the sound in a very interesting way, but it cannot be run synchronously with the song tempo. To create a similar effect to the “Bouncing” Object—but in-sync, you could use a “Disturb” object Type, and move it by modulating its vertical position (Timbre) with an LFO.

“Breath” control is available to all users of Sculpture, even if you don’t own a breath controller. To do so, record breath controller modulations into the recordable envelopes (using a mod wheel or other controller), and then reassign the recorded modulation path (use the *CtrlEnv 1* and/or *2* parameters) with each NoteOn.

Programming: In Depth

This tutorial explains how you can program sounds with Sculpture from scratch. Based on Sculpture's string model, you'll learn how to use the individual sound shaping parameters in order to recreate the physical properties of an instrument in detail.

Note: You will find the settings for these tutorials in the Tutorial Settings folder in the settings menu (in the head of the Sculpture Plug-in window).

Programming Electric Basses with Sculpture

We'll be concentrating on a single instrument type: the electric bass, including all of its important variations and articulations. The physical nature of electric basses is not as complex—and encumbered with acoustic issues—as is the case for many acoustic instruments. This instrument is therefore an excellent choice for our sound programming tutorial, the goal of which is to acquaint you with the art of using Sculpture to accurately reproduce detailed sounds.

In order to build a bass (including all components) in Sculpture, it is necessary to understand the basic, physical process of sound production within the instrument. Before we look at the practical programming process within Sculpture, you'll find detailed information on the construction of electric basses in the next section.

The Most Important Aspects of Electric Basses

In general, the electric bass has four strings. The lowest string is usually tuned to E 0 or E (MIDI note number 28). The strings above the low E are tuned in fourths, thus A, D, and G. You can, of course, find basses today that have five, six, and even more strings. As Sculpture has no tonal limits, this is of little concern to us.

What is much more important for sound programming is the overtone content of the bass sound. This depends primarily on the qualities of the strings.

- Round wound strings: a very fine wire is wound around a steel cable core which results in a wiry, metallic sound that's full of overtones.
- Flat wound strings: (which are much less popular today) the fine wire wrapping is ground down or polished smooth, and the sound has far fewer overtones in comparison.

In contrast to guitar strings, the structure and workmanship are the same for all strings in a set. Sets combining wound and non-wound strings do not exist.

The relationship between string length and string tension has a significant impact on the overtone content. Disregarding basses that can be adjusted to different scale lengths (different vibrating string lengths), the actual playing position that is used plays an important role. When you play D at the tenth fret on the low E string, it sounds more muffled than the same pitch played on the open D string.

The number of frets differs from bass to bass and depends on the scale length. We don't need to worry about pitches higher than a single ledger line C; the actual functional range of this instrument is primarily in its two lower octaves—between E 0 and E 2.

We should also mention the fretless electric bass. Like all instruments of this type, it is freely tunable and possesses a distinctive, individual sound. Over the course of this tutorial you will discover how to program this type of instrument in Sculpture.

There are three types of articulations that we will be discussing:

- Fingered: the strings are played with the alternating index and middle fingers.
- Picked: the strings are played with a pick.
- Thumbed/Slapped: the strings are either played with the (side of the) thumb on the fingerboard or plucked strongly with the fingers.

The vibration of the strings is captured by an electromagnetic pickup. When the string is vibrating, its steel core affects the magnetic field. The pickups are almost always found some distance to the side, nearer to the bridge and stop tailpiece. There are different pickup concepts for electric basses and often two or more pickups are combined to make the “sound”. Although we can't take these things into detailed consideration at this point, there is a rule of thumb that applies:

The further you move the pickup towards the middle of the string, the “bassier” the sound will be and the more “hollow” it will sound. The further you move the pickup towards the end of the string, the more the sound's overtone content will increase, becoming more dense and compact. The sound will have more mid-range frequencies or “buzz” and less bass. If the pickup is positioned at the very end of the string, the sound becomes very thin. We can find parallels to the actual playing position of a real string here: If you play more towards the middle of the string, you'll get a smooth, even, and powerful sound which contains limited harmonic denseness (overtones), from time to time. If the string is played at the bridge, the sound develops a nasal twang and features more “buzz” and more overtones.

Now to the body of the instrument, and its resonant properties. Almost all electric basses have a steel rod running through the neck, to strengthen it, and a body made of solid wood. This construction allows the strings to vibrate relatively freely (sustain), even though very little direct sound is generated. The pickups and the amplifier and speaker systems are responsible for the actual sound of the instrument.

The acoustic interaction between body, strings, and external sound sources is much less complex than with pure acoustic instruments.

The vibration of the strings is, of course, naturally hampered by several physical factors: the radius of motion of the string (antinode) is impeded by the left bridge or by the first fret that's pressed down upon (and the frets in between). This can lead to the development of overtones which can take the form of anything from a slight humming or buzzing to a strong scraping or scratching sound.

In addition, factors such as the material properties of the strings and the instrument, as well as the softness of your fingertips also serve to dampen the vibration of the string

Programming a Basic Bass

In the following section we will discuss the programming of a basic bass sound. This will serve as the foundation for the different bass sounds which we will be creating thereafter.

Heed the following steps to create the proper working environment for design of your own “homemade bass”:

- 1 Make sure the range from C 0 to C 3 is available on your keyboard by either; transposing your master keyboard, or using the *Transpose* function in Logic's Region Parameter box.

Note: You can, of course, transpose sounds within Sculpture but this isn't the best solution in this case. The reason is that sounds would not be compatible with MIDI Regions in which note number 60 as middle C is considered to be the measure of all things.

- 2 Choose the default setting within Sculpture.

In order to recreate the sound characteristics of a typical bass instrument:

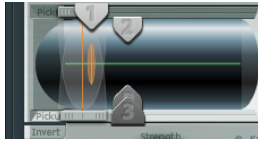
- 1 Set the *Attack* value of the Amplitude Envelope to its minimum value (0.00 ms). You'll find the *Attack* slider just to the right of the Material Pad.
- 2 Shorten the *Release* time of the Amplitude Envelope to a value between 4 and 5 ms. Play a key on your keyboard. The note should stop abruptly when you release the key and should be free of artefacts (a digital crackle or snap). If you encounter any artefacts during the course of this tutorial, please carefully increase the *Release* time.
- 3 Play some sustained notes in the range above E 0. These will die away (too) quickly. Correct this quick die-out with the *Media Loss* parameter by pulling the slider located to the left of the Material Pad almost all the way down to the bottom. For your information: the low E string on a high quality bass can sound for over a minute!

Our basic bass should simulate a fingered articulation, which means the sound is created by striking the strings with the fingers.

- 4 Choose the *Pick* entry from the Type menu of Object 1.

Don't be confused by the name of the Object Type: despite its name, this model is appropriate for simulating the playing of strings with your fingers.

Play some notes in the lower range. you'll note that the sound is very muffled, hollow, and distorted. Before we adjust further parameters in Object 1, we need to set the position of the *Pickup*.



This is accomplished in Sculpture's *Pickup* window located to the left of the Material Pad (see the GUI detail above). You'll find three trapezoidally shaped sliders, representing *Objects 1* to 3. Both of the transparent bell-shaped curves help you to visualize the position and width of *Pickup A* and *Pickup B*.

On electric basses, the pickups are found quite a way off to the side and near the bridge. We also assume that our bass only has a single pickup.

We can simulate the behavior of a single pickup by placing both *Pickups* at exactly the same position.

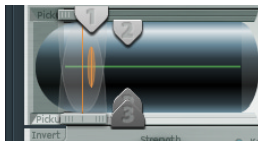
- 5 Keep an eye on the Help Tag, and drag *Pickup B* to the exact position of *Pickup A*. The two thin orange lines should overlap perfectly.

Note: Make sure that the *Invert* switch to the lower left of the Pickup window isn't turned on, as this would cause the *Pickups* to completely cancel each other out.

- 6 As a suitable value for our example, set both *Pickups* to 0.10.

It's now time to determine the playing position:

- 7 Grab the *Object 1* slider in the Pickup window and move it in a horizontal direction. Play the keyboard while doing so, to hear the changes it makes.
- 8 You'll quickly realize that you can only achieve a precise, crisp sound when relatively far away from the middle of the string. Move *Object 1* closer to the Pickup (position 0.15—see GUI detail below).



- 9 The low notes are still distorted. You can remedy this by adjusting the *Level* dial found to the right of the Amplitude Envelope. Set a value of -10 dB.

Although you can already recognize the sound of an electric bass, it doesn't sound "wiry" enough yet. Let us now direct our attention to the bass strings themselves.

- 1 Move the ball in the Material Pad up and down at the left edge. Pay attention to how the overtones react. Move the ball to the lower left hand corner. The sound should vaguely remind you of the sound of a low piano string. As the overtones sustain too long, the tone sounds somewhat unnatural.
- 2 Move the ball upwards until you hear an acceptable sound. As an example, we recommend the following position:



Note: In general, a splaying of the overtones in low wound strings is typical. You can recognize it by the slightly impure, metallic sound. This occurs because the partials (overtones) are not exact whole number multiples of the fundamental frequency, but rather they are shifted somewhat higher. An example of this effect in the real world of electro-acoustic instruments are the low strings on a Yamaha CP70. Although we don't want to take it that far, our bass model does need a small amount of this effect.

How to splay overtones in Sculpture:

- 1 Move the ball in the Material Pad gradually to the right. The sound takes on a more impure and bell-like character.
- 2 To realistically simulate the splaying of overtones, we recommend the following example setting:



The vibration of a bass string does not occur in a vacuum. The antinode of the string frequently encounters the natural, physical limitations of the instrument. This is heard as the typical buzzing and rattling that occurs when the strings touch the frets.

We can simulate these disturbing elements with Object 2:

- 1 Activate Object 2, and choose the *Bouncing* Type menu item. The sound should now vaguely remind you of a mandolin tremolo. This is way too strong an effect for the kind of sound we're after.
- 2 Move Object 2 all the way to the right (a value of 1.00).
- 3 Experiment with Object 2's parameters. A discrete and realistic result can be achieved with the following parameter values: *Strength*: 0.33, *Timbre*: -1.00 and *Variation*: -0.69.

Play some low notes, and you'll find that, once again, the overtones sustain a little too long, reminding us of the lowest strings on a piano.

We'll use Object 3 to dampen these overtones:

- 1 Activate *Object 3* and choose the *Damp* Type menu option.
- 2 Also move *Object 3* all the way to the right (value 1.00).
- 3 Set the *Strength* parameter to 0.18.

Note: Experiment with how the *Strength* parameter of *Object 3* interacts with the *Inner Loss* Material Pad parameter. The higher the *Inner Loss* value, the smaller the *Strength* value can be, and vice versa.

In order to more realistically replicate the different tonal ranges of the bass, we'll use Sculpture's scaling function.

Firstly, we need to activate the scaling function display:

- Click on the *Keyscale* button located at the bottom of the Material Pad. The key-scaling below C3 is displayed in green, the range above in light blue. The Material Pad with its key-scaling parameters activated:



Note: The most relevant performance range for basses is found exclusively below C3. For this reason, we'll be using the green sliders to set the actual timbre of the sound. The "primary" sliders found around the ring determine the timbre of the sound above C3. For the moment, we'll disregard the blue sliders (which control high key-scaling) and simply set them to the same positions as the main sliders.

Once activated, the key-scaling function is used to adjust the timbre of the sound, independent of pitch. To do this, we'll initially employ the *Resolution* parameter. This is normally used to set the balance between DSP load and sound quality. As the overtone spectrum is reduced at low *Resolution* values, this parameter can also be used to shape the sound.

To use the Resolution parameter to control the timbre, independent of pitch:

- 1 Play some notes at the higher end of the bass's range (around C2), then move the *Resolution* slider all the way to the right and then gradually back towards the left.
- 2 You can hear how the sound loses overtones, yet simultaneously becomes louder. At low *Resolution* values, an inharmonic metallic "rattling" is heard in the sound.
- 3 Increase the *Resolution* value until the metallic rattling disappears. We recommend setting the slider to the following position:



- 4 Play some notes in the bottom range (around E 0). You'll note that the sound is quite muffled and vintage-like. Move the green *Low Keyscale* slider (found below the main Resolution slider) all the way to the right; the low range should now sound a little more "wiry".

With most stringed instruments, the overtone content decreases as the pitch becomes higher. Strictly speaking, this is only true of open strings, and even then, in a limited sense. If the strings are fingered, the length of the string is shortened (especially in the high register) and the effect becomes more significant.

This is why we use the Inner Loss parameter to scale the overtone content, dependent on pitch:

- 1 Move the Material Pad ball above the words *Inner Loss*. Try to move the ball solely in a vertical direction, in order to maintain a constant *Stiffness* value.
- 2 Grab the green line next to the ball, and pull it towards the bottom until the small green diamond is located directly above the word *Steel*.

When playing, you'll recognize the smooth transition that takes place between the wiry, overtone-rich sound at the bottom end and the extremely dampened sound in the upper register. This (exaggerated) setting was chosen to clearly demonstrate the scaling principle in stringed instruments. In order to achieve an authentic sound and timbre, we recommend the following setting:



In basses in particular, low notes sustain far longer than the high notes. Sculpture allows you to authentically and convincingly simulate this behavior with the *Media Loss* parameter.

To use the Media Loss parameter to scale the fading phase of the note, dependent on pitch:

- 1 Play a few held notes in the range around C2 and above. You'll hear that these notes die out much too slowly. Move the *Media Loss* slider up until this range begins to fade out quickly enough. The downside is that the lower notes now die out too quickly!
- 2 Pull the green Media Loss Key Scale slider down until the fade-out phase of the lower range is sufficiently long enough.
- 3 Compare your results with these recommended values:



We've now completed our task for this section, and created a basic bass that's articulated with your fingers. Save this as "E-Bass Fingered Basic". In the following sections we'll be using this basic bass as a foundation for the construction of further basses.

Modifying the Frequency Spectrum of our Basic Bass

The scope for sound design, by altering the frequency spectrum of electromagnetic instruments, is far more flexible than that offered by acoustic instruments. In addition to the number of pickups, the choice of amplifier, the equalization setting within the amplifier and, last but not least, the physical properties of the speakers and their enclosing cabinet also play a major role.

The central features of our electric bass sound are complete, but the sound can be improved upon by paying close attention to some details. Here are a few general suggestions:

- Vary the position of the *Pickups*. Try placing each of them in different positions. This will cancel out certain frequencies, and others will be summed together.
- Try turning on the *Invert* switch, even though this effect is not typical for electric basses.
- What is typical for bass sounds is the placement of the Pickups in the outer left third of the string model. The farther you move them to the left, the thinner and more nasal the sound will become.
- Shifting Object 1 will have a similar effect. Try different combinations here as well.

The *Body EQ* is ideal for giving the bass sound that final, finishing touch. Our electric bass sound could be a little less smooth, and be a bit more precise in its attack phase. Bassists like to use the terms “drier” and “more bite” to describe this phenomenon.

To alter the frequency spectrum of our basic bass with the Body EQ:

- 1 Load the “E-Bass Fingered Basic” Setting.
- 2 Select the standard *Lo Mid Hi* Body EQ model.
- 3 Reduce the low bass frequencies by setting the *Low* dial to a value of -0.30 .
- 4 Boost the mid-range frequencies substantially by setting the *Mid* dial to a value of 0.50 . Grab the *Mid* frequency dial and drag it to the graphical display of the frequency spectrum (to the right), and pull downwards. Experiment a little, then choose a value of 0.26 .
- 5 You’ll probably find that the boosting of the low mid frequencies is a little too strong at this point, so please return the *Mid* value to 0.30 (see the GUI detail below).



- 6 The sound could stand to be a little more “wiry”, so set the *High* dial to a value of 0.30 .
- 7 To finish off, set the *Level* dial (to the right of the Amplitude Envelope) to a value of -3 dB. The sound is now as loud as possible, without the low notes distorting.

- 8 Save this sound Setting, as we'll need it for further modifications later. Please name it "E-Bass Fingered Basic EQ1".

Pick Bass

Our basic bass is played with the fingers. In the following example we will simulate playing the strings with a pick. We'll also use the *Pick* Object Type for this sound. We'll make use of the *Timbre* parameter to adjust the relationship between the speed and intensity at which the string is struck. We'll also make use of the *Variation* parameter to define the virtual material density or hardness of the pick.

If we imagine the fingers to be very soft picks, it makes sense to alter the *Pick* parameters so that a hard plastic pick is the outcome.

To simulate playing with a pick:

- 1 Please load the "E-Bass Fingered Basic" Setting.
- 2 Set the *Timbre* parameter of Object 1 to its maximum value of 1.00. You'll note that the attack is now stronger.
- 3 Try several different *Variation* settings to get a feel for the material qualities of the pick.

Note: Not all positions will deliver usable results for the entire range of the instrument.

- 4 You'll get a consistent, working setting for the two octaves above E 0 with the following parameter settings: Position 0.17 (pickup window), Strength 1.00 (maximum), Timbre 0.90, Variation 0.56.

When these settings are used, you'll find that the sound has become softer and very thin. In fact, it's somewhat reminiscent of a clavinet.

We'll compensate for this side effect with the Body EQ.

- 1 Activate the Body EQ and add a healthy portion of bottom end to the sound by setting the *Low* parameter to 0.60. *Mid* should be set to 0.33.
- 2 Set the *High* dial to -0.45 as the sound is now so bright that rolling off a few of the highs can't hurt.
- 3 Now bring the volume into line. If you adjust the *Level* dial to 2.5 dB, nothing should be distorting. If this isn't the case, try reducing some more of the bottom end with the *Low* dial.
- 4 Save this setting as "Pick Open Roundwound".

Damping

Playing with a pick is often combined with a damping technique that employs the ball of the thumb. The right hand, which also holds the pick, should physically lay on top of the strings at the bridge. This technique results in the sound having less overtone content but become more percussive and “punchy” at the same time. You can variably control the timbre of the sound through the angle and pressure of your hand while playing.

Object 3 will be used to emulate the virtual ball of the thumb in this example. The *Timbre* parameter determines the kind of damping that occurs. *Variation* dictates the length of the string section that is being dampened.

To achieve this effect, proceed as follows:

- 1 Set the Object 3 Type to *Damp*.
- 2 Set Object 3's *Strength* parameter to 0.50.
- 3 Move Object 3 a little bit to the right in the Pickup window (to position 0.95) to simulate the width and position of the ball of the thumb lying on the bridge.
- 4 Set *Timbre* to its minimum value (–1.00) to achieve a very soft damping effect.
- 5 Set the *Variation* parameter to its maximum value of 1.00.

You'll note a metallic ringing that occurs during the attack phase still can be heard in the octave above E0.

To suppress it:

Note: Move the small green diamond on the Material Pad to a position directly under the ball. In doing so, you've just increased the Inner Loss value for the low key range.

Note: In order to place the diamond exactly under the ball, you can also click it while pressing the Option key.

- 6 Save this setting as “Pick Bass Half muted”.

Harmonics

Harmonics are single partials (overtones) of the overall sound. They can be heard by damping certain points along the string. This is done by lightly laying the fingers of the left hand (assuming a right-handed bass player) on the string (not pressing down) before the note is articulated. The first overtone, the octave, is achieved by placing your finger at the exact middle of the string, in effect separating the string into two halves. The next overtone is the fifth above the octave and the position of your finger should divide the string into a ratio of one-third to two-thirds. The next highest overtone separates the string into proportions of one-quarter to three-quarters and so on.

To simulate fingers lightly touching the strings:

- 1 Object 3 is used as a damper. Select the *Damp* Type.
- 2 Adjust Object 3's *Timbre* parameter to its maximum value of 1.00.
- 3 *Variation* must be set to its initial value of 0.00. Simply click on the slider while holding down the Option key to do so.
- 4 Move Object 3 to the exact middle (0.50) of the Pickup window. Play the keyboard, and you'll hear the first overtone as a harmonic.
- 5 While playing, very slowly move Object 3 towards the left of the Pickup window. In doing this, you are effectively "scrolling" through the overtone series, so to speak.
- 6 Save this setting as "Flageolet Xmple."

Vintage Flat Wound Pick Bass

Now, in just a few easy steps, we would like to transform our pick bass into a vintage pick bass with flat wound strings. This bass sound is typical for the funk and soul music of the 70's, but you'll also find it in many easy listening arrangements.

Proceed as follows:

- 1 First load the "Pick Bass Half muted" Setting.
- 2 Drag the Material Pad ball upwards and the sound becomes more muffled. Here are the values we recommend:



- 3 Increase the Object 3 *Strength* parameter to 0.70. The result is a muted pick bass with flat wound strings.

Tip: If you turn Object 3 off, you'll hear a sound that is reminiscent of a 1970's Fender Precision Bass.

- 4 Save this Setting as "Flatwound Pick Damped".

To get a nice percussive sound a la "Bert Kaempfert," proceed as follows:

- 1 Turn Object 3 back on.
- 2 Move both *Pickups* a little to the left (position 0.08).
- 3 Our virtual pick (Object 1) can also be moved a little further to the outside (position 0.10).
- 4 We can add the icing to the cake with the Body EQ. Turn the *Low* dial to its maximum value (1.00).



- 5 To remove the smacking in the attack phase, we'll use the graphical display to choose a value of 0.48 for the Body EQ *Mid* frequency, then use the dial to increase this value to 0.51. Option-click on the Body EQ *High* parameter to set it to a value of 0.00.
- 6 Save this setting as "Easy Listening Pick Bass".

Slap Bass

We're actually dealing with two different articulations here. The low notes originate when the thumb literally slaps the strings on the upper part of the fingerboard. The high notes are produced when the strings are strongly plucked or "popped" with the fingers. This is achieved by hooking a finger under the string, pulling it away from the instrument and then allowing it to slap back onto the fingerboard. In conjunction, these articulation methods make up the typically aggressive and overtone-rich "slap bass" sound.

To simulate a slap bass:

- 1 Load the "E-Bass Fingered Basic EQ1" Setting.
- 2 Turn off the Body EQ.
- 3 Also turn off Object 2 and Object 3, for now.

As the basic sound of a slap bass is brighter than a standard fingered bass, we need to adjust some Material Pad settings:

- 4 Return the *Low Key Scale* parameter to its initial value by clicking on the little green triangle, while holding down the Option key.

- 5 Move the ball down a little, and the sound becomes more wiry. The ball should now be located directly above the word “Steel” on the horizontal axis.



From the models at our disposal, *Strike* is the most suitable for simulating a thumb physically striking the strings from above. This model is not, however, as appropriate for the slapped (popped) strings. It makes the most sense to choose the *Pick* model for this purpose.

- 6 To be safe, turn the *Level* dial to -25 dB.
- 7 Select the *Pick* model for Object 1.
- 8 Move Object 1 to position 0.90 in the Pickup window. This position corresponds to a playing position above or on the fingerboard.

Note: Given its universal concept, *Sculpture* will not react exactly like a bass, where one would tend to play in the middle of the string on the upper part of the fingerboard. Try moving Object 1 to this position and see how it sounds. You'll find that the sound is little too smooth.

Setting the parameters for Object 1:

- 1 Set *Timbre* to a value of 0.38; this corresponds to a rapid attack.
- 2 Set the *Strength* parameter to 0.53.
- 3 Set the *Variation* parameter to -0.69 ; this defines the softer “material” that constitutes the fleshy part on the side of your slapping thumb.

You're probably familiar with the sound of low notes when played with your thumb. What's missing, thus far, is the typical bright rattling that is created when the string strikes the fingerboard. We'll use Object 2 to this end, and select the *Bound* Type menu option. *Bound* limits the antinode of the string in exactly the same way as the fingerboard on a real electric bass.

Let's review the functions of these parameters: *Timbre* determines the angle of the obstacle to the string while *Variation* defines the type and degree of reflection.

Let's adjust Object 2's parameters to the following settings:

- 1 Set *Timbre* to 0.39. This corresponds to a fingerboard that runs almost parallel to the string.
- 2 Set the *Strength* parameter to 0.33.

Note: Try some higher values as well. You'll see that the sound becomes softer and softer until it's completely dampened by the obstacle.

- 3 An appropriate value for *Variation* is 0.64. Despite the overtone-rich reflection, the string can still vibrate freely.

Note: Try some negative values: you'll see that the reflections can no longer develop in an unhindered fashion.

- 4 Set the *Level* dial to -3 dB; the *Bound* obstacle has made the sound softer.
- 5 The sound is still too smooth for a real slap bass; so let us direct our attention to the Body EQ again. Switch it on, and adjust the parameters to the following settings: Low = 0.25, Mid Frequency = 0.59, Mid = 0.43, High = 0.51.
- 6 Save this sound as "Slap Bass Basic#1".

Fretless Bass

With the exception of shared playing techniques, the fretless bass differs from a "normal" bass through its buzzing, singing sound. As the frets on the fingerboard of a standard bass function as a collection of "mini-bridges" and allow the string to vibrate in an unobstructed fashion, the direct "collision" of the string's antinode with the fingerboard on a fretless bass is responsible for its "typical" sound. The string length on a fretless bass is markedly shorter than that of an acoustic double bass. The upshot of this is that a "controlled buzzing" is produced, even when played with a weak attack. This "buzzing" can be consistently reproduced in the high register, even on fretless basses that have very short string lengths. The use of the comparatively soft tip of your finger, instead of a hard, metallic fret, to divide or shorten the string also plays a role.

This is how you program a fretless bass:

- 1 Load the "E-Bass Fingered Basic EQ1" Setting.
- 2 Turn Object 3 off. We'll come back to it later.
- 3 Choose the Object 2 *Disturb* Type menu option.

Tip: The *Disturb* model functions in the following way: the *Timbre* parameter determines how far the string is deflected from its resting position by the obstacle. Positive values precipitate no deflection of the vibration from its resting position. *Variation* defines the length of the string section that is "disturbed": positive values correspond to a longer section of string, negative values to a correspondingly shorter string section.

- 4 Adjust Object 2's parameters to the following values: Strength 0.14, Timbre -0.05, Variation -1.00.
- 5 Object 2's Pickup position remains at the far right; please enter a value of 0.99. You'll note that the range between C2 and C3 already sounds quite acceptable, but the buzzing in the lower notes is still too strong. It is somewhat sitar-like, so keep this disturb model in mind when it comes to creating a "home-spun" sitar.
- 6 Try different settings for the *Strength* parameter for both the higher and lower playing ranges. You'll see that, at best, only a compromise is possible. The buzzing is either too loud in the low range or not present enough in the high range.

Obviously, we need to scale the effect over the relevant tonal range. Unlike the parameters for the string, Objects 1–3 don't have a directly addressable key scaling function. We've got to be a little clever here. Both LFOs offer a key scaling function. As we don't want the buzzing to be modulated by a periodic oscillation, we need to reduce the LFO speed to "infinitely slow" or 0. In this way, we can deactivate the LFO itself but still use its modulation matrix.

- 7 Activate LFO2 by clicking on the LFO2 button at the bottom left, and set the *Rate* dial to a value of 0.00 Hz.
- 8 Click on the button marked "1" (next to the *Rate* slider, to the upper right) to activate the first modulation target.



- 9 Choose *Object2 Strength* as the *Target* parameter.
- 10 Select the *KeyScale* entry in the *via* column.
- 11 Move the lower slider labeled *Amt* (amount) to the right while you are playing. You'll quickly realize that the "singing" buzzing fades out in the lower range, while gradually being retained as you move towards C3. Set the slider to a value of 0.15. The buzzing will now be far more moderate in the low range.



- 12 Switch Object 3 back on. Set *Timbre* to its minimum value (-1.00) and *Variation* to its maximum value (1.00). Object 3 should be positioned all the way to the right, at a value of 1.00.

- 13 Vary the Object 3 *Strength* parameter. You'll discover that the overtone content of the buzzing can be controlled very effectively. A *Strength* value of 0.25 is recommended here.
- 14 Save this Setting as "Fretless Roundwound#1".

Modulation and Detuning

Detuning and ensemble effects are normally achieved using a modulation effect or by combining doubling and detuning. When using a fretless bass for a solo part, a broad chorus effect adds a nice touch.

As Sculpture can only synthesize one note at a time at any certain pitch, we can't work with doubling. There are, however, alternatives for bringing movement and life into the sound. Almost all of the *Type* parameters of the different Objects can be modulated by LFOs, resulting in a vast number of possible combinations.

Emulating a chorus effect by modulating the Pickup positions:

- 1 Make sure that you've got the "Fretless Roundwound#1" Setting loaded.
- 2 Adjust the position of *Pickup B* to 0.20.
- 3 The *Stereo* control element is located to the left of the *Level* dial. Click-hold on the (*Stereo*) *Pickup* semicircle, and move the mouse in an upwards direction. Both of the light blue dots move downwards towards the letters L and R. You can hear how the stereo breadth of the fretless sound has increased. *Pickup A* is sent out on the right channel, while *Pickup B* occupies the left channel.



Note: Although only modern basses offer such stereophonic features, it's still fun to process conventional sounds (such as those created in the previous examples) with this effect. Note that not all pickup positions are monophonic compatible; you can check this by returning the *Stereo Pickup* setting to mono (by clicking on the *Pickup* semicircle while depressing the Option key).

Now we need to make the pickups move:

- 4 Select LFO1.
- 5 Activate the first modulation target by clicking on the "1" button (next to the *Rate* slider, to the upper right).
- 6 Choose *Pickup Position A-B* as the modulation target.
- 7 Set the *Rate* dial to 1.00 Hz.

- 8 In order to hear the effect, the modulation intensity (amount) has to be set. Familiarize yourself with this effect by moving the slider labeled *Amt* (amount) gradually to the right. Set it to a final value of 0.15, a moderate rate that doesn't "wobble" too much.
- 9 Save this Setting as "Fretless Chorus Dry".

Tip: At the maximum stereo breadth, effects based on detuning are not as prominent, especially when the "beats" heard in the sound result from signal differences between the left and right channels. This is only valid to a certain degree because the motion of the pickup doesn't create a true chorus or harmonizer effect. Try it out and see what happens when the stereo breadth is reduced a little. Also test other modulation targets: *Pickup Pos A+B*, *Pickup Pan A+B*, *Pickup Pan A-B*, and *String Stiffness* are recommended. Have fun!

Reverb and Reflections

As a rule, basses are mixed without effects ("dry") and you probably haven't missed any reverb or delay effects in our examples, so far. Having said this, a little bit of reverb can be quite appealing on a fretless bass, when it's used as a solo instrument. We can use Sculpture's *Stereo Delay* section to emulate this.

In order to create an unobtrusive atmospheric space, proceed as follows:

- 1 Load the "Fretless Chorus Dry" Setting.
- 2 Turn on the stereo delay section by clicking on the *Stereo Delay* button.
- 3 Set the *Input Balance* slider to 1.00.
- 4 Switch off the tempo synchronization of the delay by deactivating the small *Sync* button (found directly to the right of the *Delay Base Time* slider).



- 5 Set the *Delay Base Time* slider to 90 ms.
- 6 Set the *Crossfeed* dial to 0.30.

The individual reflections are still too brash. In order to make the effect more discrete and unobtrusive, we are going to adjust the frequency spectrum and amplitude of the reflections. let's start with the frequency spectrum:

- 7 Set *LoCut* to 200 Hz and *HiCut* to 1000 Hz (in the stereo delay section).

The *LoCut* at 200 Hz excludes the low frequencies in the reflections, thus avoiding a "muddy" sound. The comparatively drastic cut to the highs with the *HiCut* parameter blurs the individual reflections, thereby creating the impression of a small room with soft surfaces.

- 8 Reduce the total level of the effect by setting the *Wet Level* dial to a value of 25%.
- 9 Save this Setting as “Fretless Chorus+Ambience”.

This example shows that the stereo delay section can be used as a substitute reverb for small spaces. For sophisticated reverb effects, it's best to process Sculpture's output with one of Logic's reverb plug-ins.

Creating a “drowned” in delay effect:

- 1 Reload the “Fretless Chorus Dry” Setting.
- 2 Switch the stereo delay section on.
- 3 Move the *Input Balance* slider all the way to the right (1.00).
- 4 Set the *Delay Base Time* value to “1/4t” (quarter note triplet).
- 5 Set the *Feedback* dial to a value of 0.20.
- 6 Adjust the *Crossfeed* dial to a value of 0.30.
- 7 Set the *LoCut* to 200 Hz and the *HiCut* to 1600 Hz.
- 8 Now adjust the overall level of the effect; we recommend setting the *Wet Level* dial to a value of 45%.
- 9 Now vary the stereo position and rhythmical structure of the delay by moving the small light blue diamond around the *Stereo Delay Pad*.
- 10 Save this Setting as “Fretless Chorus+Wet Delay”.

Synthesized Sounds

In the preceding sections, you learned how to program natural bass sounds with Sculpture: by authentically reproducing the real physical interaction that occurs between a string and the exciting agent that acts upon it. While producing such lifelike models is undoubtedly a forte of Sculpture's architecture, its sonic capabilities extend to the creation of very different sounds as well.

Sculpture contains a number of functions that you can use to create new and novel synthesized sounds. This includes the Morph Pad, which can be automated, as well as recordable and programmable Envelopes that can be used in a rhythmic context.

Such features are usually unnecessary when reproducing natural bass sounds, as no electric bass that exists can alter the tonal characteristics of the string during the decay phase of a note—perhaps from wood to metal—and rhythmically synchronize this change to the tempo of the song. These functions are very useful, however, when creating sustained, atmospheric sounds where slow and interesting modulations help it to “come alive.”

In the following sections, we'll be demonstrating Sculpture's ability to create somewhat spacy and less "organic" sounds, using several pad patches as examples. After having studied the modeling of bass sounds in the preceding sections, we'll now introduce you to a totally different group of sounds. Provided that you're willing to supply the necessary level of curiosity and time investment for experimentation, you'll discover a wide variety of interesting and animated sounds.

Within the framework of these short experiments, it's of course impossible to comprehensively cover all of Sculpture's possibilities. We'd like to expressly encourage you to experiment with the suggested settings and closely observe the results of the changes you make. In this way you can learn a lot about the instrument, and hopefully be inspired to create new sounds and variations.

Note: You will find the settings for these tutorials in the Tutorial Settings folder in the settings menu (in the head of the Sculpture Plug-in window).

The Sustained Sound

First, please load Sculpture's default setting again; the very simple sound consisting of a plucked string that vibrates and fades away. Obviously, we'll need to edit this sound drastically as we need a sustained or extended sound for pads, rather than one that dies away.

Have a look at the three Objects: you can see that only Object 1 is active, and acts on the string with an *Impulse*. As in the pick example in the bass section, the string is briefly excited when the note is played, and then the sound decays. For a sustained pad sound we require an exciting agent that constantly acts upon the string; the appropriate Object Types are *Bow* or *Bow wide* (the string is played with either short or long, extended bow strokes), *Noise* (excited by a random noise signal) or *Blow* (excited by being blown—much like a clarinet or flute).

Please test the abovementioned Object Types one after the other, and move the Object 1 (Pickup) slider, responsible for the exact position of the exciting agent, up and down the string while you're playing. You will come to two conclusions. First, the sound is now sustained for as long as you hold the key down. Secondly, shifting the Object 1 slider, with the *Bow* Type selected, results in the most pronounced sonic changes. This setting promises the most rewarding possibilities for varying the sound, and that's why we've decided to use this Type.



Recording an Envelope

The sonic variations created by the *Bow* Type are very appealing when the virtual bow stroke is moved along the string. We'd like to control this movement through the use of an Envelope, thus creating the foundation of our pad sound.

It makes more sense (and it is more convenient) to record the Envelope, rather than programming it, even if the latter is easily achieved with the graphic display.

Proceed as follows to record an envelope:

- 1 Move the Object 1 slider all the way to the left. Starting from this position (where it only generates an overtone-rich scratch), we want to start animating it by using the Envelope.
- 2 Locate the Envelope section in the lower right corner of the Sculpture window. Choose the first of the two Envelopes by clicking on the *Envelope 1* switch, if necessary. In the left part of the Envelope section, you can see two routing possibilities that allow you to assign a modulation target to the Envelope.
- 3 Activate the first routing link by clicking on the "1" button, and choose *Object1 Position* as the modulation target in the *Target* menu. Set the modulation intensity to its maximum value by moving the horizontal slider all the way to the right.



The Envelope can now be recorded. We assume that your MIDI keyboard has a modulation wheel which outputs the corresponding MIDI controller message (CC number 1) and that option *1 ModWh* is selected for control of Envelope 1 (*CtrlEnv 1*) in the dark bottom edge of the Sculpture window.

- 4 Click on the button labeled *R*, located in the upper right corner of the Envelope section below “Record Trigger”, to prepare the Envelope for recording. Select the *Note+Ctrl* option for the recording. This option specifies that the recording of the controller messages from the modulation wheel will begin the instant the first note is played.



- 5 Play a note when you want to start the recording, and move the modulation wheel slowly upwards while keeping the key on the keyboard depressed. Pay attention to the sound variations you create while moving the modulation wheel.
- 6 At the end of the recording, return the wheel to its initial starting position and, after you’ve released the depressed note, click on the “R” button to deactivate the recording mode.

You can now see the curve you recorded in the graphic display. You’ll note that the curve arches exclusively above the zero axis—this is no surprise as the modulation wheel only sends unipolar values, which means between zero and up to a positive maximum value. As we already moved the Object 1 slider all the way to the left end of the string, it can only be shifted all the way to the right by the Envelope when the maximum modulation intensity is reached.

Play a note, or better yet, an entire chord and listen to the modulation you recorded. If you’re not satisfied, you can, of course, repeat the procedure described above as often as you’d like. The maximum available recording time is 40 seconds. It is therefore possible, if you’d like, to control parameters and create modulations that extend far beyond the well-worn path of a simple ADSR envelope progression.

Note: By moving the junction points, you can edit the shape of the envelope when the need arises. By clicking in the empty part of the graphic display and moving the mouse to the right or left, you can zoom in and out of the display. When you activate the small *Sync* button, the junction points snap to a rhythmic grid (further information about these parameters and their functions can be found on page 397).

Select *Loop Alternate* in the *Sustain mode* menu below the Envelope graphic display. As the sustain point is found at the end of the Envelope, the Envelope repeatedly travels from the beginning to the end and backwards from the end to the beginning, creating a continuous flux within the sound.



Summing up: we now have a rudimentary, but appealing and organic-sounding pad which we will use as the foundation for further shaping and refinement. Save this rough version of the sound as a basis for further experiments (Preset 0001 raw pad).

Increasing Stereo Breadth and Chorus

The next thing we'll do is give our very dry sounding pad a little more stereo breadth and chorus effect. We'll use the trick we discussed while creating bass sounds. Namely, modulating the Pickup positions, and assigning them to the left and right channels.

Here's a quick description of the process:

- 1 Click-hold on the *Pickup* semicircle in the *Stereo* control element, and move the mouse upwards to separate the stereo pan positions of the *Pickups*. Both of the light blue dots should come to rest near the line that separates both semicircles.



- 2 Activate both of the modulation links in LFO1 by clicking on the buttons labeled 1 and 2. For the first link, modulate the position of Pickup A (*Target: PickA Pos*) with a small positive value and with the second link, the position of pickup B (*Target: PickB Pos*) with a small negative value (intensity at ± 0.15). Reduce the *Rate* of the LFO to about 0.3 Hz.



You will hear a pleasant “beating” or chorus effect in the sound, which makes it broader and more full, alleviating the unpleasant, dry character (preset 0002 lfo>pick pad). Another unpleasant aspect is that the sound is too strong in the mid frequency range and could use some equalization. We’ll use the Body EQ to correct this.

Activate the Body EQ, and experiment with the *Lo Mid Hi* model (which is the standard setting). Our favorite setting is when *Mid* is reduced 0.5 and *Mid Frequency* is set to 0.37 but this, of course, is a matter of taste...



Now we want to give the pad a little depth by using the Stereo Delay section. Activate the Stereo Delay, set the *Delay Time* to $1/4$ and adjust the *Crossfeed* dial to 30%. The pad now has a pleasant and unobtrusive ambience; you can leave the other Stereo Delay parameters at their original values (preset 0003 eqfx pad).



Finally, we want to optimize the sound so that it is a little more animated. This is not-hing too spectacular, so we’ll employ the Jitter modulators for the job. The Jitter modulators are basically LFOs that use a random waveform.

Using the jitter modulators to make the sound become more lively:

- 1 Activate the display for both of the Jitter modulators by clicking on the *Jitter* switch below the LFO.
- 2 Turn on the first link in *Jitter 1* by clicking on the button labeled *1*, and choose *Object1 Timbre* in the target menu.

- 3 Adjust the *Intensity* to -0.40 with the slider below the *Target* menu, and reduce the *Rate* parameter to 1 Hz. There should be subtle inconsistencies in the pressure applied by the bow to the string. To better recognize the effect, temporarily increase the *Intensity* level.



We'll use the second Jitter modulator for random position deviations with the modulation target *Pickup Pos A+B* (pickup position A and B).

- 4 Activate *Jitter 2* and choose the *Pickup Pos A+B* option in the *Target* menu.
- 5 Set the slider under the *Target* menu to an *Intensity* of about 0.2, and adjust the *Rate* dial to 1.5 Hz. As you increase the *Intensity*, the sound develops a distinct clinking or rattling—use this effect to taste.

We now have a satisfactory pad sound, so we'll leave it at that, even though a few *Sculpture* features such as; the *Filter* and the *Waveshaper* lie idle, not to mention the two additional *Objects*—but it's smart to quit while you're ahead... We've saved an especially exciting function, *morphing*, for the end and we'd like to use it to bend and twist our pad sound a bit.

Morphing

In the middle of the lower part of *Sculpture*'s window you can see the *Morph Pad*. In each corner, it can contain a different setting for a diverse number of parameters and by moving the red ball—which can be seen in the center of the *Morph Pad*—you can crossfade between these settings and “morph” the sound.

Choose the *Paste to all Points* command in the context menu (accessible via a Control-click on the *Morph Pad*) to copy the current patch setting into all four corners of the *Morph Pad*. If you move the ball in the *Morph Pad* now, you won't hear any changes in the sound because the settings (in each corner) are all the same. Not for long...

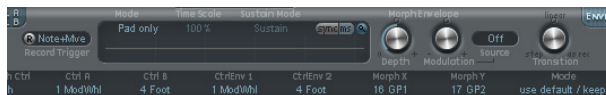


To vary the sound with the Morph Pad:

- 1 When you move the ball to one corner, the corresponding “partial” sound is selected; you can recognize this by the gray blue arches that light up in the graphic display. Choose each of the four corners one after the other, and vary the sound by altering several parameters directly in Sculpture’s GUI.
- 2 Use of the Material Pad allows you to achieve an especially noticeable variation: carefully move the ball around in the Material Pad, and try to find a position where our pad sound takes on a new and interesting character. Also try the extreme corners, for example.



As soon as you’ve chosen different settings for the Morph Pad corners (A to D), moving the morph ball will create marked sound variations—even though the intermediate stages will not all exhibit a tonal character. You can automate the morphing process by assigning two MIDI controllers to the *MorphX* and *MorphY* menus at Sculpture’s bottom edge. You can also automate the Morph Pad using a recorded Envelope—you’ll find further information about this on page 395.



Surrounding the Morph Pad, you’ll find a randomizing function which randomly varies sounds to a chosen intensity level (or amount of randomization). This is especially useful for subtle changes to natural sounds, but it can also provide for rewarding variations to synthesized sounds as well.

To use the randomizing function:

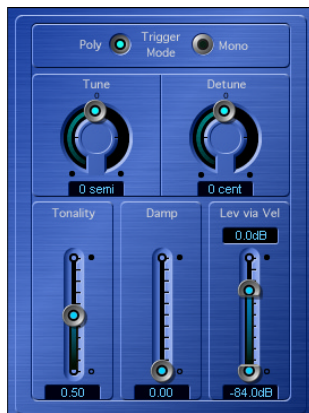
- 1 On the left side of the Morph Pad, choose the number of corners that are to be varied by selecting one of the “cubes”.
- 2 Use the slider to the right of the Morph Pad to adjust the *Intensity* of the random deviations.

- 3 Perform the randomization by clicking on the *Rnd* button. When you next move the morph ball, you'll hear the variations you just created.



We have now reached the end of our programming tutorial. By demonstrating how to create basic sounds, detailed emulations of various electric bass sounds and explaining how to approach the generation of synthesized sounds, we hope to have given you a few insights into the interplay of Sculpture's functions and parameters. We also hope that you've gained some inspiration and insights for programming your own sounds—we wish you much fun and success while creating your own sonic sculptures!

KlopfGeist is an instrument that is optimized to provide a metronome click in Logic.



KlopfGeist is inserted on Audio Instrument channel 64 by default. Logic automatically assigns this channel to the Metronome Object, making KlopfGeist the synthesizer responsible for the metronome click.

Theoretically, any other Logic or third-party instrument could be used as a metronome sound source on Audio Instrument channel 64. Similarly, KlopfGeist can be inserted on any other Audio Instrument channel for use as an instrument.

A look at KlopfGeist's Plug-in window will, however, clearly show that it is a synthesizer designed to create the metronome's clicking sound.

KlopfGeist can operate as a monophonic or polyphonic (4 voice) instrument, as determined by the *Trigger Mode* radio buttons. There are two tuning parameters; *Tune* for coarse tuning in semitone steps, and one for fine tuning (*Detune*) in cents. The *Tonality* parameter changes the sound of KlopfGeist from a short click to a pitched percussion sound—similar to a Wood Block or Claves. *Damp* controls the release time. The shortest release time is reached when *Damp* is at its maximum (1.00) value. *Level Via Vel* determines the velocity sensitivity of KlopfGeist. It is a two part slider; the upper half of the slider determines the volume for maximum velocity, the lower half for minimum velocity. By clicking and dragging in the area between the two slider segments, you can move both simultaneously.

The overall level of the virtual click sound is determined by the Audio Instrument channel 64 *Volume* fader.

Note: A Klopfgeist (knocking ghost) is a ghostly little fellow, usually German, who restricts himself to producing knocking and tapping sounds, unlike his big brother, the Poltergeist. Audio Instrument channel 64 is fitted with a Ghost Buster facility—the button labeled *M*.

This chapter introduces Logic's EVB3 virtual Hammond organ.

An electro-mechanical Hammond organ, with tonewheels and a Leslie sound cabinet, cannot be replaced by anything other than an electro-mechanical Hammond organ, with tonewheels and a Leslie sound cabinet! Having said that: The EVB3 offers a truly portable alternative to this classic pop, rock, and jazz instrument.

Concepts and Function

The EVB3 software instrument mimics the sound, and use, of the Hammond B3 and Leslie sound cabinet. Being a software instrument, the EVB3 is not only lighter and more portable than its big, heavy counterpart, it's also much easier to integrate into Logic's production environment. There's no need for any cabling or microphone setup, for live-performances. The same applies to the recording of realistic organ tracks, with the added bonus of full editing facilities at any stage of production.

The EVB3 simulates an organ with two manuals (keyboards) and a pedalboard—each of which can have its own registration (sound settings). The sound generation process is not limited to the mere additive synthesis of sine choirs. Rather, it fully simulates the tone wheel generators of an electro-mechanical Hammond organ, down to the smallest detail. This includes certain charming flaws, such as the Hammond's enormous level of crosstalk and the scratchiness of the key contacts. You may adjust the intensity of these peculiarities to meet your tastes. This flexibility allows for flawlessly clean, through to dirty and raunchy sounds, and everything in-between. Further quirks of the original were the robbing of sine choir voices (multiplexing) and the specific repetition behavior. No need to tell you that the EVB3 emulates this too!

The EVB3 simulates three different types of Leslie sound cabinets with rotating speakers, with and without deflectors. Beyond this, three tube overdrives—with different tonal characteristics, an equalizer, a wah wah and a reverberation effect are also incorporated. You may freely define the signal flow of these effects as you wish. You can also set the stereo intensity of the microphone position as desired.

If you're familiar with the original B3, you'll remember the inverted (black) keys of the lowest octave on each manual. These inverted keys are switches that recall preset registrations (a preset of your drawbar settings). This feature is emulated by the EVB3 as well, but has been improved significantly, given that you won't need a screwdriver to change the registration settings of your presets. A morphing feature allows seamless crossfades between two registrations, without the need for an external fader box. The EVB3 can be played with two manuals and a MIDI pedalboard, if you wish (and own these devices). It also offers functions which allow you to play all registers (Upper, Lower, and Pedal) with a single-manual master keyboard.

MIDI Setup

If you want to fully exploit all features of the organ, you will need a MIDI (bass) Pedal unit, and two 73-key MIDI keyboards. As the EVB3 also emulates the B3's preset keys, the lowest octave of attached MIDI keyboards can switch the EVB3 registrations, just like the original B3. Please read "Playing Both Manuals and the Pedals Live" on page 456 for more information.

The EVB3 can, of course, be played with single-manual keyboards with the standard 61 keys (5 octaves C to C). Please read "Keyboard Split" on page 458 for more information.

Playing Both Manuals and the Pedals Live

The EVB3 receives the notes for the Upper and Lower manuals, and the Pedalboard, on three subsequent MIDI channels.

Note: The MIDI channel (*MIDI Cha*) of the Audio Instrument must be set to *All*, and *Keyboard Mode* must be set to *Multi*.

The default Upper-manual receive channel is MIDI channel 1, channel 2 for the Lower manual, and channel 3 for the Pedal registers. If your master keyboard sends MIDI notes on channel 1, it will play the Upper manual, if it sends on channel 2, it will play the Lower manual, and if it sends on channel 3, it will play the Pedal register.

You may use any of your MIDI interface (Unitor 8 MkII or AMT 8, for example) inputs for your master keyboard(s) and/or Pedalboard. You can also use a single-manual master keyboard—with different keyboard zones or a keyboard split feature—that sends data on different MIDI channels. Regardless of the input device(s) used, the only relevant factor is the MIDI send channel!

Logic records the channel information of incoming notes. With most other MIDI and software instruments, this information is not used at all. This is due to Logic's MIDI channel setting (in the instrument parameters), which has priority, and overrides the original channel information. This can be circumvented by setting *MIDI Cha = All*, which is recommended for the EVB3. This will force the original channel information to be used. This enables you to make use of the two manuals, and Pedal register of the organ directly and simultaneously—when playing live.

Note: Please read the users manual of your master keyboard, to learn how to set its MIDI transmission channel, often abbreviated as TX Channel.

Changing MIDI Channels

You also can set the EVB3 to receive on MIDI channels other than 1 for *Upper*, 2 for *Lower* and 3 for *Pedal*. This is done with the *Basic MIDI Ch* parameter in the *General* section. The receive channel number for *Lower* is always one (channel) number higher than the channel assigned to *Upper* with the *Basic MIDI Ch* parameter. The *Pedal* register receive channel is always two (channel numbers) higher than the *Basic MIDI Ch* selected for *Upper*.

Note: The *Basic MIDI Ch* parameter is only available in the Controls view. Click-hold the *Editor* pull-down menu at the top of the Plug-in window to access.

Note: *Basic MIDI Ch* only works if *Keyboard Mode* is set to *Multi*.

Note: When the *Basic MIDI Ch* is set to 16, the *Lower* manual receives on channel 1, and the *Pedal* register on channel 2. When the base *MIDI Ch* is set to 15, *Lower* receives on channel 16, and the *Pedal* register receives on channel 1.

Note: Selection of different MIDI channels may be necessary in a live performance situation, particularly if you need to change the MIDI transmission channel of your master keyboard, in order to play other sound generators.

Keyboard Ranges of the Upper and Lower Manual

The lowest playable MIDI note is 36 (C1). The range of the preset keys is note # 24 to 35 (C0 to B0).

Note: 128 notes are defined in the MIDI specification, but even the largest master keyboards are only equipped with 88 keys, just like a concert grand piano.

One example: If your master keyboard ranges from C to c (5 octaves—61 keys), and the *Region* and *Object* parameters of Logic are set to zero (*Transpose. = 0*), you can play the entire keyboard range—every possible note of the EVB3. The preset (registration) keys are positioned one octave lower. If you set *Transpose. = -12*, you can use the lowest octave to switch between presets.

Keyboard Split

The EVB3 can also be played perfectly with a single MIDI keyboard (one manual) that is only capable of transmitting on one MIDI send channel. You can split the keyboard in order to play *Upper*, *Lower*, and *Pedal* sounds on different keyboard zones.

In the parameter field in the bottom center of the GUI, set *Keyboard Mode* = *Split*.



Set the keyboard zones with the *UL Split* and *LP Split* parameters, in conjunction with the *Set* buttons. The abbreviations are for: *Upper/Lower* and *Lower/Pedal*. To do so, simply click on the appropriate *Set* button (it will turn orange), and press the desired note on your MIDI keyboard.

Note: If the *LP Split* is set to a value above the *UL Split*, the other splitpoint is moved (and vice versa).

Note: If you select the same value for both splitpoints, the Lower manual is not heard/active.

Note: In order to use preset switching in conjunction with *Keyboard Mode Split*, you might need to adjust the transpositions for the Upper Manual (*Trans UM*), Lower Manual (*Trans LM*) and Pedal (*Trans Ped*).

Transposition (Octave Range)

You can individually transpose the Upper (*Trans UM*), Lower (*Trans LM*) and Pedal (*Trans Ped*) registers up/down one or two octaves, independent of the global transpose functions. Transposition is useful for setting the various registers to a particular range, matching your needs. This facility is of particular importance when using split mode.

Note: As with Logic's global transposition functions (which can be set in semitones), these manual-specific transpositions have no impact on the preset keys. Also see "Preset Keys and Morphing" on page 464.

MIDI Mode

This parameter allows you to define how the drawbar settings will respond to remote MIDI control change messages. Normally, you won't need to change anything here. If you own a MIDI drawbar organ, however, you'll probably want to use its hardware drawbars to control the EVB3. Most hardware drawbar organs utilize an independent control change number for every drawbar. Some models allow you to freely define these control change numbers. Lists of control change number assignments for the following *MIDI Modes* can be found in the "MIDI Controller Assignments" section, on page 477.

If you select *MIDI Mode = RK*, every drawbar responds to a specific MIDI control change number, commencing with CC #70. (Non-drawbar parameters may be set using control change messages up to CC #118).

If you select *MIDI Mode = HS*, all of the EVB3 drawbars will be controlled by just a few control change numbers—CC #80–82. Its values are intelligently mapped to all drawbars. The resolution of this technique is not particularly high (much like the original B3), but it works well. The Hammond-Suzuki XB-2 uses this controller assignment method, allowing you to remotely control the EVB3's drawbars.

If you select *MIDI Mode = NI*, the Native Instruments B4 settings will be replicated, allowing the use of the Native Instruments B4D Drawbar controller.

If you select *MIDI Mode = off*, the EVB3 will not respond to MIDI drawbar messages.

Note: Before you commence a recording session with drawbar movements, performed on a hardware organ and the EVB3 set to *MIDI Mode HS*, we recommend that you disable *MIDI Data Reduction* in Logic. The parameter is located in *File > Song Settings > Recording*. The function thins out the incoming data stream, which is hardly ever noticed with normal controller assignments. The controller mapping of the Hammond XB-2's drawbars, however, exploits the controllers to their limits. Therefore, *MIDI Data Reduction* can lead to incorrect values. Switch the parameter on after the drawbar recording session, in order to regain memory and processing resources.

The EVB3 Parameters

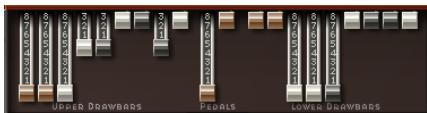


The graphical user interface (Editor view) opens when you double-click on the EVB3 Input slot of an Audio Instrument object. You can open and close the wooden lid by clicking the button underneath the *Volume* control. Keep it open while reading this section of the manual, because we'll be discussing every parameter in detail.



Drawbars

The principles of additive synthesis with sine drawbars is further explained in “Additive Synthesis With Drawbars” on page 481. You can intuitively pick up the basic principles by playing a little with the drawbars. The further down you drag the drawbars, the louder the selected sine choir(s) will be—the drawbars behave like reversed mixer faders. MIDI control of the drawbars is also reversed, when using a standard MIDI fader box.



Drawbars of the Upper and Lower manual, plus Pedal drawbars.

Pedal Drawbars

The organ features two drawbars for the bass Pedals. The waveform of the bass is not a pure sine wave, but a mixed waveform, that realistically simulates the Hammond B3 bass. The two registers differ in pitch, and in the following ways:

- the Lower 16' register contains more octaves
- the 8' register has a more prominent fifth portion.

Volume

Relative Volumes—Upper/Lower/Pedal

In the *Organ* parameters section, you can set the volume of the Lower manual, and the Pedal, relative to the Upper manual. The parameters are called *Lower Volume* and *Pedal Volume*.



Volume Control and Expression Pedal

The overall volume of the EVB3 is not only controlled by the Audio Instrument volume fader and control change #7, but also with the *Volume* control in the EVB3 graphic user interface.



Warning: The *Volume* must be lowered whenever crackling or other digital distortion occurs in the Audio Instrument channel. Volume levels over 0 dB can occur if you maximize all registers, play numerous notes, and make use of the Distortion effect.

You can control the volume in real-time with an Expression Pedal (Swell Pedal). Extensive, and often rhythmic, use of the Expression (volume) Pedal forms part of the style of many organ players. The expression control also emulates the tonal changes of the B3 preamplifier. Bass and treble are not attenuated as strongly as the mids, much like a Hi-Fi amplifier that features a loudness correction facility.

Normally, you would connect an Expression Pedal to the quarter inch Expression jack of your master keyboard. Your master keyboard should transmit MIDI control change #11 when the pedal is moved.



Tune

The simulated tone wheel generator can be tuned in cents (percentages of a semitone). 0 c is equal to A = 440 Hz.



Scanner Vibrato

The vibrato of the organ itself must not be confused with the Leslie effect, which is based on rotating speaker horns. The EVB3 simulates both.



The *Scanner Vibrato* is based on an analog delay line, consisting of several lowpass filters. The delay line is scanned by a multipole capacitor, which has a rotating pickup. It is a unique effect that cannot be simulated with simple LFOs.

Like the Hammond B3, the EVB3 features three types of vibrato—with different intensities (V1, V2, V3). Vibrato speed is set with the *Rate* parameter. In the V1, V2, and V3 positions of the *Type* parameter, only the signal of delay line is heard.

The C1, C2, and C3 Chorus positions of the *Type* parameter mix the signal of the delay line with the original signal. Mixing a vibrato signal with an original, statically-pitched signal, results in a chorus effect. The *Chorus* parameter allows you to freely mix the dry signal with the vibrato signal. The *Chorus* parameter is only active if one of the chorus settings is engaged.

If the C0 setting is active, chorus and vibrato are disabled. Note that the treble portion of the organ is boosted slightly if any vibrato setting is used. This treble boost is maintained in the C0 setting.

The *Upper* and *Lower* buttons allow you to switch the scanner vibrato (and its associated treble boost) on/off, individually, for the Upper and Lower manual. As the B3 mixes the bass register (Pedal) with the Lower manual, the Pedal register is affected by the Lower manual's scanner vibrato settings. This side-effect reflects the technical limitations of the original B3.

Note: Check out the chorus and vibrato effects, and compare them with the sound of the rotor cabinet simulation! The organ's chorus sounds different to modern chorus effects (such as Logic's Chorus plug-in). Many organ players rarely use the Scanner Vibrato, preferring to work with a Leslie, in isolation. Others, like B3 virtuoso Brian Auger, prefer the integrated organ vibrato over the Leslie.

Percussion

Percussion is only available for the Upper manual—as it does on an original B3. The percussion of an electro-mechanical organ is polyphonic, but is only (re)triggered after all keys have been released. If you release all keys, new notes or chords will sound with percussion. If you play legato, or sustain other notes on the Upper manual, no percussion will be audible.

On the original B3, percussion is only available if the “B” preset key is selected (see “Preset Keys and Morphing” on page 464). If you want this restriction to be simulated, set the *Perc* parameter to *Only B*. You find this parameter in the *Organ* section. If you always want percussion to be available, select *Always*.



Activate percussion by pressing the *On* button in the Percussion section, to the top-right of the GUI. The percussion is heard in the 4' register or in the 2 2/3' register, if the *2nd/3rd* button is set. Set the percussion decay time with *Time*, and set its level with *Vol*. This improves on the B3, where *Time* and *Vol* could only be switched on/off, whereas these parameters can be varied.

Note: Time has a maximum setting called Paradise. In this position, the percussion doesn't decay at all. The name is derived from a famous Jimmy Smith recording “Groovin’ at Small’s Paradise,” where Jimmy used a B3 that had a defect in the percussion trigger. The cool thing about this technical flaw is that one harmonic sounds without Chorus-Vibrato, while the drawbar harmonics feature Chorus-Vibrato. While we recognize that this is very specialized, we thought it would be a nice inclusion, especially for jazz-oriented players.

The EVB3 percussion register can be played with velocity sensitivity, unlike the B3. Set the percussion velocity sensitivity with *Vel*.

If you engage percussion on a B3, the volume of the normal, non-percussive registers is reduced slightly. The *Up Level* parameter simulates this behavior, allowing you to define the volume of the Upper manual, with percussion engaged.

Preset Keys and Morphing

The Hammond B3 is equipped with 12 switches, located below the lowest octave of both keyboard manuals. These are the *preset keys*, and are laid out as an inverted keyboard octave (black keys, white sharps). They are used to recall drawbar registrations. These presets could only be altered with a screwdriver on the original B3.



Drawbars and Preset Section

MIDI notes 24 to 35, the octave below the lowest octave of a (non-transposed) 5-octave keyboard, are used as the preset keys. See “Keyboard Ranges of the Upper and Lower Manual” on page 457.

You can click these preset keys directly in the graphic user interface of the EVB3. They are located to the left (Upper) and right (Lower) of the *Morph* wheel. The current drawbar registrations are indicated by small vertical lines on each key.

The presets only relate to the registration of a single manual. The presets do not store vibrato or other parameter settings. The preset keys work in real-time, thus overall settings (including effects) can be stored and recalled via the *Settings* menu.

Note: On keys C# to A#, the percussion only works in *Percussion Mode Always* (see “Percussion” on page 463).

Disabling MIDI Preset Switching

You can disable the switching of presets with MIDI notes 24 to 35, thus avoiding any problems that may arise from transpositions. To do so, *Disable* the *MIDI to Presetkey* parameter.

Cancel Key, Registering while Playing

The lowest preset key (C) is the cancel key. if you depress it, all drawbars are moved to their minimum setting. The other 11 keys, from C# to B, recall registrations. You can edit recalled presets immediately. The preset memorizes these alterations instantaneously, with no further action required. This means that if you recall a new preset, the former preset memorizes the drawbar settings at the time the new preset was recalled.

If you hold the cancel key (C) on your master keyboard with the small finger of your left hand, and sustain a chord with your right hand, you can trigger the chord with different registrations, by pressing the preset keys with the other fingers of your left hand. This results in an organ-specific gater type effect, which wouldn't be possible with the right hand alone.

Morphing

You can switch between the presets of the Upper manual with any keyboard. To be more precise, you can switch between registers with any MIDI controller—such as the modulation wheel. Choose the desired MIDI controller with the *Midi CC* parameter.

The *step* or *linear Mode* parameter options determine whether an abrupt switch, or a seamless morph (cross-fade), occurs between presets.

Note: Switching the registers with this morphing function is only available for the Upper manual.

Range

After deciding on a controller to use for switching or morphing the Upper manual registers, you can determine the number of preset keys that are affected.

The morphing (or switching) always begins with the top preset key, the *B. Range* defines the “end” preset key. If *Range* = *A#*, you will switch or morph between two presets. If *Range* = *G#*, you will switch or morph between four presets (B, A#, A, and G).

Save To

In *Linear Mode* (morphing instead of switching), the seamless crossfades result in a variety of new drawbar registrations you might want to save. Before saving, you may wish to alter some drawbar positions manually. Click the words *Save To*, and select the destination preset key. You do not need to do anything else, in order to save the setting to another preset key.

Note: As soon as you morph, the Morph text turns orange, indicating changes—in other words, you're performing a morph. The morphing result(s) can be further modified with the drawbars, but are lost if you don't save it/them. The Morph text will start to flash if any drawbar modifications are made, indicating changes to the Morph values.

Organ

The *Organ* parameters adjust the overall behavior of your EVB3.

The *Lower Volume* and *Pedal Volume* parameters are discussed in “Relative Volumes—Upper/Lower/Pedal” on page 461. The *Perc* parameter is discussed in “Percussion” on page 463.

Max Wheels



Calculating (emulating) all tone wheel generators consumes considerable CPU processing power. A reduction of this parameter value reduces the EVB3's hunger for processing resources.

Note: This will diminish some overtones, so you should not reduce this value if you're after an ultra-realistic simulation.

Tonal Balance

Tonal Balance changes the mix relationship of the higher/lower sounding tone wheels. Positive values result in a lighter and brighter sound. Experiment with different *Tonal Balance* and *Equalizer* settings. See the "Equalizer" on page 471, for further information.

Shape

While the Hammond's tone generators produce pure sine waves (despite technical artefacts), other organs deliver distorted waveforms. You can produce sounds resembling those of Farfisa, Solina or Yamaha organs with the *Shape* parameters.

You can subtly alter the waveform of the sounds emanating from the tonewheel generator with the *Shape* parameter. Moving the parameter to the right will make the tone brighter (and louder), and moving it to left will make it duller (and softer).

Note: The Shape parameter is placed after the filters which follow the sine generators.

Bass Filter

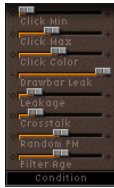
The sound of the Pedal drawbars often appears to be somewhat "brilliant", within the overall musical context. To circumvent this issue, and to suppress the treble of the bass register, please make use of the *Bass Filter*. When active, you will only hear a solid bass organ fundamental in the bass register.

Ultra Bass

If you switch on *Ultra Bass*, another low octave will be added to the playable range of the Upper and Lower manuals. These additional low octaves, and the ability to independently transpose both manuals (see "Transposition (Octave Range)" on page 458), are not available on the original B3.

Condition

Technical limitations of electro-mechanical drawbar organs with tonewheels cause some strange tonal artefacts, such as crosstalk. These quirks form an integral part of the B3's charm. You can adjust the following parameters to define the age of your EVB3.



Note: Read more about the Click parameters in “Click” on page 468.

Drawbar Leak

Even if all drawbars are at their minimum position, the tonewheel generators of the B3 aren't completely quiet. This is due to “leakage” of the tone wheels, cross-talking with the output. *Drawbar Leak* allows you to set the minimum output level of the drawbars, when set to minimum. At a value of 0, you can completely eliminate this leakage. At the maximum setting, drawbar leakage is most definitely audible.

Leakage

Leakage adds a sound resulting from the crosstalk of all tone wheels—including those of the notes that you don't play. This provides “breath” to the organ sound.

Crosstalk

The Hammond's tone wheels are divided into compartments of four—with the same key, but in different octaves. There are two tonewheels, four octaves apart, on each rotating shaft. The signal of the Lower wheel contains a small amount of signal, induced by the higher wheel, and vice versa. This “crosstalk” can be adjusted with the *Crosstalk* slider. Note that crosstalk is only audible on certain wheels, avoiding “rumble” when chords are played.

Random FM

If the tonewheel generator is clean, all frequencies are “straight”—the frequencies are even/in tune. The three-fold decoupling of the tonewheels—via springs, flexible couplings and flywheels—is very effective, but can't compensate for irregularities that come with dirt and grease in the driving gear. This gradual build-up of grime in the mechanism makes the revolution of the tonewheel assembly irregular on its axis. This is transmitted to the tonewheels. The *Random FM* slider allows you to simulate this effect.

Note: The effect only becomes audible in the higher frequency ranges.

Filter Age

The high frequency output signals of the B3's tone wheel generators are filtered by bandpass filters. The center frequency of these filters varies as the capacitors age. *Filter Age* allows you to alter the center frequencies of the filters. This colors the sound of the jitter applied by *Random FM* and the background noise resulting from *Leakage*. (See "Leakage" on page 467 and "Random FM" on page 467.) This parameter also influences the intonation of the organ, if you use the pitch bender.

Click

The key contacts of electro-mechanical tonewheel organs tend to "saw" a little on the busbar, introducing a short click sound. If any corrosion occurs to the key contacts or busbar, this will increase the length, and level, of this click. This aspect of the B3's design causes irregular scratching noises (commonly referred to as "keyclick"), when striking and releasing keys. Hammond fans like these clicking noises, as they introduce a transient, percussive quality to the note. The EVB3 allows you to adjust the volume and sound of the key click. The click sound is altered randomly, and independently, from the click on and click off (release) volume settings.



Click On/Click Off

These two knobs independently control the click volume for the beginning (*Click On*), and release, of the note (*Click Off*). The click off is quieter, even if both controls are set to the same position.

Click Min/Click Max

Not only are the tone color, and volume, of clicks altered randomly, but also their duration. Click duration can vary between a short "tick" and a longer "scratch". Minimum duration is defined by *Click Min*, and maximum duration with *Click Max*. The duration is displayed in milliseconds, as you move the sliders.

Note: Even if both parameters have identical values, there is still a random variation in sound. This variation makes some clicks seem shorter than the value set with *Click Min*.

Click Color

Set the tone color of the click here. Despite its random variation, you can define its treble portion globally.

Pitch



Compared to the original B3, the EVB3 offers several parameters to change its pitch behavior.

Note: The *Trans UM*, *Trans LM* and *Trans Ped* parameters are explained in “Transposition (Octave Range)” on page 458.

Stretch

The EVB3 is tuned to an equal-tempered scale. As a deviation from this standard tuning, you can stretch the tuning in the bass and treble ranges, much like acoustic pianos (especially upright pianos). If you select a value higher than 0 for *Upper Stretch*, the pitch of the higher notes will be raised. If you select a value higher than 0 for *Lower Stretch*, the pitch of the Lower notes will be lowered.

Note: The tones of clavinet, harpsichords, and pianos have inharmonicities in their harmonic structure. The frequencies of these overtones (harmonics) are not exact, whole-number multiples of the base frequency, as Pythagorean theory dictates. They are only approximate and are, in fact, a little higher. The overtones of lower (tuned) notes, therefore, are more closely related to the main frequencies of the upper notes. Due to the lack of strings, this inharmonic relationship is not true of organs. The stretch feature was included for situations where you may wish to use the EVB3 in an arrangement alongside an acoustic or digital piano (EVP88 and EVD6 Clavinet). When arranged in conjunction with an orchestra or synthesizers, the stretched tuning facility should not be used. Please experiment!

Lower Stretch controls the amount of deviation from the equal-tempered scale in the bass end of the sound. The higher the value, the further *down* the low notes are tuned. At a setting of 0, the EVB3 is tuned to an equal-tempered scale, with each octave below exactly halving the frequency. *Upper Stretch* controls the amount of deviation from the equal-tempered scale in the treble end of the sound. The higher the value, the further *up* the high notes are tuned. At a setting of 0, the EVB3 is tuned to an equal-tempered scale, with each octave up exactly doubling the frequency.

Warmth

Warmth controls the amount of random deviation from an equal-tempered scale. High values add “life” to organ sounds, but tend to sound a little out of tune.

When applying *Warmth* and *Stretch*, you should consider that these parameters may result in a detuned sound, which is similar to the overuse of a chorus effect. Straight tuning is nice too, so set *Warmth* to 0 if you're after a "pure" sound.

Pitch Bender, Brake Effect

The Hammond organ has no pitch bender. As such, use of the pitch bender is not suitable for realistic simulations, but it does provide a number of creative options.

Pitch up/down bender sensitivity can be set independently, in semitone steps, with the *Pitchbend Up* and *Pitchbend Down* parameters. The maximum sensitivity for upward bends is one octave.

You can set *Pitch Bend Down* to *Brake*, which gradually slows the movement of the tone wheels down to a total stop, at the pitch bender's minimum position.

Note: The *Pitch Bend Down = Brake* setting recreates an effect that is audible at the end of "Knife Edge" by Emerson, Lake and Palmer. Keith Emerson's virtuoso Hammond work was recorded on a reel-to-reel tape recorder. At the end of the song, you can hear the tape recorder being gently slowed to a total stop.

Sustain

Synthesizer players call the time the note takes to fade out after the release of the key the "Release Time." The EVB3 allows you to control this parameter as well; it's called "*Sustain*" in the organ lexicon. The three controls allow for individual settings in the Upper (*Up*), Lower (*Low*) and Pedal (*Ped*) registers.



If you select the *Smart Mode*, playing new notes will cut the sustain (release) phase of released notes. *Normal Mode* allows polyphonic sustain phases—all released notes will continue to sustain. *Smart Mode* allows long sustain times, even in the bass register, which would normally cause rumbling dissonances.

Effects

The EVB3 features a three-band equalizer, a reverberation effect, a pedal-controllable wah wah, and a distortion effect that simulates the sound of an overdriven tube amplifier. Finally, the signal can be processed by the rotor effect.

Effect Chain and Effects Bypass

Effect Chain

The EVB3's signal flow is as follows: the organ's signal runs through the equalizer, wah wah and distortion effects. You can choose between four different signal flow routings for the equalizer, wah wah and distortion effects in the *Effect Chain* pull-down menu. This treated signal is then fed into Reverberation and finally passed to the Rotor effect.

A "classic" B3 patch would be: an EQ'ed organ, plugged into a wah wah pedal, amplified by an overdriven Leslie. Select *EQ-Wah-Dist*.

The sound of the overdrive changes if the input signal is being filtered—be it by the EQ, or the wah wah. If you patch the EQ before the overdrive, the sound of the overdrive becomes much more flexible. The output signal of the distortion effect always contains high frequency content. If you want to suppress these frequencies, the wah wah must be the final effect in the chain—*EQ-Dist-Wah*.

If you wish to create a "screaming" sound (achieved by distorting the wah wah output), you can minimize any "harshness" by choosing the *Wah-Dist-EQ* routing.

You can suppress the brutal overtones of extreme distortions with two filters: Select *Dist-EQ-Wah*.

Effect Bypass

The Distortion, Wah, and EQ effects can be bypassed separately for the Pedal register. Set *Effect Bypass* to *Pedal* to do so. This avoids the entire bass portion of your organ being suppressed by the wah wah. It also avoids undesirable intermodulation artefacts, when utilizing the overdrive effect.

If you select *None*, the entire output of the organ is processed, as if you had plugged the B3's mono output into a Leslie cabinet.

Equalizer

As a Logic user, you won't have any questions about the *EQ Low*, *EQ Mid* and *EQ High* controls. The EQ algorithm is derived from Logic's Fat EQ. *EQ Level* is a master EQ volume control, allowing you to specify any gain level to the distortion effect. See "Effect Chain and Effects Bypass" on page 471 for details on effects routing options.

Reverb

Box, Small, Medium, Large, Big, and Spring are the names of the reverb algorithms.



Reverb level is defined by the *Reverb* parameter. A *Reverb = 0* value conserves processing resources.

You can also select *Bypass* in the *Reverb Mode* pull-down menu, if you want to disable the reverb without changing its level.

The reverb is always patched *after* the EQ, wah wah and distortion effects, but *before* the rotor effect. This means that the reverb always sounds as if it is played back through the rotor speaker. To hear the reverb after the rotor, switch off the organ's reverb, and use an aux send to apply reverb to the Audio Instrument object.

Wah

The name “Wah Wah” comes from the sound it produces. It has been a popular effect with electric guitarists since the days of Jimi Hendrix. The pedal controls the cutoff frequency of a band pass, low pass, or—less commonly—high pass filter. The wah wah pedal has also been used extensively with the Hammond organ.



MIDI Pedal Control

It is recommended that you permanently attach an Expression Pedal to your MIDI master keyboard. Your master keyboard should transmit MIDI control change #11 (Expression real-time volume), if you attach an Expression Pedal to the Expression jack and move it. This would normally be used to control the volume while playing.

If you program an EVB3 setting, set the *Expression* parameter to 0, and then define a wah wah effect (with controller 11 controlling the wah wah's cutoff frequency), you can control the wah wah with the pedal—without having to program anything on the master keyboard.

Read more on this in the “MIDI Setup” on page 456. You should also consult the users manual of your keyboard.

Wah Wah Control with other MIDI-Controllers or Aftertouch

You can use any MIDI control change message to control the wah wah effect. You can select any controller number and Channel Aftertouch (*Touch*) in the CC field.

Mode

Mode allows you to enable/disable the wah wah effect. If you select *Mode off*, the effect is disabled. There are six different filter types available:

- ResoLP (Resonating Low Pass Filter)
In this mode, the wah wah will work as a resonance-capable low pass filter. At the minimum pedal position, only low frequencies can pass.
- ResoHP (Resonating High Pass Filter)
In this mode, the wah wah will work as a resonance-capable high pass filter. At the maximum pedal position, only high frequencies can pass.
- Peak
In this mode, the wah wah will work as a peak (bell) filter. Frequencies inside the center frequency, which is controlled by the selected MIDI controller, will be emphasized.
- CryB
This setting mimics the sound of the popular Cry Baby wah.
- Morley 1
This setting mimics the sound of a popular wah pedal, manufactured by Morley. It features a slight “peak” characteristic.
- Morley 2
This setting mimics the sound of the Morley distortion wah pedal. It has a constant Q.

Range

Range controls the sensitivity of the wah to controller movements. If you only intend to make slight alterations to the cutoff frequency, choose a small value.

Bite

Bite is the name for the resonance parameter of the wah filter. You’ll know the parameter’s meaning, if you’re a synthesizer player. Anyway, check it out: The cutoff frequency is boosted. High values make the wah sound more aggressive.

Distortion

The distortion effect simulates an overdriven two-stage tube amplifier. Its primary role is the simulation of the Leslie amplifier—or whatever amp might be used to feed the Leslie speaker cabinet.



Type offers the choice between three different tube amp types: *Growl*, *Bity*, and *Nasty*. *Growl* simulates a two-stage tube amplifier, resembling the Leslie 122 Model, the classic partner for the Hammond B3 organ. *Bity* is reminiscent of a bluesy guitar amp. *Nasty* delivers hard distortions, and is well suited for very aggressive sounds.

The *Tone* control only affects the distorted portion of the sound, while the dry signal portion remains unaffected. This allows for very warm overdriven sounds that won't become "scratchy" if you try to get more treble out of the instrument.

Drive controls the amount of overdrive distortion. The output level is automatically compensated for, so there's no need for another master volume control adjustment facility.

Rotor Cabinet

The Hammond story can't be fully told without a chapter on the rotor sound cabinets, manufactured by Leslie. In fact, playing the B3 organ without a rotor cabinet is viewed as a "special effect" these days. The EVB3's rotor cabinet section simulates not only the speaker cabinet itself, but also the microphones which pick up the sound.



Cabinet

There are five settings available:

Off

In the *Off* setting, there's no rotor effect at all and you'll hear the direct output signal of the organ, and/or the other effects. There's an alternative to switching the rotor effect off: in the *Brake* mode, the speakers don't rotate, but are still picked up by the simulated microphones, in a random position. (see paragraph "Rotor Speed" below).

Wood

The *Wood* setting mimics a Leslie with a wooden enclosure, and sounds like the Leslie 122 or 147 models.

Proline

The *Proline* setting mimics a Leslie with a more open enclosure, similar to a Leslie 760 model.

Single

In the *Single* setting, the sound of a Leslie with a single, full-range, rotor is simulated. The sound resembles the Leslie 825 model.

Split

In the *Split* setting, the bass rotor's signal is routed more to the left side, and the treble rotor's signal is routed more to the right side.

Wood & Horn IR

This setting uses an Impulse Response of a Leslie with a wooden enclosure.

Proline & Horn IR

This setting uses an Impulse Response of a Leslie with a more open enclosure.

Split & Horn IR

This setting uses an Impulse Response of a Leslie with the bass rotor signal routed more to the left side, and the treble rotor signal routed more to the right side.

Rotor Speed

The Rotor Speed switches work as follows: *Chorale* = slow movement, *Tremolo* = fast, and *Brake* stops the rotor.

Speed Control

Organ players generally switch between *Choral* and *Tremolo*. With the EVB3, you can switch speeds remotely via controllers as the Modulation Wheel (*ModWhl*), Channel Aftertouch (*Touch*), Sustain Pedal (*SusPdl*), or, alternately, exclusively via mouse (*off*).

ModWheel—If you choose the modulation wheel, you can set all three speed settings in real-time. *Brake* is selected around the modulation wheel's center position, while *Choral* is selected in the lower, and *Tremolo* in the upper third of the modulation wheel's travel.

All other entries in the *Speed Control* pull-down menu work as follows: They toggle between *Tremolo* and the speed set with the *Rotor Speed* radio buttons, that means either between *Choral* and *Tremolo*, or between *Brake* and *Tremolo*. If *Rotor Speed* is set to *Tremolo*, you will toggle between *Tremolo* and *Choral*. The difference between the entries is how the toggle action takes place.

ModWhl Toggle—Toggles as soon as the Mod Wheel exceeds its center value on its way from its low position to its high position. If the Mod Wheel passes its center value on its way from its high to its low position, there will be no toggle. This has been implemented for Roland keyboards with combined Pitch Bend- and Modulation levers.

ModWhl Temp—Toggles as soon as the Mod Wheel passes its center value, regardless if you move the Mod Wheel from high to low or from low to high. This has been implemented for Roland keyboards with combined Pitch Bend- and Modulation levers.

Touch—Toggles as soon as you press Aftertouch. If you release Aftertouch, there will be no toggle.

Touch Temp—Toggles as soon as you press Aftertouch. If you release Aftertouch, there will be another toggle.

SusPdl Toggle—Toggles as soon as you press the Sustain Pedal. If you release the Sustain Pedal, there will be no toggle.

SusPdl Temp—Toggles as soon as you press the Sustain Pedal. If you release the Sustain Pedal, there will be another toggle.

Note: The Hammond B3 isn't equipped with a Sustain Pedal. This allows you to make use of your MIDI master keyboard's Sustain Pedal as a speed switch.

Rotor Fast Rate

Rotor Fast Rate defines the maximum possible rotor speed (Tremolo). While moving the slider, the Tremolo rotation speed is displayed in Hertz.

Acc/Dec Scale

The Leslie motors need to physically accelerate and decelerate the speaker horns in the cabinets, and their power to do so is limited. *Acc/Dec Scale* determines the speed at which the motors can accelerate the rotors (time it takes to get the rotors up to a determined speed), and the length of time it takes for them to slow down. If the slider is set to its far left position, you can switch to the preset speed immediately. If you drag the slider to the right, speed changes take more time to occur.

In the default position of *1*, which, like any default in Logic, can be set by clicking the slider while holding Option, the behavior is Leslie-like.

Horn Deflector

If you look inside a Leslie cabinet, you'll see a double horn, with a deflector at the horn mouth. This deflector "makes" the Leslie sound. Some people, however, removed it to alter the Leslie sound, as removal of the deflector increases amplitude modulation, and decreases frequency modulation. This parameter allows you to switch the deflectors *on* and *off*, without needing to order spare deflectors.

Mic Distance/Mic Angle

The *Mic Angle* slider defines the stereo image, by changing the angle of the simulated microphones. An angle of 0° results in a mono sound, while an angle of 180° causes phase cancellations. Experienced sound engineers tend to avoid wide spreads.

Mic Distance makes the sound darker, and less defined, when set to higher values. This is typical of microphones, when positioned further from the sound source.

Motor Control

The original Leslie amps have simple AC outlets for connection of the motor plugs. Modification is easy, you just remove or swap the plugs, for the motors.

Note: With *Single Cabinet*, the *Motor Ctrl* setting is irrelevant, because there are no separate bass and treble rotors in a *Single Cabinet*.

A "fancy" variation is the inverse mode (*Motor Ctrl = inv*). If you switch to *Tremolo*, the bass compartment rotates with fast speed, while the horn compartment rotates with slow speed, and vice versa in *Chorale* mode. In *Brake* mode, both rotors will stop.

The *910*, or "Memphis" mode, stops the bass drum rotation at slow speed, while the speed of the horn compartment can be switched. This may be desirable, if you're after a solid bass sound, but still want treble movement.

Another Mode is *Sync*: The acceleration and deceleration of the horn and bass drum are about the same. This sounds as if the two were locked, but is only clearly audible during acceleration/deceleration.

Additional Parameters

A number of additional parameters are accessible via the 001/011 button at the top of the EVB3 Plug-in window. The *Upper/Lower Stop Position* sliders allow you to set an exact stop position for the Leslie horn or bass rotator, respectively. This is something that the original Leslie could not do, sometimes resulting in a horn that was aimed at the back of the cabinet when it came to a halt, and a less than desirable sound. The *Velo to Click* slider allows you to set the velocity sensitivity of the *Click* parameters (see “Click Parameters” section, from page 494 onwards).

MIDI Controller Assignments

MIDI controller assignments are only updated if the default setting is loaded, or if a setting that was saved with a song is loaded.

All parameters that allow you to select a MIDI controller feature a *Learn* entry. If this option is selected, the parameter will automatically be assigned to the first appropriate incoming MIDI data message.

Note: As the new entry is added at the top of the list, existing automation data needs to be increased by one.

In most circumstances, Logic’s track automation is your best option for the recording of manual movements in the Plug-in window.

MIDI automation is the better option when you wish to make use of your hardware MIDI drawbar organ as a remote control for the EVB3. The following tables outline the MIDI controller assignments of the EVB3. The assignments are optimized for use with several popular MIDI drawbar organ models. In order to match the default settings of all popular organs, some parameters are assigned twice. You need not concern yourself about this.

If you use a *Hammond XB-series* organ, switch to *MIDI Mode HS*. If you use a *Roland* or *Korg* organ, choose *RK*.

This table describes the MIDI Control Change Message number assignment when MIDI Mode is set to *RK*. This is the correct setting if you use a *Roland VK series* or *Korg CX-3* drawbar organ as a remote controller for the EVB3.

Controller Number	MIDI Mode RK: Parameter Name
70	Drawbar 16'
71	Drawbar 5 1/3'
72	Drawbar 8'
73	Drawbar 4'
74	Drawbar 2 2/3'
75	Drawbar 2'
76	Drawbar 1 3/5'
77	Drawbar 1 1/3'
78	Drawbar 1'
Rotor Cabinet	
80, 92	Chorale/Brake/Tremolo
81	Chorale/Brake
Reverb	
82	Reverb Level
Vibrato	
85	Upper Vibrato on/off
86	Lower Vibrato on/off
87	Chorus Vibrato Type
Percussion	
94	on/off
95	2nd/3rd
102	Percussion Volume
103	Percussion Time
Equalizer	
104	EQ Low
105	EQ Mid
106	EQ Hi
107	EQ Level
Wah	
108	Wah Mode
109	Wah Bite
Distortion	
110	Distortion Type

Controller Number	MIDI Mode RK: Parameter Name
111	Distortion Drive
112	Distortion Tone
Click Levels	
113	Click On Level
114	Click Off Level
Balance	
115	Main Volume
116	Lower Volume
117	Pedal Volume
Rotor Fast Rate	
118	Rotor Fast Rate

Only the controller assignments 80 to 82 differ, when *MIDI Mode* is set to *HS*. This setting matches the controller mapping of Hammond XB-series organs. The other controllers remain as described above. Switch *MIDI Data Reduction* off while recording these data from an XB-Series organ (*File > Song Settings > Recording Settings*).

Controller Number	MIDI Mode HS Drawbar Assignment
80	All Upper Drawbars
81	All Lower Drawbars
82	Pedal Drawbars, Scanner Vibrato, Bass Filter

This table describes the MIDI Control Change Message number assignment when MIDI Mode is set to *NI*. This setting matches the controller mapping of the Native Instruments B4D controller.

Controller Number	MIDI Mode 0: Parameter Name
12	Upper Drawbar 16'
13	Upper Drawbar 5 1/3'
14	Upper Drawbar 8'
15	Upper Drawbar 4'
16	Upper Drawbar 2 2/3'
17	Upper Drawbar 2'
18	Upper Drawbar 1 3/5'
19	Upper Drawbar 1 1/3'
20	Upper Drawbar 1'
21	Lower Drawbar 16'
22	Lower Drawbar 5 1/3'
23	Lower Drawbar 8'

Controller Number	MIDI Mode 0: Parameter Name
24	Lower Drawbar 4'
25	Lower Drawbar 2 2/3'
26	Lower Drawbar 2'
27	Lower Drawbar 1 3/5'
28	Lower Drawbar 1 1/3'
29	Lower Drawbar 1'
Vibrato	
31	Upper Vibrato on/off
30	Lower Vibrato on/off
Brightness	Vibrato
Attack Time	Chorus Intensity
Percussion	
Sostenuto	Percussion on/off
Release Time	Percussion Harmonic (2nd/3rd)
Sound Variation	Percussion Volume
Harmonic Content	Percussion Time
Equalizer	
90	EQ Low
70	EQ Mid
5	EQ High
Distortion/Click	
76	Distortion Drive
78	Distortion Tone
75	Click On Level
Leslie	
Pan MSB	Microphone Angle
3	Microphone Distance
GP 8	Leslie Accelerate/Decelerate
GP 7	Leslie Fast
ModWheel MSB	Leslie Speed
68	Controls brake functionality: If Value = 0.0, switch Leslie to Brake. All other values switch Leslie to previous speed.

Additive Synthesis With Drawbars

The Hammond B3 is *the* classic drawbar organ. As with an acoustic pipe organ, the registers (drawbars, or “stops” on a pipe organ) can be pulled out, in order to engage them. But in contrast to a pipe organ, the B3 allows seamless mixing of any drawbar registers. The more you drag the drawbars down, the louder they will become.

Despite characteristics such as key clicks, intonation undulations, distortions, and crosstalk (which are emulated by the EVB3), playing a single note, with a single register, results in a pure sine tone. Mixing harmonic sine tones results in more complex spectra, and is known as “additive synthesis”. Organs—even acoustic pipe organs—can be regarded as additive synthesizers. There are, however, several limitations that need to be considered before viewing the instrument in this way. These limitations, on the other hand, constitute the character of any real musical instrument, loaded with charm.

The naming of the drawbars is derived from the length of organ pipes, measured in feet ('). This naming convention is still used with electronic musical instruments. Halving the length of a pipe doubles its frequency. Doubling the frequency means nothing other than: one octave up.

The lowest register, 16' (far left, brown drawbar), and the higher octaves 8', 4', 2' and 1' (white drawbars) can be freely mixed, in any combination. 16' is commonly described as the “sub-octave”. When we’re regarding this register as the fundamental, the octave above 8' is the second partial, 4' the fourth, 2' the eighth and 1' the sixteenth partial.

With the 5 1/3' register—the second brown drawbar—you can add the third partial. This is the fifth above the 8'. Basically, the drawbars are arranged by pitch, but there is, however, an exception. The second drawbar (5 1/3') is sounding a fifth higher than the third drawbar. See the “Residual Effect” on page 482, for an explanation.

2 2/3' gives the sixth, 1 3/5' the tenth and 1 1/3' the twelfth partial. So the electromechanical tone-wheel organ gives you the partials 1 (16'), 2 (8'), 3 (5 1/3'), 4 (4'), 6 (2 2/3'), 8 (2'), 10 (1 3/5'), 12 (1 1/3') and 16 (1'). As you can see, the harmonic spectrum is nowhere near “complete”. That’s the reason why overdrive distortion effects are so popular with electromechanical tone-wheel organs—they enrich the harmonic spectra by generating more partials.

Note: The term “partial” is basically the same as “harmonic”, but they are counted in a slightly different way. The fundamental is counted as the first partial. its octave, twice the frequency, is the second partial, but is known as the first harmonic. The fifth partial oscillates at five times the frequency of the fundamental. The fifth partial is known as the fourth harmonic, because with harmonics, the fundamental is not counted (which makes the term “harmonic” less practical to use).

Note: $2\ 2/3'$ is the fifth over $4'$. $1\ 3/5'$ is the major third over $2'$. $1\ 1/3'$ is the fifth over $2'$. In the bass range, this can lead to inharmonic tones, especially when playing bass lines in a minor key. This is because mixing $2'$, $1\ 3/5'$ and $1\ 1/3'$ results in a major chord.

Residual Effect

The residual effect is a psychoacoustic phenomenon. Human beings can perceive the pitch of a note, even when the fundamental is completely missing. If we didn't "hear" this way, it would make listening to music with a kitchen radio impossible. Its speaker will never play back the fundamental of the bass line, as this frequency is far below the range that the speaker can transmit. If you pull out all registers of the drawbar organ, except for the fundamental— $16'$, you'll still perceive the same pitch. The sound becomes thinner, with less bass and less warmth, *but the pitch remains the same*.

Setting drawbar registrations often involves this psychoacoustic phenomenon. In the lower octaves, mixing the $8'$ and $5\ 1/3'$ sine drawbars creates the illusion of a $16'$ sound, although the frequency is missing. Old pipe organs also make use of the residual effect, by combining two smaller pipes, eliminating the need for long, heavy, air-hungry, and expensive giant pipes. This tradition is continued in modern organs, and is the reason for arranging the $5\ 1/3'$ under $8'$: The $5\ 1/3'$ tends to create the illusion of a pitch that is one octave lower than $8'$.

A Short Hammond Organ Story

Three inventions inspired Laurens Hammond (1895–1973), a manufacturer of electric clocks, to construct and market a compact electro-mechanical organ with tone wheel sound generation. The Telharmonium by Thaddeus Cahill was the musical inspiration, Henry Ford's mass production methods, and the domestic synchron clock motor were the other factors.

The Telharmonium was the first musical instrument that made use of electromechanical sound generation techniques. In the year 1900, its man-sized tone wheel generators filled a two-storey building in New York. For a short period around this time, subscribers could order Telharmonium music over the New York telephone network (the streaming audio system of the time). The only amplification tool was the telephone's mechanical diaphragm, as a proper tube amplifier and acceptable speakers had not yet been invented. The Telharmonium was a commercial flop but its historical status as the predecessor of modern electronic musical instruments is undeniable. The Telharmonium also introduced the principles of electronic additive synthesis (see "Additive Synthesis With Drawbars" on page 481).

Laurens Hammond began producing organs in 1935, in Chicago, Illinois, making use of the same sound generation method. The differences were; much smaller tone generators, and fewer registers. The patent for his model A organ dates from 1934.

Hammond also holds the patent for the electro-mechanical spring reverb, still found in countless guitar amplifiers today!

The Hammond B3 was manufactured between 1955 and 1974. It is the Hammond model preferred by jazz and rock organ players such as: Fats Waller, Wild Bill Davis, Brother Jack McDuff, Jimmy Smith, Keith Emerson, Jon Lord, Brian Auger, Steve Winwood, Joey DeFrancesco, and Barbara Dennerlein. In addition to the B3, there are a number of smaller Hammond instruments, known as the “spinet” series (M3, M100, L100, T100). Bigger console models, many of which were designed to suit the needs of American (USA) churches or theatres (H100, X66, X77, E100, R100, G-100), were also manufactured.

The production of electro-mechanical organs ceased in 1974. Thereafter, Hammond built fully electronic organs. Today, people at Hammond-Suzuki are more conscious of their glorious tradition and produce fine electronic drawbar organs. In 2002, they even introduced a new digital B3 model which mimics the design and functions of the classic B3 (except for the weight). The new B3 utilizes a real, mechanical, rotor speaker cabinet.

Tonewheel Sound Generation

Tonewheel sound generation resembles that of a siren. Of course, there’s no air being blown through the holes of a revolving wheel. Rather, an electro-magnetic pickup, much like a guitar pickup is used.

A notched metal wheel, called a tone wheel, revolves at the end of a magnetized rod. The “teeth” of the wheel cause variations in the magnetic field, inducing an electrical voltage. This voltage/tone is then filtered, sent through the manuals, amplified, has vibrato and expression applied to it, and is then amplified.

A long drive shaft is driven by an AC synchronous motor. 24 driving gears are attached to the shaft, with 12 different gear sizes. These gears drive the tone wheels. The frequency depends on the gear ratios, and the number of notches in the wheels. The Hammond is tuned to an almost exact well-tempered scale.

As with pipe organs that feature “multiplexed registers,” the Hammond organ uses certain generators for more than one purpose. Some high frequency wheels serve as the fundamental for high notes, and provide harmonics for lower notes. This has a positive impact on the overall organ sound, avoids detuning and stabilizes levels.

The Leslie

Don Leslie developed his rotor cabinets in 1937, and began marketing them in 1940. Laurens Hammond wasn't keen on the concept of rotating speakers at all! Leslie's approach was to simulate a variety of locations in the pipes, resulting in a new spatial perception for every note. The rotor speaker cabinets could simulate this effect, and the sense of space that they impart is incomparable, when placed side-by-side with any fixed speaker. The periodic undulations in sound and volume, and the vibrato caused by the doppler effect aren't all there is to the Leslie "sound"—it's the space effect, too!

The first Leslie, the model 30, had no chorale, just tremolo and stop. The Chorale idea (which came much later) was borne of a desire to add a vibrato to the organ. Chorale offers far more than a simple vibrato, and was first introduced to the market with the 122/147 models. At this time, Leslie also added the "Voice of the pipe organ" label to his cabinets.

It wasn't until 1980 that the two companies and brand names came together, six years after the last tonewheel organ was built. Mechanical Leslie rotor cabinets are still being built today, by the Hammond-Suzuki company. Even the newest digital B3 model is combined with a real, mechanical, Leslie cabinet.

As an interesting piece of trivia, Don Leslie never actually owned a Hammond organ!

If you're interested in every detail of the Hammond organ's history, models, facts, and recommendations for buyers, you should take a look at "The Hammond Organ—Beauty in the B." This book, by Mark Vail, is highly recommended.

This chapter covers everything about the EVD6, a virtual emulation of the Hohner Clavinet D6.

The sound of the Hohner Clavinet D6 is synonymous with funk, but was also popularized in the rock, pop, and electric jazz of the 1970s, by artists and groups such as: Stevie Wonder, Herbie Hancock, Keith Emerson, Foreigner, and the Commodores. If you've heard "Superstition" or "Higher Ground" by Stevie Wonder, then you'll know that the D6 is the funkier instrument alive!

The EVD6—Concept and Functions

The EVD6's synthesis engine emulates the sound of the Hohner D6 Clavinet, and does not make use of any sampling technology. It improves on the original in that it can be used in stereo, and on a noise level, there's no comparison.

The dynamics and scaling of the sounds, over the entire 60-key range (F to E) of the original instrument, has been extended across the full MIDI range (127 notes).

The EVD6's engine also simulates the various string buzzes, key clicks, and the tone of the pickups found in the original instrument. It synthesizes the "pluck" and "bite" of the attack phase, as well as the "sticking" of the hammer pads. The sound generator reacts smoothly, musically, and precisely to the 127 steps of velocity sensitivity, as defined in the MIDI specification. You can almost feel the strings beneath your keyboard!

The extensive set of *String* parameters allow you to radically alter the tone of the EVD6, enabling you to simulate an ageing Clavinet, or to create new "instruments." The EVD6 is capable of some truly unique sounds, which you'll discover when exploring it, and auditioning some of the included settings.

You'll appreciate the perfect integration of the EVD6 into Logic. In use, it's much easier to handle than a real world Clavinet. There's no need to transport a bulky and heavy instrument, or to attach any cables to it. In addition, the EVD6 eliminates the problems of reliability, getting new parts, and tuning—all of which are becoming increasingly difficult with the original instruments.

You will also discover an integrated effects processor incorporated into the EVD6's luxurious front panel, which provides a number of classic effects popularly-used with the original Clavinet. The algorithms featured in the effects processor have been adapted, and optimized, for the EVD6. Included are: a great sounding Wah, Modulation, and Distortion circuit.

Parameters of the EVD6



Most of the “slider” parameters on the EVD6 interface are mapped to zero-centered ranges—if a slider is in its middle “neutral” position, it doesn’t affect the base sound of the selected EVD6-model. If the slider is moved to the left or right, it will scale the original parameter value positively or negatively by that amount.

The EVD6 front panel can be broken down into five main sections, namely: the *Global* silver panel section at the bottom, the *Excite/Click* and *String* parameters to the top left, the *Pickup* window in the top center, and the *Effects* section at the top right.

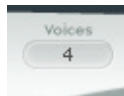
Global Parameters

The Global Parameters are found in the lower-left portion of the EVD6 Graphical User Interface (GUI).



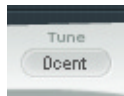
Voices

The *voices* parameter allows you to set the maximum number of voices that can sound simultaneously. Lowering the value of this parameter limits the polyphony and also the processing requirements of the EVD6. Minimal CPU power is used when the instrument is operated monophonically. There are two monophonic settings: *mono* and *legato*. Each setting provides only one voice for playing the EVD6. In the *mono* setting, the EVD6 voice is triggered each time a key is pressed. In the *legato* mode, the EVD6 sound shaping processes is not triggered if the notes are played legato—only the pitch changes. If the notes are played staccato, an EVD6 voice with all sound shaping processes is triggered. The maximum setting is 24, allowing for sustained glissandi. A setting of 24 will, of course, be more processor-intensive. 10 is the default. Click-hold, and use your mouse as a slider to adjust.



Tune

The global *Tune setting* works in cent steps. A value of 0 equals concert-pitch A 440 Hz. The range is ± 50 cents or, in more “musical” terms, plus/minus half a semitone. For transpositions in semitone or octave steps, please use the Instrument Parameter box in the Arrange window, as per any standard MIDI instrument. Click-hold, and use your mouse as a slider to adjust.



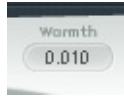
Bender

This parameter determines the bend range, in semitone steps. Click-hold, and use your mouse as a slider to adjust.



Warmth

Amount of random deviation from an equal-tempered scale. High values add “life” to sounds. It can be useful for simulating an instrument which has not been tuned for a while, or for slightly “thickening” a sound. When playing chords, the *Warmth* parameter creates the warm detuning or beating effect between the chord’s notes. Click-hold, and use your mouse as a slider to adjust.



Stretch

The EVD6 is tuned to an equal-tempered scale. As a deviation from this standard tuning, you can stretch the tuning in the bass and treble ends of the sound. This simulates the way stringed keyboard instruments such as pianos are tuned, attempting to find a more constant tuning balance between high and low notes. The *stretch* feature was included for situations where you may wish to use the EVD6 alongside acoustic pianos. When arranged in conjunction with an orchestra or synthesizers, the stretch tuning facility should *not* be used. Click-hold, and use your mouse as a slider to adjust.



Note: The tones of clavichets, harpsichords, and pianos have inharmonicities in their harmonic structure. The frequencies of these overtones (harmonics) are not exact, whole-number multiples of the base frequency, as Pythagorean theory dictates. They are only approximate and are, in fact, a little higher. The overtones of lower (tuned) notes, therefore, are more closely related to the main frequencies of the upper notes. Also see “Stiffness/Inharmonicity” on page 496.

Note: When applying *Warmth* and *Stretch*, you should consider that these parameters may result in a detuned sound, which is similar to the overuse of a chorus effect.

Pressure

On an original D6, applying pressure (aftertouch) to a depressed key raises the pitch slightly. The *Pressure* parameter allows you to do this, or alternately lower the pitch by pressure. Click-hold, and use your mouse as a slider to adjust. Range: -1.00 to $+1.00$



Filter Switches

The four filter switches emulate the original switches on the D6, with one exception. When all switches are set to *off*, you'll hear the unfiltered sound, rather than the original D6's "humming silence". Simply click anywhere on each switch to toggle between its on/off position. Active switches are indicated by pale green lettering, and by being depressed towards the bottom of the Plug-in window. You may use the filter switches in any combination of on/off positions.



- *Brilliant*—makes the sound nasal/cuts bass.
- *Treble*—makes the sound sharper/cuts bass more gently.
- *Medium*—makes the sound thinner/slight bass reduction.
- *Soft*—makes the sound softer/more muted.

Pickup Switches

As with the original D6, the two pickups can be used in different modes. The *AB* and *CD* switches are used to change modes. The internal wiring of the two pickups is changed in accordance with the different switch positions, and with it, the sound at the combined pickup output. The EVD6 features an additional menu that displays the current pickup mode above the pickup switches. More information on the use of these parameters/the pickups is found in "Pickup Parameters" on page 497.



C/D Switch	A/B Switch	What It Does
down	down	"neck" pickup—warm sound
down	up	"bridge" pickup—bright sound
up	up	both pickups—full sound
up	down	both pickups out of phase—thin sound

Stereo Spread

Pickup—While the original D6 only has a mono output, the EVD6 has stereo capabilities. When both pickups are active (*upper+lower* and *upper-lower* modes), the two pickup signals can be spread across the stereo spectrum. To adjust the position of the *stereo spread*, click-hold on the up/down arrows in the lower half of the circular button—in the *Pickup* section.



Turning the *Pickup Spread* control up will move the signals of both pickups away from the center position—one to the right, and the other to the left. Extreme left/right positions are reached when *Spread* is set to its maximum value. Range: *0.00* (center, no effect) to *1.00* (full left/right stereo).

More information on the use of the pickups is found in “Pickup Parameters” on page 497.

Key—This parameter allows a key scale modulation of the panning position—panning is determined by keyboard position. The center position is MIDI note 60 (Yamaha C3). To adjust the *Key*(board) position, click-hold on the up/down arrows in the top half of the circular button—in the *Key* section.

When fully turned up, the extreme left/right position will be reached at MIDI note 60 ± 30 semitones. Range: (center, no effect) to *1.00* (full left/right stereo).

Note: You can use both spread types at the same time. They will automatically be mixed.

Note: The effect of both stereo spread parameters is reflected graphically in the area around the circular *Stereo Spread* button.

Model

The *Model* parameter allows you to select a basic type of tone, or “model”. Each model has its own unique tonal characteristic, and each is suitable for the creation of very different sounds. Each model is an instrument in its own right, and can immediately be played, without any further editing. We will discuss each model below, and encourage you to experiment with each.



To select a model, simply click-hold in the area between the *Stereo Spread* and *Level* controls, and make your choice from the pop-up list. Release the mouse button once your selection is made. All EVD6 parameters are available to further shape the tonal character of the model.

In some respects, the *Model* parameter can be viewed much like making an oscillator waveform selection in a synthesizer. As with raw synthesizer waveforms, the editing parameters can affect the model quite differently. As an example, particular *Excite* settings may make one model more “nasal” sounding, and another model more “noisy”. These behavioral differences are a result of the unique harmonic structures used by each model.

The Models

Class(ic) D6

An almost 1:1 emulation of the original D6. It includes string noises on long decays, and realistic release behavior, following the release of the key(s). Each D6 was unique in its way, so feel free to adjust the many sound shaping controls, in order to match the sound of D6 units that you have used, or heard.

Old D6

This model emulates a well-worn D6. Hammers and strings are a bit aged, and worn. The sound of the sticky hammer heads is emulated, as well as the typically “richer” sound in the bass range.

Sharp D6

Very sharp with a lot of bite—nice with wah wah and phaser.

Mello(w) D6

As the name suggests, a mellow fellow.

Basic

Basic, simple clavinet

Domin(ation)

A powerful model with a strong and punchy attack—reacts more aggressively to velocity than other models.

GuruFnk (Guru Funk)

In the lower bass-octave ranges, the string oscillations become increasingly resonant over time, until they finally collapse (after 20 to 30 seconds). Higher notes have a much shorter decay, which also applies to their resonating behavior. This model invites heavy, funk-style bass playing in the lower octaves. It's nice with a little phaser, and sustained chords, when playing low bass notes. Adding a Logic Delay plug-in is also a great option!

Harpsi(chord)

Harpsichord-like model.

Pluck

Plucked string—changing the pick-up positions allows further modifications, making the sound more guitar-like. “Harp” style sounds are also possible, by positioning the lower pick-up around the mid position. To get a harp sound increase *String Decay*, *Release*, and *Excite Shape* and decrease *Excite Brilliance*.

(Tuned) Wood

Somewhat wooden, thin, and with some inharmonic overtones. Can sound slightly detuned in some contexts.

LtI (Little) India

Sitar-like sound, rich in resonance.

S(tring) Bells

A bell model with strong inharmonic overtones (inharmonicitities).

Dulcimer

Dulcimer-like model.

Picked

This model emulates a picked nylon string.

Special Notes about the Models

You may note some “zones” on the keyboard where the sound changes significantly between adjacent keys. This is intentional, and reflects the behavior of some of the real clavinet models emulated by the EVD6. The original D6 has some strong key-to-key timbral differences, with the most obvious one being between the highest, wound string, and the lowest, non-wound string.

If you're a player who likes the original's sound, but not the original's mechanical timbre jumps, the EVD6 offers a smoothed model—*MelloD6*.

When using a setting with both pickups quite close to the upper end of the strings and *Brilliant* + *Treble* filter switches active, the fundamental tone is quite weak in the output signal. As such, you will mostly hear the overtones that are *not* exactly in tune for inharmonic models (Wood, for example). Try moving the pickups to the center, and deactivate all filter switches to circumvent this detuned effect.

Level

Sets the (post -Effects) level, in dB (decibels). Click-hold, and drag, to adjust. If the MIDI controller used for Expression is not assigned to Wah or Damper, it is used to scale the output level.



Damper Wheel and Damper Ctrl

The original D6 features a damper slider on the right-hand side of the keyboard that allows the player to create muted string sounds. Click-hold, and drag, or make use of a MIDI controller, such as your keyboard's mod wheel, to adjust the *Damper Wheel*. The *Damper Wheel* position is saved with the sound.



The *Damper Ctrl* (number) parameter allows you to select the MIDI controller that moves the *Damper Wheel*. Click-hold, and make a controller number/name selection from the pull-down menu. Release the mouse button, once your selection is made.

Note: You can use MIDI Velocity to control the *Damper Wheel*. Just select the *Velocity* parameter from the pull-down menu.

The software-wheel is moved on-screen, when controlled via MIDI. MIDI control can be disabled by selecting the *off* option, found in the pull-down menu.

Note: The *Wah Ctrl* parameter is discussed on page 501.

Velocity Curve

There are nine preset velocity curves available for the EVD6. These allow you to set up a curve which is suitable for your playing style, or the sound.

The nine curves available are: *fix25%*, *fix50%*, *fix75%*, *fix100%*, *convex1*, *convex2*, *linear* (the default), *concave1*, and *concave2*.

Excite Parameters

Excite describes the string excitation, the physical power which stimulates the string to oscillate.



Shape

Shape adjusts the attack shape, allowing you to simulate the hardness of the rubber hammers in the original D6. As the instrument aged, the hammers would become worn, split, and so on, which had an impact on the overall brightness/tone of the D6. Negative values (to the left) provide a softer attack, while positive values result in a harder attack. Range: -1.00 to $+1.00$

Brilliance

Controls the harmonic content of string excitation. Positive values (to the right) result in a sharper sound. Negative values result in a more muted sound. Range: -1.00 to $+1.00$

Click Parameters

The rubber hammers of the original D6 age and decay, just like piano hammer felts. Well-loved (worn out) D6's produce a distinctive click when a key is released. This is due to the string sticking to the rubber hammer, before being released. The characteristics of this release click are part of each model, and can be finely adjusted with the following parameters.



Intensity

Positive values increase the level of the release click above the original model setting. Negative values reduce the level—a value of -1.00 equals no release click. If you'd like to simulate an old D6, increase the value, by moving the slider to the right. Range: -1.00 to $+1.00$

Random

Controls the amount of random click level variations across the keyboard. This slider simulates the wearing of some hammers, but not all of them, emulating the real-world wear and tear of the original. The further to the right the slider is moved, the greater the variation between key clicks on some keys. If all the way to the left, all keys have the same level of key click. Range: *0.00* to *+1.00*

Velocity

The *Velocity* parameter controls the level of release click modulation by velocity—Note On velocity or Note Off/release velocity. Range: *0.00* to *1.00*. The selection of note on/off information as the modulation source is determined by the KeyOn/Key Off buttons.

KeyOn/KeyOff Button

Press the appropriate button to select the type of velocity information that should be used for release click level modulation—press the *KeyOn* button, if you wish to use your attack velocity (how hard you hit the keyboard) as the value for the key click. If you wish to use your release velocity (how quickly you release the keys on your keyboard) to determine the value of the key click, press the *KeyOff* button. This requires a keyboard with release velocity facilities.

Needless to say, the *Velocity* parameter must be set to a reasonable level in order for the *KeyOn/KeyOff* modulation to be effective.

String Parameters

The behavior of the strings is basically determined by the *Model*, but the following parameters allow you to modify several string characteristics, relative to the model setting. See “Model” on page 491, for further information on model selection.



Decay

Positive values provide a longer *Decay* time after attacking a note. Negative values reduce the decay time. Range: *-1.00* to *+1.00*

Release

As per the *Decay* parameter, but for the *Release* time (following the physical release of a key). Range: *-1.00* to *+1.00*

Damping

The *Damping* parameter allows you to modify the damping of strings. Damping is essentially a faster decay for the higher partials/harmonics in a sound, and is a property of the string material used (high damping for catgut strings, medium damping for nylon strings, low damping for steel strings). Sonically, damping results in a more mellow and rounded, or woody sound, dependent on the *Model* in use. A positive value will make the sound more mellow, and a negative value will allow more of the higher partials through, making the sound brighter. Range: -1.00 to $+1.00$

Tension Mod

Tension modulation is a non-linear effect on strings, which usually results in the pitch being slightly higher, immediately after being plucked/struck/strummed. It is common to all stringed instruments, like the D6, guitars, and so on. This string characteristic is built into each model, but can be further modified with the *Tension Mod* parameter. The range of this parameter is quite large, and can be used to obtain weird sound effects from the EVD6. It can also be used to simulate an out-of-tune Clavinet, or as a quick and dirty sitar sound, for those “Norwegian Wood” covers. Range: -1.00 to $+1.00$

Stiffness/Inharmonicity

These two parameters allow you to intensify/reduce the strength of inharmonicity in the sound. When combined at different levels, these parameters can create metallic, bell-like sounds, or DX-like electric piano style sounds. They can also be useful for wood bass sounds. Experiment with both parameters, on each *Model*.

The higher the level of the *Inharmonicity* parameter, the lower its threshold to incoming frequencies. In other words, the *Inharmonicity* parameter determines the lowest harmonic, above which inharmonic spectral spreading becomes relevant. Range: -1.00 to $+1.00$

Stiffness controls the intensity of this stretching/spectral spreading. Range: -1.00 to $+1.00$

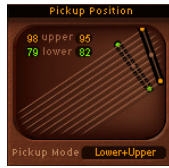
The keynote is not affected by these parameters.

Pitch Fall

Due to the physical construction of the original D6, the pitch of each note falls immediately after releasing the key. The intensity of this effect, which varies with each model, can be modified with this parameter. To completely deactivate the pitch fall, regardless of the selected model, set this parameter to the leftmost position (-1.00). Range: -1.00 to $+1.00$

Pickup Parameters

The original D6 is equipped with two electromagnetic pickups, much like those found in electric guitars: one below the strings (lower) and one above (upper).



Pickup Position

In contrast to the fixed pickups of the original instrument, the EVD6 pickups can be set to arbitrary positions *and* angles. To do so, simply click-hold on one end of the desired pickup (*Upper* or *Lower*) and drag the end to another position. Release the mouse button when done. Both values can be moved simultaneously. To do so click and drag the point in the middle of the pickup to a new position.

The numerical *upper* and *lower* panels, to the top-left of the window, indicate the current position of each pickup—with respect to the string. A value of 50 (percent) means that the specific pickup end is positioned above/below the centre of the string, resulting in a full-bodied tone. When the pickup approaches either end of the string (values near 0 or 99), the tone becomes thinner.

In the graphical pickup window the strings are aligned from left to right in respect to pitch—low strings to the left, high strings to the right.

It is recommended that you repeatedly strike a note when moving pickup positions, in order to hear the effect that the pickup position has on the overall tone of your sound.

Interesting, phaser-like effects can be achieved by automating the pickup positions.

Important: It is possible to cross-over the pickups in the Pickup Position window. This may lead to a “hole” (non or very soft sounding notes) within your keyboard range. This is due to a phase-cancellation between the pickups. If you encounter such cancellations, adjust one (or both) of the pickups until the required notes are playable.

Pickup Mode

Pressing the *AB* and *CD* switches will change the virtual wiring of the two pickups. The current wiring, the EVD6 calls it Pickup Mode, is displayed in the *Pickup Mode* panel. You can also click directly on the *Pickup Mode* panel, and select the desired mode from a pull-down menu.



- $C + A = \text{Lower}$
- $C + B = \text{Upper}$
- $D + A = \text{Lower+Upper}$
- $D + B = \text{Lower+Upper}$

Also see “Stereo Spread” on page 490, and “Pickup Switches” on page 489.

Effects Parameters

No Clavinet simulation would be complete if it didn't include a selection of effects processors. The EVD6 doesn't disappoint in this regard, incorporating three footpedal effects that have formed an integral part of “classic” Clavinet sounds over the decades. Each effect was painstakingly modelled on effects pedals that were available in the heyday of the Clavinet—the 1970's—ensuring that vintage sound in your performances.

Needless to say, you can also take advantage of Logic's extensive range of effect plug-ins, to further tailor your sound.

Distortion

The integrated distortion effect can be adjusted in both intensity and tone.

Range: *Tone* –2000 Hz to 20,000 Hz, *Gain* –0 dB to 20 dB. Using low *Tone* and *Gain* settings allows the Distortion unit to create warm overdrive effects. Bright and screaming distortion effects are produced with high *Tone* and *Gain* settings.



Compressor

Please note that the *Distortion* effect is always preceded by a compression circuit (shown in the panel above the *Tone* knob—with a ratio of 1:19.5) This allows you to increase/decrease the perceived gain, to provide the desired input level to the *Distortion* circuit.

The compressor allows for really “crunchy” distortions, coupled with wah, or phaser. It can also be useful for enhancing the keyclick sound, and emphasizing harmonics in the various models.



Compression Ratio

The *Compression Ratio* panel allows you to adjust the slope of the compression applied. To adjust, simply click-hold on the panel, and use the mouse as a slider.

The Compressor is tied to the Distortion effect, and always precedes it. As such, the *Effects Order* parameter is very important for placement of this compressor in the effects chain. Please see “FX Order” on page 502 for further information.

Note: If the Compressor/*Distortion* is used as the last effect in the chain, and its gain is turned down, but the *Compression Ratio* is high, you will effectively compress the output signal of the EVD6.

Wah

The typical Wah effect is generated by a dynamically moving filter. The EVD6 offers simulations of several classic wah effects, as well as some basic filter types. Possible values are: *off*, *ResoLP*, *ResoHP*, *Peak*, *CryB*, *Mor11*, *Mor12*. The abbreviations are for Resonant Low and High Pass filters, Peaking filter, CryBaby, Morley 1, and Morley 2. The latter three are famous effects pedal models that continue to be manufactured.



Note: The combination of wah, followed by distortion, delivers those sought-after funky fuzz-wah results.

Wah Mode

Simply click-hold on the (Wah) *Mode* panel, and select the desired (pedal effect) model from the pop-up menu.



Range

The *Range* setting determines the cutoff frequency of the filter (set with *Wah Mode*). With *Range* set to the left, the cutoff will only move in a narrow range. To provide a wider control range, turn the *Range* knob to the right.

Envelope (Depth)

An auto wah effect is produced by using an envelope follower to control the filter cutoff automatically. The envelope shape follows the dynamics of your performance. The sensitivity of the envelope in respect to your performance and thus the resulting filter modulation depth is set with the *Envelope* parameter. Turn *Envelope* to the right to increase the modulation depth.

Wah Ctrl

The *Wah Ctrl* parameter allows you to define the MIDI Controller (number/name) used as a manual wah effect control—a MIDI foot controller for example. You can also use MIDI Velocity to control the wah effect. Just click into the respective parameter field and select velocity from the ensuing pop-up menu. MIDI-control/coupling can be disabled by selecting *off*.



Note: Both envelope and a manual controller can control the wah simultaneously. In this situation, the effect of the envelope and manual controls are mixed.

As with the *Damper*, there is a slider named *Wah Pedal Position* that always represents the current pedal position. This ensures that the most recent pedal position is saved with the sound. The Wah Pedal Position Slider is only available in the Controls View. Use the pull-down menu found in the upper right corner of the gray portion of the Plug-in window to switch from the *Controls* view to the *Editor* view. The *Pedal Position* can be automated by either recording the MIDI controller messages, or by using the Track Automation system.

Modulation

The EVD6 features a Modulation unit with three switchable modulation effect types.



Mode

The Mode panel allows you to select either a *Phaser*, *Flanger* or *Chorus* as the modulation effect. Click-hold, and make your choice from the pop-up menu.

Phaser

The *Rate* parameter adjusts the speed of phasing, and the *Intensity* parameter adjusts the depth of phasing. Ranges: *Rate* 0.10 Hz to 10 Hz, *Intensity* 0 to 100

High values lead to very deep, self-oscillating phase shifts, for those cutting (and ear and speaker damaging, so take care!) sounds.

Chorus

The *Rate* parameter adjusts the speed of the Chorus effect, and the *Intensity* parameter adjusts its depth. High *Intensity* values lead to ensemble-type effects.

Ranges: *Rate* -0.10 Hz to 10 Hz, *Intensity* -0 to 100

Flanger

The *Rate* parameter adjusts the speed of flanging, and the *Intensity* parameter adjusts the depth of flanging. Ranges: *Rate* -0.10 Hz to 10 Hz, *Intensity* -0 to 100 .

FX Order

The order of the serial effects combination can be selected here. The four choices are:



- *WDM*—Wah > Distortion > Modulation
- *DWM*—Distortion > Wah > Modulation
- *MDW*—Modulation > Distortion > Wah
- *WMD*—Wah > Modulation > Distortion

Just like the foot pedals which could be freely connected to each other in series, the EVD6 effect section invites you to experiment.

The freely assignable effect routing is especially useful for selecting whether a distorted signal shall be wah-filtered, or if the wah-filtered sound shall be distorted (for screaming sounds), as one example.

Controlling the EVD6 via MIDI

It is possible to control and automate the parameters of the EVD6 and other plug-ins using the MIDI controls provided by many master keyboards or MIDI fader boxes.

MIDI controller assignments (Wah Ctrl, Damper Ctrl) are only updated if the default setting is loaded or a setting that was saved with a song is loaded, with one exception: If *Velocity* is chosen in a setting, the assignment is updated. When selecting a setting that has any controller other than velocity assigned, the default value or the last manually selected value (other than velocity) is used.

All parameters that allow you to select a MIDI controller offer a *Learn* entry. If this option is selected, the parameter will automatically be assigned to the first appropriate incoming MIDI data message.

Note: As the new entry is added at the top of the list, existing automation data needs to be incremented by one.

A Brief History of the Clavinet

German Company, Hohner, was the manufacturer of the Clavinet. Hohner were known mainly for their reed instruments (harmonicas, accordions, melodicas, and so on), but had made several “classic” keyboards, prior to the first incarnation of the Clavinet, known as the “Cembalet”.

Musician and inventor, Ernst Zacharias, designed the Cembalet in the 1950’s. This was intended to be a portable, amplifiable version of the Cemballo, or Harpsichord. It’s mechanism worked by plucking the end of a flat reed with the key, which was then picked up and amplified in much the same way as an electric guitar.

A year or two after the Cembalet’s release, two “Pianet” models appeared. Both the “CH” and “N” models used flat reeds for tone generation, but employed a very different plucking/striking action. When a key was depressed, it engaged a “sticky pad” with a foam backing, which actually stuck to the reed. When the key was released, the weight of the key caused the pad adhesive to free itself from the reed. This made the reed vibrate, and this vibration was then amplified.

The model “T” Pianet was released several years later, and utilized a soft rubber “suction pad” on the reeds, rather than the adhesive of the “CH” and “N” models. This method still had several drawbacks, however, as the dynamics available from the keyboard were limited. As a further shortcoming, all reeds were damped on release, thus negating the possibility of obtaining sustain via a foot pedal. Despite these glaring problems, the sound of the model “T” Pianet was popularized by bands such as the Zombies and Small Faces, in the 1960’s.

In the years between the releases of the Pianet “N” and “T” models, Zachariah invented what was to become Hohner’s most successful, and certainly funkiest keyboard—the Clavinet. The Clavinet was designed to replicate the sound of a Clavichord, but with an altogether fuller sound. (The Clavichord was notoriously thin sounding)



Original image from the D6 Users Manual.

The early models—Clavinet I with built-in amp, Clavinet II with tonal filters, Clavinet “L” with its bizarre triangular shape, all led to the Clavinet model “C”. This, in turn, was refined into the D6—a portable, amplifiable keyboard. The D6 used a hammer striking a string against a metal surface to produce its tone. It had a fully dynamic keyboard—as the striker is directly underneath the key, meaning the harder you hit, the louder and more vibrant the tone.

Mention the Clavinet today and most people will automatically think of Stevie Wonder’s “Superstition”—a recording that owes as much to the D6 as it does to the artist that wrote and performed it. The D6 was later superseded by the “E7” and the “Clavinet/Pianet Duo”. These were basically the same as the D6, but more roadworthy, quieter and better protected against proximity hums than previous models.

How the D6 Clavinet Works

Each D6 keyboard key forms a single arm lever. When a key is depressed, a plunger underneath touches the string and presses it onto an anvil. The string impinges on the anvil with a strength according to key velocity. This affects the dynamics of the sounding string.

These mechanical vibrations are converted into electrical frequencies through magnetic pick-ups which are amplified and reproduced through the loudspeaker.

As the key is released, contact between plunger and anvil is immediately broken, leaving the wool-wound part of the string free, so that the string vibration is immediately muted.

Double-Triggered Notes

When experimenting with the EVD6, or auditioning some of the included Settings, you may encounter sounds which seem to be triggered on both the note on *and* the note off.

This is actually a feature, which emulates the original D6. The real D6 has the “problem” of the strings sticking to the hammers if they are worn out, producing a second trigger when the key is released. You can adjust the intensity of this key-off click, with the *Intensity* slider in the *Click* section (see the “Click Parameters” on page 494). Move the slider to the left, and the second key off trigger will no longer be audible!

This chapter covers Logic's EVP88 virtual e-piano.

The sounds of various Fender Rhodes pianos are among the most popular keyboard instrument sounds used in the second half of the twentieth century. The various Rhodes models have been popularized in a wide range of musical styles, ranging from pop and rock, electric jazz, jazz rock, soul, and in countless ballads, plus recent house and hip hop genres. Nearly as popular was the Wurlitzer piano, which enjoyed most of its success in the seventies. The Rhodes, Hammond organ and subtractive analog synthesizers were considered the “fundamental” instruments in the keyboard rigs of rock musicians between 1965 and 1985, and they appeared to be incomparable and unbeatable ... until now.

The EVP88—Concept and Functions

The EVP88's piano synthesis engine simulates the sound of different Rhodes and Wurlitzer pianos, as well as the sound of the Hohner Electra piano. The piano synthesis engine is designed solely for the simulation of electric pianos, and does not make use of any sampling technology. As such, we can proudly claim that the EVP88 is most definitely not a digital piano! The EVP88 does not feature acoustic piano or grand piano sounds. Its only purpose is the ultra-realistic simulation of electric pianos. The dynamics and scaling of the sounds over the entire 88-key range is silky smooth and do not suffer from the abrupt changes in sound that typify sampled instruments. There are no audible loops, and we promise that you'll never hear any lowpass filters closing while the sound of a note is decaying.

The EVP88's engine also simulates the physical movement of the various electric piano reeds, tines, and tone bars in the electric and magnetic fields of the pickups found in the original instruments. It synthesizes the ringing, smacking, and bell-like transients of the attack phase, as well as the hammer action and damper noises. The sound generator reacts smoothly, musically, and precisely to the 127 steps of velocity sensitivity as defined in the MIDI specification.

Incorporated into the EVP88's front panel, you will discover an integrated effects processor which provides a number of classic effects popularly used on electric piano sounds. The algorithms featured in the effects processor have been specifically designed, adapted, and optimized for the EVP88. Included are: a great sounding equalizer, an overdrive, a stereo phaser, a stereo tremolo and stereo chorus.

Parameters of the EVP88



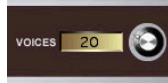
Global Parameters Model



The big switch shown above allows you to choose the electric piano model. When selecting a new model, all currently active (sounding) voices are muted, and all parameters are reset to standard values. As such, it is advisable to select the model before attempting to edit the effect and parameter settings. There are several Rhodes models available, such as the Mark I, Mark II and the suitcase piano plus the Wurlitzer and Hohner Electra Models. The EVP88 simulates the sound of these instruments which have (re)written modern popular music history. You can read more about the simulated instruments in "A Brief History of the Clavinet" on page 503.

The names of these instruments are registered trademarks and are protected by law.

Voices



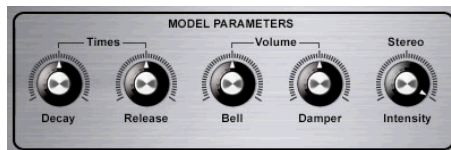
The *voices* parameter allows you to set the maximum number of voices that can sound simultaneously. Lowering the value of this parameter limits the polyphony and processing requirements of the EVP88. When the parameter is set to 1, the instrument is monophonic, and uses minimal CPU power. The maximum setting is 88, allowing for glissandi over the entire keyboard range with the sustain pedal depressed. A setting of 88 will, of course, be more processor-intensive.

Tune



The global *Tune setting* works in cent steps. A value of 0 equals concert-pitch A 440 Hz. The range is ± 50 cents or, in more “musical” terms, plus/minus half a semitone. For transpositions in semitone or octave steps, please use the Region Parameter box in the Arrange window, as per any standard MIDI instrument.

Model Parameters



Decay

Decay time of the piano sound. The lower the value, the less the sound is sustained, and the higher the level of damping applied to the vibration of the “tines.” When short values are used for this parameter, the main tone is more pronounced, and sounds longer than the transient harmonics. The effect is somewhat reminiscent of an electric guitar string being damped with the palm of the picking hand. Electric pianos can be modified in a similar way. Longer settings result in more sustain and a less dynamic feel.

Note: Check out Logic’s compressor plug-ins and experiment with different settings for decay.

Release

The release parameter determines the amount of “damper” applied after the keys are released. Extremely long settings allow you to play the piano like a vibraphone.

Bell

Bell determines the level of the inharmonic treble portion of the tone. It is useful for emulating a number of classic and typical electric piano sounds.

Damper

This parameter sets the level of the damper noise caused by the damping felt hitting the vibrating tine.

Stereo

If *Stereo* is set to high values, bass notes sound from the left, and treble notes from the right channel. The effect is nice and spacey, but it is not typical for vintage electric piano sounds. Even with acoustic pianos, the effect is less intense than one might expect.

Note: The stereo control is not restricted to the bass sound from the left, treble from the right use outlined above. Using Logic’s plug-ins, you can process the upper notes differently to the lower ones. With appropriate signal processing routings, you can, for example, add some bass via an EQ in the left “bass” channel and apply a little echo to the higher notes. As another option, you could listen to the summed post-effect signals in mono.

Stretch and Warmth



The EVP88 is tuned to an equal-tempered scale. As a deviation from this standard tuning, you can stretch the tuning in the bass and treble ranges, much like acoustic pianos (especially upright pianos). You can also modulate the tuning of each note randomly. The main tuning parameter is *Tune*.

Note: The tones of upright pianos, and—due to their longer strings, less so—grand pianos have inharmonicities in their harmonic structure. The frequencies of the harmonics are not exact, even multiples of the base frequency as dictated by Pythagorean theory. They are only approximate and are, in fact, a little higher. The harmonics of lower (tuned) notes, therefore, are more closely related to the main frequencies of the upper notes. Due to the lack of strings, this inharmonic relationship is not true of electric pianos, nor the EVP88. The stretch feature was included for situations where you may wish to use the EVP88 in an arrangement alongside an acoustic piano. When arranged in conjunction with an orchestra or synthesizers, the stretched tuning facility should not be used.

Lower Stretch

Deviation from the equal-tempered scale in the bass end of the sound. The higher the value, the further *down* the low notes are tuned. At a setting of 0, the EVP88 is tuned to an equal-tempered scale, with each octave down exactly halving the frequency.

Upper Stretch

Deviation from the equal-tempered scale in the treble end of the sound. The higher the value, the further *up* the high notes are tuned. At a setting of 0, the EVP88 is tuned to an equal-tempered scale, with each octave up exactly doubling the frequency.

Warmth

Amount of random deviation from an equal-tempered scale. High values add “life” to sounds.

Note: When applying *Warmth* and *Stretch*, you should consider that these parameters may result in a detuned sound, which is similar to the overuse of a chorus effect.

Effects

Equalizer

Treble

This is a conventional filter for the high frequency range. Depending on the model selected, shelving or peak type filters are utilized, with optimized frequency ranges for each model pre-selected.



Bass

This is a conventional filter for the low frequency range. Depending on the model selected, shelving or peak type filters are utilized, with optimized frequency ranges for each model pre-selected.

Note: By defeating the treble and bass frequency ranges, you can achieve a very direct and aggressive sound with a more dominant mid range. If you require more precise equalization, remember that you can insert any of Logic's equalizer plug-ins into the Audio Instrument channel strip. There's also a *Tone* control in the overdrive circuit which can be used to further contour the sound.

Drive

Playing an electric piano is best when using tube amplifiers. They offer a wide range of tones, ranging from the subtle warmth of crunchy guitar amplifiers through to psychedelic, screaming rock distortion. The EVP88 features an overdrive effect, which simulates the saturation characteristics of a tube amplifier stage. The overdrive process is the first signal processing circuit in the effects chain available in the EVP88.



Tone

The *Tone* control is used to EQ the sound before being sent to, and distorted by, the virtual tube amplifier circuit. You can choose a more mellow tonal color here, and still boost the treble with the equalizer after the overdrive circuit. If you prefer harsh distortion characteristics that come closer to overdriven transistor stages, use higher tone parameter values. If the sound becomes too hard, you can defeat the treble via the *Treble* control, post the overdrive process.

Gain

The *Gain* control determines the amount of harmonic distortion.

Phaser



Phaser pedals used by electric guitarists are “classic” effect tools for electric pianos as well—especially in the electric jazz, jazz-rock and pop styles of the seventies. Classical four-stage phasing effects are based on phase shifting using modulated all-pass filters. Mixing the phase-delayed signal with the original signal results in characteristic notches in the frequency response curve, also known as the comb-filter effect. The frequencies of the notches in the frequency range are not harmonic (as with the resonances known from the flanger effect), and these notches are shifted up and down through the sonic spectrum via LFO (low frequency oscillator) modulation.

Note: Logic offers more parameters in its Phaser and other modulation plug-ins. You can use these effects alternately to, or in conjunction with, the EVP88’s Phaser. The parameters found in the EVP88 Phaser have much in common with the best analog phasers of the 60’s and 70’s, including subtle analog-style distortion. It offers the same 32 Bit internal processing and sound quality of the Logic Phaser plug-in.

Rate

Speed of the phasing effect. When set to 0, the Phaser is switched off.

Color

Intensity of sound coloration introduced by the Phaser, caused by feeding the Phaser output signal back into its input.

Stereophase

Relative phase shift between the left and right channels, ranging from 0° to 180°. With 0° selected, the effect is most intense, but not stereophonic. With 180° selected, the effect symmetrically rises in the left channel while simultaneously falling in the right channel, and vice versa.

Tremolo



A periodic modulation of the amplitude (level) of the sound is known as tremolo. The modulation is controlled via an LFO. The Fender Rhodes suitcase piano features a stereo tremolo and many other electric pianos have a simple, but quite obtrusive, mono tremolo that can introduce a strange kind of polyrhythmic feel to performances.

Note: The original Wurlitzer piano has a mono tremolo with a fixed modulation rate of 5.5 Hz. For an authentic Wurlitzer sound, choose 0°. For Rhodes sounds, select 180°. The settings in-between result in nice spacey effects, especially at low LFO rates.

Rate

Speed of the tremolo effect (LFO frequency).

Intensity

Amount of amplitude modulation. With 0 selected, the tremolo effect is switched off.

Stereophase

At a setting of 0°, the level undulates in phase on both channels. With 180° selected, the modulation is perfectly out of phase, resulting in a stereo tremolo effect that is also known as “auto panning”. The effect is similar to manually turning the pan pot from side to side.

Chorus Intensity

The well-known chorus effect is based on a delay circuit, the delay time of which is permanently modulated by an LFO, while the delayed effect signal is mixed with the original dry signal. It is the most popularly used effect on electric piano sounds. This parameter regulates the intensity (the amount of delay time deviation), while the LFO rate is fixed at 0.7 Hz. Pay close attention when using high values as this may result in the piano sounding detuned.



Additional Parameters

The EVP88 features a number of additional parameters that are accessible via the 001/100 button at the top of the EVP88 Plug-in window.

The *Volume* slider sets the overall output level of the EVP88 (Range: -20 to +20 dB). The *Bend Range Down/Up* sliders determine the pitch bend range in semitone steps. The *Chorus Rate* slider sets the speed of the Chorus effect, in Hz. The *Delay PP/FF* sliders determine the delay time (in milliseconds) when the keys are struck pianissimo (PP—soft) or forte (FF—hard).

The E-Piano Models Emulated

Rhodes

The most commonly known and widely used electric piano model was constructed by Harold Rhodes (born 1910). Designed in 1946 as a piano surrogate for practice, education, and army entertainment, the Rhodes piano was successfully marketed by guitar manufacturer Fender from 1956. The Fender Rhodes has become one of the most popular musical instruments in jazz, especially electric jazz. Its popularity in pop and rock music occurred after CBS took over production of the Rhodes in 1965. Despite further changes in ownership throughout the company history, the instrument is most commonly called the “Fender Rhodes.” There are also a number of “Rhodes” synthesizers (which were developed by the now-defunct synthesizer manufacturer ARP). Japanese synth and music technology manufacturer Roland were the proprietor of the Rhodes name for a while, and released several digital pianos which carried the Rhodes moniker. From 1997, until his death in December 2000, Harold Rhodes again inherited the name.

The method of sound generation used by the Rhodes piano is based on metal reeds which function much like a tuning fork. These are hit by a hammer action that works in a similar fashion to that of a grand piano action. The asymmetrically designed “tuning fork” consists of a thin tine and a massive tone bar, which are bolted together. Due to construction considerations, some of the tone bars are rotated by 90 degrees. The piano is kept in tune by the mass of a spring which can be moved along the tine. The tine oscillates in front of an electric pickup, similar to that of an electric guitar. This functions along inductive principles, with permanent magnets placed around the tine having a damping effect on its movement, thereby affecting the sound.

Like the output signal of an electric guitar, the Rhodes output signal is rather weak and needs quite a bit of pre-amplification. The Rhodes sound is not harmonically-rich. This is why a treble boost or an overdrive effect, which can both add harmonics, is quite welcome when it comes to playing the Rhodes. Playing the Rhodes is, as mentioned earlier, at its best when using tube amplifiers.

The Rhodes piano was also made available as a suitcase piano (with pre-amp and two-channel combo amplifier) and as a stage piano, without amplifier. Both of these 73-key “portable” versions have a vinyl-covered wooden frame and a plastic top. In 1973, an 88 key model was introduced. Smaller “Celeste” and bass versions were less popular. The Mk II (1978) had a flat top instead of a rounded one. This allowed keyboardists to place extra keyboards on top of the Rhodes. In 1984 the Mark V was introduced, and even sported a MIDI output. Around this time, Rhodes production decreased as most keyboard players invested in the more flexible (and lighter) digital synthesizers available. These keyboards could emulate the sound of older pianos, like the Rhodes, and also had the bonus of a range of great new piano sounds.

The individual characteristic sound of each Rhodes piano depends more on the adjustment and maintenance of the instrument than on the model. Early models had hammers covered with felt, resulting in a smoother sound than the newer models, which had neoprene-covered hammers. The suitcase piano featured a pre amplifier which could create a sound with a very dominant mid range. But appropriate pre amplifiers and equalizers can make a stage piano sound the same. The stage piano has no power cord—just like an electric guitar.

The MkII has no resonance clamps in the treble range, unlike former models. This is why it has a little less sustain in the treble range. The most significant differences in terms of sound depend on how “deeply” the tine is adjusted. In cases where it is in a deep position—closer to the pickup—the bell characteristic becomes more prominent. In the eighties, many Rhodes pianos were adjusted so that they had more “bell”—the taste of the time.

There is little use in naming the most prominent Rhodes players and styles. Practically every keyboard player of the electric jazz, jazz rock, crossover, soul pop, and rock styles used to play it, at least in the seventies. Many still do. One of them is Ray Charles, who played the role of a blind music shop owner in the Blues Brothers movie. Negotiating the price of a used Rhodes, the Blues Brothers mentioned the lack of keyboard “action”. In the ensuing furious—and famous—Rhodes solo, Ray Charles proved that this particular Rhodes had plenty of “action”. For those of you who have never played the original instrument, the keyboard action feels a little smooth in travel and sticky when fully depressed. This makes its “feel” a little unusual for many players unfamiliar with it. Obviously, though, its feel is good enough for Ray Charles!

Rhodes Models:

- Suitcase MkI
- Suitcase V2
- Bright Suitcase
- Stage Piano MkI
- Stage Piano MkII
- Bright Stage MkII
- Hard Stage MkII
- MarkIV
- Metal Piano
- Attack Piano

The Metal Piano and Attack Piano models feature sound qualities that can be “aimed at” with the original Rhodes instruments, but not to the extent of these models. They do not sound realistic, but they are included as sound “ideals” that the Rhodes technicians might have had in mind when preparing their keyboards.

Wurlitzer Piano

This well-known manufacturer of music boxes and organs also built electric pianos, the portable versions of which have written pop and rock music history. The 200 series Wurlitzer pianos are smaller and lighter than the Rhodes pianos, with a keyboard range of 64 keys from A to C and an integrated amplifier and speakers.

The action resembles that of a conventional acoustic piano. It can be played with velocity sensitivity, just like the Rhodes. Its sound generation system is based on spring steel reeds which can be tuned with a solder weight. The Wurlitzer has electrostatic pickups: The reeds are supplied with a 0 volt current and move between the teeth of a “comb”, connected to a 150 volt current. The tone of the Wurlitzer, which was first manufactured in the early sixties, features many odd harmonics. If you were to ever try to emulate its sound with an analog synthesizer, you would start by switching the oscillator to output a 60% rectangular (PWM) wave.

The Wurlitzer is best known as the signature piano sound of the band “Supertramp”. You will know it from their “Crime of the Century” album. It can be heard on “Bloody Well Right”, “Dreamer”, “Hide in Your Shell” and also in “The Logical Song”. You might also recognize the Wurlitzer sound when listening to Pink Floyd’s “The Dark Side of the Moon” or “Wish You Were Here” (“Have a Cigar”, “Money”, “Time”) and “I am the Walrus” by the Beatles.

Wurlitzer Models:

- Wurlitzer 200 A
- Wurlitzer 240 V
- Soft Wurlitzer
- Funk Piano

The model *Funk Piano* does not sound realistic in the bass. We've added this special synthetic sound of the piano engine as a bonus.

Hohner Electra Piano

Not to be confused with the all-electronic RMI Electrapiano, the extremely rare Hohner Electra Piano offers striking hammers like those of the Rhodes, but a stiffer keyboard action. It was designed to resemble the look of a conventional acoustic upright piano. John Paul Jones of Led Zeppelin played it on "Stairway to Heaven," "Misty Mountain Hop" and "No Quarter."

Hohner Electra Model:

- Electra Piano

EVP88 and MIDI

Adaptation of Your MIDI Keyboards Velocity Sensitivity

The EVP88 responds with extreme sensitivity to the velocity information transmitted with MIDI note messages. It's advisable to set Logic's *velocity* and *dynamic* track parameters with care. In Logic Pro, you can try the following tip to fine-tune the velocity curve if you find that you're not getting the right feel with your MIDI keyboard.

- Create a Transformer object in the Environment, and cable it between the *Physical Input* and *Sequencer Input* objects on the *Click and Ports* layer.
- Set the transformer parameters so that all MIDI events with the condition "note" are set to *Use Map* under *Vel* in the lower operation line.

You can then "draw" your own individual keyboard velocity curve. For more detailed information, please refer to the Environment chapter of the Logic Pro reference manual.

MIDI Controller List

E-Piano	Volume	11
	Model	64
	Voices	72
	Tune	73
	Decay	65
	Release	66
	Bell	67
	Damper	68
	Intensity	71
Stretched Tuning	Lower	69
	Upper	70
	Warmth	74
Equalizer	Treble	75
	Bass	76
Overdrive	Gain	77
	Tone	78
Phaser	Rate	79
	Color	80
	Stereophase	81
Tremolo	Speed	82
	Intensity	83
	Stereophase	84
Chorus	Intensity	85

This chapter introduces Logic's EXS24 mkII sampler.



The EXS24 mkII offers all of the facilities that you would expect to find in a hardware sampler, without the cost and bulk of this type of device. As a purely software-based instrument, the EXS24 is perfectly integrated into Logic, and makes use of your computer's RAM and hard disks. This integration within the computer environment offers instant access to all audio data and Sampler Instruments used in a Logic song file. These files are stored on your computer's hard disks. This integration simplifies sample library management and eliminates the need for separate physical devices and the cables required to connect them.

You can make use of the editing features of Logic, or another audio editor for your samples. This is far more convenient, not to mention faster, on a computer monitor than on the cramped display found on most hardware samplers. As the samples are stored within the computer, the slow and often unreliable transmission of sample data back and forth between your sampler and Macintosh is eliminated.

The EXS24 is compatible with the EXS24, AKAI S1000 and S3000, SampleCell, ReCycle, WAV, AIF(F), Gigasampler, and SoundFont2 sample formats, as well as the Vienna Library, allowing access to large and comprehensive sampler libraries.

The EXS24 offers numerous sample processing and synthesis options, enabling you to tailor sounds to meet your needs.

Last, but not least: as a highly optimized Logic instrument, the EXS24 offers great performance, even on slower machines. The EXS24's performance is scalable, so you can look forward to enhanced functionality and increased polyphony on future computer technology. The number of possible Sampler Instruments available for simultaneous playback is directly related to the computer's processing and RAM resources. The more RAM you have, and the faster your CPU, the more Sampler Instruments can be loaded and played.

And what of the sound?

As the EXS24 uses high-end algorithms with 32-bit internal processing, is completely digital, and seamlessly integrates into Logic, you are guaranteed pristine, clear sample playback—up to 24-bit and 96 kHz, if you wish (and your audio hardware is appropriate). With the EXS24, there's no need to concern yourself over sound quality or compatibility issues with future audio formats.

Using Instruments

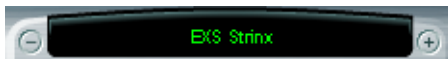
Folder Structure

The following items will be installed in the main Logic program folder:

- The *Sampler Instruments* folder—contains all of the Sampler Instruments received with the EXS24. This folder will also be used for the storage of all Sampler Instruments added or created in future. A Sampler Instrument contains all sample mapping information plus the modulation, filter, volume, and pan settings needed for a fine Grand Piano multisample, as an example.
- The *EXSamples* folder—contains all of the raw samples (audio files) that the Sampler Instruments make use of.

Loading and Playing an Instrument

The EXS24 ships with a ready-to-play Sampler Instrument library. These Instruments can be found within the *Sampler Instruments* subfolder of the *Logic* program folder. Once the EXS24's graphical interface is opened, you can select one of the Sampler Instruments by clicking on the pull-down menu above the silver panel area (directly above the Cutoff knob). The selected Sampler Instrument will then load.



Change the sound by twisting the knobs, pressing switches and moving sliders—and don't worry—you can't destroy the original Sample Instrument.

Creating and Editing Instruments in the Instrument Editor

Instruments are created and edited in the Instrument Editor. It is also used to organize samples and to convert foreign sample formats (AKAI S1000/3000 and so on). You can also assign samples to keys or key ranges and set start, end, and loop points plus all of the normally tedious tasks associated with sampling in the EXS24 Instrument Editor. Fortunately, the EXS24 Instrument Editor is much easier and more pleasant to work with than that of a hardware sampler. This is illustrated by the transparent architecture of the EXS24: samples are assigned to *Zones*, *Zones* are assigned to *Groups*. The end product of these assignments is a *Sampler Instrument*.

Now that you know how to load a Sampler Instrument, it would be a good time to briefly introduce you to the Instrument Editor. Please open the Instrument Editor window via the *Audio > EXS24 Instrument Editor* menu. The parameters and functions of the Instrument Editor window are described in this section.



The Instrument Editor shown above is empty as no Instrument has been loaded or created. The keyboard in the upper window area can be used to trigger notes for the EXS24 in the currently selected track. Below the keyboard, a number of Zones are shown.

Zones

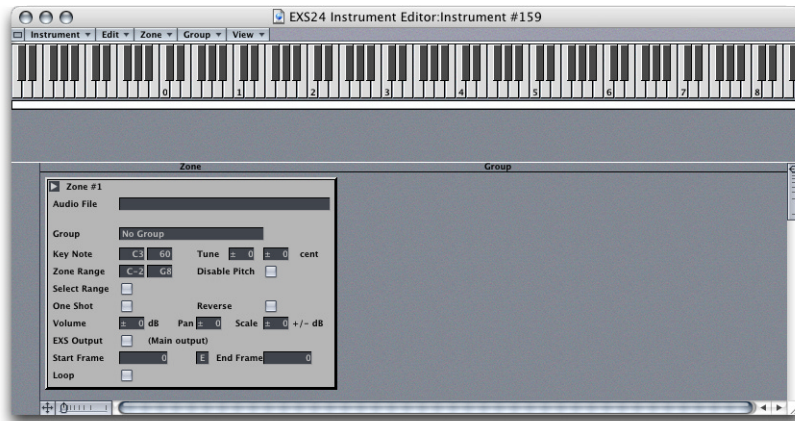
A *Zone* is a location into which a single sample (or audio file, if you prefer this term) can be loaded from hard disk or CD ROM. The sample loaded into the Zone is memory resident—it uses the RAM of your computer. A Zone offers various parameters for controlling the playback of the sample. Each Zone allows you to determine the range of notes over which the sample should be heard (Key Range), and the “root key” (Key Note)—the note at which the sample sounds at its original pitch. In addition, sample start, end, and loop points plus volume and several other parameters can be adjusted within the Zone. You can define as many Zones as you wish. Each Zone requires at least one EXS24 voice when played.

To create a new Instrument and a Zone

- 1 Select *Instrument > New* from the Editor window's menu. A new instrument is created.

Note: In order to hear your edits, please ensure that the correct Instrument is loaded into the EXS24 instance assigned to the currently selected track, and is selected in the editor.

- 2 Go to *Zone > New Zone* to create a new zone. A small window will appear to the left of the editor window.



- 3 Click on the empty field alongside the *Audio File* label. A file selection dialog box will launch, allowing you to select a sample from the hard disk or CD ROM.
- 4 Select the sample. It is loaded into the Zone.

Drag and Drop Zone creation

A new zone (and a new instrument, if none is currently displayed in the editor) can also be created by dragging a file onto one of the keys of the onscreen keyboard. The start key, end key and root key are all set to the note that the file was dropped on. This drag and drop functionality works for audio files from the following sources: Project Manager, Audio Window and the Finder.

You can create multiple zones by drag and dropping multiple files from these sources. When you do so, a dialog window will launch, asking how you would like these multiple files to be handled. See the "Load Multiple Samples" section, on page 561 for further information.

Note: If an audio file is dragged and dropped onto an existing zone (in the lower section of the EXS Instrument Editor window), the file referenced by that zone is changed to the new (dropped) file.

To move a Zone:

- 1 Move the mouse cursor (it will change to a two-headed arrow) over an existing zone bar (this is one of the bars displayed directly under the keys).
- 2 Click-hold and drag the zone to the desired position below the onscreen keyboard.

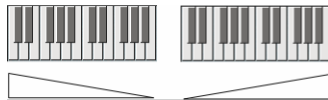
To change the start/end note of a Zone:

- 1 Move the mouse cursor to the beginning or end of a zone bar. The cursor will change to a left (start) or right (end) bracket, surrounded by arrowheads.
- 2 Click-hold and drag the start or endpoint of the zone to the desired length.

Note: You can also change the root key of the zone, by pressing Option and Command simultaneously, while dragging the zone.

Adjusting the Zone Parameters

Set up a key range for the sample with the two *Zone Range* parameters; *Key Note* allows you to determine the note used to trigger the sample at its original pitch. *Reverse* plays the sample from its end to the beginning. This option works non-destructively, and doesn't change the audio data. Adjust volume and pan position for the sample with the corresponding parameters. Negative *Scale* values make notes lower than the note position defined by the *Key Note parameter* sound louder than higher ones; positive values have the opposite effect. Use this parameter for balancing the volume of a sample across the selected key range.



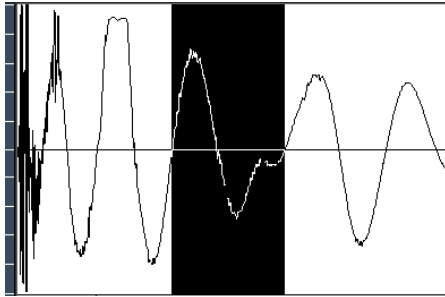
Negative *Scale* values increase the volume of lower notes (left of the *Key Note*), positive values increase the volume of higher notes (right of the *Key Note*). If necessary, adjust the playback start and end points for the sample with the *Start Frame* and *End Frame* parameters.

Activate *Loop* if desired; the loop parameters are hidden when the loop parameter is deactivated. You can set a start and end point for the loop and fine-tune the loop with the *Tune* parameter, if necessary.



Editing Samples

You may have noticed the small *E* buttons next to the start, end and loop point parameters. Clicking on these will launch the selected sample in Logic's Sample Editor, allowing you to edit the sample borders graphically, and use all of the Sample Editor's functionality. When loop is activated, you can also edit the loop points graphically: the *LS* marker indicates the loop start point and *LE*, the loop end point.



Groups

Imagine a drum kit has been created, with a number of different samples being used in several Zones, mapped across the keyboard. In many musical circumstances, it would be great to be able to treat each of the samples independently with the EXS24's sound editing parameters—to alter the decay of the snare, or to use a different cutoff setting for the hi-hat samples, for example.

This scenario is where the *Groups* come in—they allow for the very flexible organization of samples. You can define as many Groups as desired, and can assign each Zone to one of these Groups. In a drum set, for example, you could assign all kick drums to Group 1, all snares to Group 2, all hi-hats to Group 3 and so on.

Why might you want to do this?

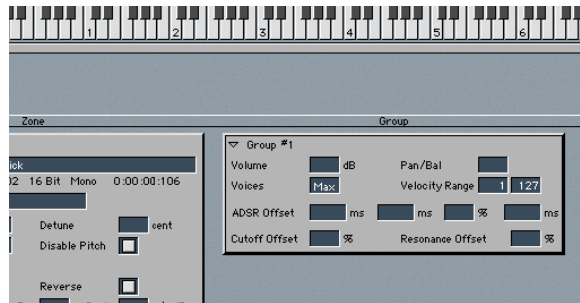
A Group makes it possible to define a velocity range for all assigned Zones, allowing you to specify a velocity window in which the grouped Zones should sound, as one example. Each Group also features offset parameters for the amplitude envelope and filter settings made in the Plug-in window.

It's also possible to play all Zones without defining and assigning even a single Group—in this case, the parameters in the Plug-in window work in an absolute manner for all Zones. To clarify, all samples in all Zones will be affected equally by the parameter adjustments made in the Plug-in window.

Given that up to 64 EXS24 instruments (dependent on your version of Logic) can be used simultaneously, opening several instances of the EXS24 provides the advantage of a dedicated channel strip for each and every sound you use. This allows full control over the sound (via EXS and effects parameters) during composition and mixdown.

To assign a Group to a Zone

- 1 Select *Group > New Group* in the editor's menu to create a new Group. A Group window will appear on the right-hand side of the editor.



- 2 Select the new Group as a target in the Zone's *Group* pull-down menu. The Group parameters will now affect the sample in the Zone.



Multiple Zones and Groups

You may create as many Zones and Groups as you wish, and can assign as many Zones to a Group as desired. The Groups offer several parameters for simultaneous control over all assigned Zones:

- The *Voices* parameter allows you to determine the maximum number of voices for a Group. A practical use of this would be to set up a classic “hi-hat mode” within a full drum kit, mapped across the keyboard. In this scenario, you could assign both an open and closed hi-hat sample to a Group, and set the *Voices* parameter of the Group to 1. In this example, the most recently triggered of the two hi-hat samples will mute the other, as only one voice is allowed for the Group. This mirrors the real-world behavior of hi-hats. When samples in Zones are assigned to another Group, the other sounds of the drum kit can still be played polyphonically.
- The Group *Volume* and *Pan* parameters simultaneously affect the settings of all Zones assigned to the Group. This works much like a sub group on a mixing console.
- The two *Select Range* parameters are used to set up a velocity window for the Group. Use these parameters for sounds where you wish to mix, or switch between, samples dynamically by playing your MIDI keyboard harder or softer—with layered sounds, or when switching between different percussion samples, for example.

- Each Group offers separate *ADSR* parameters for offsetting the ADSR volume envelope settings made in the Plug-in window: The *Attack*, *Decay*, and *Release* time parameters can be adjusted by ± 9999 ms, the *Sustain* level by $\pm 50\%$.
- Similarly, the *Cutoff* and *Resonance* settings of the Plug-in window can be offset by $\pm 50\%$ for each Group.

It is possible to play all Zones without defining and assigning a Group—in such cases, the parameters defined in the Plug-in window work in an absolute (identical) manner for all Zones.

Detailed descriptions of all Zone and Group parameters can be found in “Zone Parameters” on page 564 and “Group Parameters” on page 566.

File Organization

File Types and File Organization

The EXS24 uses the following file types and hierarchical structures:

Audio File

A single sample on your hard disk. The EXS24 is compatible with all audio file formats supported by Logic. Audio files are handled in the EXS24 Instrument Editor, where they can be edited and organized into Sampler Instruments.

Sampler Instrument

A Sampler Instrument points to one or more audio files, and organizes them as multi samples or drum maps, respectively. Within the Sampler Instrument you may assign different samples to different key and velocity ranges, set loop points, and adjust other playback parameters. You can also work with Zones and Groups (see the “Multiple Zones and Groups” section, from page 525 onwards), which always belong to a Sampler Instrument, and are not stored or loaded separately.

Note: Audio files are *not* contained in a Sampler Instrument. The Sampler Instrument simply stores information about an audio file’s name, its parameter settings, and its location on the hard disk. When you delete or rename an audio file, the Sampler Instrument that makes use of it will be unable to find it, so take care when handling audio files.

A Sampler Instrument is the file type that is loaded into the EXS24 for playing. When you select a Sampler Instrument in the EXS24’s pull-down menu, the associated audio files are automatically located on the hard disk, and are subsequently loaded into your computer’s RAM.

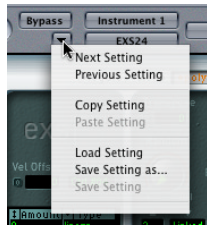
In order to be visible within the EXS24’s Sampler Instrument pull-down menu, Instruments must be stored in the *Sampler Instruments* sub-folder of the main Logic program folder.

Note: You can store your Sampler Instruments in any folder on any of your computer's hard drives. To do so, you must create an alias pointing to this folder within the *Sampler Instruments* folder located in the *Logic* program folder. Please refer to "File Organization" on page 526.

You can manually load Sampler Instruments from other locations into the EXS Instrument Editor at any time. Such Instruments also appear in the EXS24's Sampler Instrument load pull-down menu.

Settings

Settings are used to store all parameter adjustments made in the Plug-in window. Every Logic plug-in allows you to store and recall Settings, and the EXS24 is no exception. The Settings for the EXS24 are stored in the *EXS24* folder, which itself is located in the *Plug-In Settings* folder within the main *Logic* program folder.



Important: The Settings that can be stored and recalled in the Plug-in window are *not* part of the Sampler Instrument being loaded.

Settings reside above the *Sampler Instruments* in the hierarchy: A Setting contains a pointer to a Sampler Instrument, and when a new Setting is selected, the Sampler Instrument it points to is automatically loaded. As such, Settings are convenient for organizing and accessing your favorite Sampler Instruments. Settings also recall any changes made to parameters within the Plug-in window.

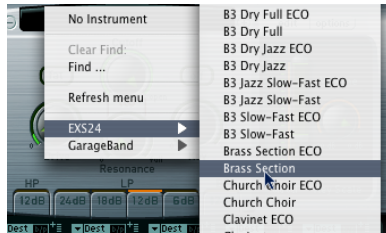
Management of Sampler Instruments

As your sample library grows, the list of Sampler Instruments will also expand. To aid you in keeping the list of Sampler Instruments manageable, the EXS24 features a sophisticated, but easy to use method of file management.

The Sampler Instrument pull-down menu directly reflects the folder structure within the *Sampler Instruments* folder. You can choose to sort your Sampler Instruments in groups such as "basses and guitars", by sound type, alphabetically, or by song.

To organize your Sampler Instruments into a preferred hierarchy:

- 1 Simply create a folder—“Basses” for example—within the *Sampler Instruments* folder, with your operating system’s file management utilities.
- 2 Drag and drop the desired EXS24 Sampler Instruments into this newly created folder. Their menu structure will be reflected when clicking on the EXS24 Sampler Instruments pull-down menu.



Note: You will need to relaunch Logic after changes are made to the folder hierarchy in the *Sampler Instruments* folder.

The menu is limited to the display of folder sub-menus that actually contain EXS instrument files. Other folders are not added to the menu. Aliases pointing to folders which contain EXS instrument files outside the *Sampler Instruments* folder can also be added to the menu. Even the *Sampler Instruments* folder itself can be an alias to a folder on a different drive or location.

When selecting a Sampler Instrument from a sub menu, a bold entry at the top of the root menu is added, to indicate the current selection. The sub menu that contains the selected Sampler Instrument is also shown in bold type, as are further sub menus. This makes it easy to trace the file path of the currently loaded Sampler Instrument.

Saving of Project-related EXS24 Instruments

This feature allows all EXS24 Instruments associated with a Project to be saved/loaded into/from a single folder location, which also contains the song file. These Sampler Instruments will then be exclusively associated with this song.

This is useful for two reasons:

- It makes the archiving and handling of songs, including the associated Sampler Instruments, easier.
- It makes it simpler to deal with a particular set of samples that will not be used in another song—vocals, modified drum kits and so on.

It works as follows: When opening a Logic Project, the EXS24 initially looks for a sub-folder named “Sampler Instruments” in the folder that contains the song file. If such a sub-folder exists, all Sampler Instruments found in this folder are added to the Sampler Instrument pull-down menu in the EXS24 GUI. This new entry in the Sampler Instrument pull-down menu will appear as a sub-menu item that matches the song file name. This behavior ensures that the EXS24 will always locate any song-related Sampler Instrument files *before* searching in the global Sampler Instrument folder, found in the Logic program directory.

To save Instruments related to a particular song

- 1 Create a new folder for a song/Project and name it.
- 2 Save the song file itself into this folder.
- 3 Create a sub folder named “Sampler Instruments” within the Project folder.
- 4 Simply copy/move the Sampler Instrument files required into this folder. Note that only the Sampler Instrument files, *not* the raw samples used by these Sampler Instruments should be copied, except when archiving (or unique samples are used), as discussed below. The “Used by EXS24” option could be useful.

Even simpler:

- 1 Save your song with the *File > Save as Project* function. More information on this can be found in your Logic manual.
- 2 When Logic is booted, the song is loaded, and an EXS24 instance is opened; a new hierarchical menu item will appear within the EXS24 Sampler Instrument pull-down menu when clicked. This new menu item will retain the song’s name and contains all of the Sampler Instrument entries copied to this folder earlier.
- 3 When saving any newly created or modified Sampler Instruments, ensure that you use the “Save as” function and browse to the “Sampler Instruments” folder inside the new song folder.

When saving on a per-song basis, you should observe the following folder hierarchy:

- The Project folder contains the song file and the “Sampler Instruments” folder.
- The “Sampler Instruments” folder contains all Sampler Instruments that are used in this song exclusively.

As the EXS24 automatically locates the audio files associated with Sampler Instruments, it generally does not matter where these audio files are stored. One circumstance, however, where the storage location of the audio files *does* matter is as follows: Should you need to archive the song with all related data, or wish to deal with a particular set of samples that will not be used in another song, you will want to store the *audio files* inside the Project folder as well.

This will change the folder hierarchy as follows:

- The Project folder contains the song file and the *Sampler Instruments* folder.
- The *Sampler Instruments* folder contains all Sampler Instruments that are used in this song exclusively—vocals, for example.
- A separate folder containing the audio files associated with the respective Sampler Instrument for *each* Sampler Instrument used.

To assist you in doing this, the EXS24 Instrument Editor provides the following functions:

- Instrument > Copy Audiofiles
Copies the audio files of any Sampler Instrument edited in the EXS24 Instrument Editor to the target directory of your choice. A folder for the audio files associated with this Sampler Instrument is created in the target location. The Sampler Instrument file itself is also copied.
- Instrument > Move Audiofiles
Moves the audio files of any Sampler Instrument edited in the EXS24 Instrument Editor to the target directory of your choice. A folder for the audio files associated with this Sampler Instrument is created in the target location.
- Functions available as Key Commands
The *Backup audiofiles of all USED and ACTIVE instruments of current song* key command copies the audio files of all (active) Sampler Instruments used by the current song to the target directory of your choice. Folders for the audio files associated with these Sampler Instruments are created in the target location. All used Sampler Instrument files are also copied.
- The *Move audiofiles of all USED and ACTIVE instruments of current song* key command moves the audio files of all (active) Sampler Instruments used by the current song to the target directory of your choice. Folders for the audio files associated with these Sampler Instruments are created in the target location.

Searching for Sampler Instruments

As a further navigational enhancement, the EXS24 features a built-in *Find* function, which works in conjunction with the hierarchical menu structure discussed earlier.

In order to minimize the number of Sampler Instruments displayed in the Sampler Instrument pull-down menu, you can make use of the *Find* function. This will limit the Sampler Instrument pull-down menu to only display Sampler Instrument names that contain the word “piano” or “bass”; as an example. This will also hide any sub-menus that don’t contain the search word. Simply select *Find* in the Sampler Instrument pull-down menu and, in the ensuing dialog box, type in the character string (search term) to search for.

The *Clear Find* option in the Sampler Instrument pull-down menu will display the full menu but does not clear the actual search term typed into the search dialog. To return to the limited menu, simply select *Enable Find*. The selection of *Enable/Clear Find* allows you to toggle between the two without re-typing the search term.

If you wish to use a different character string, select the *Find* option a second time and type in the desired search term.

Sample File Import

The EXS24 is compatible with the EXS24, AKAI S1000 and S3000, SampleCell, ReCycle, Gigasampler, and SoundFont2 sample formats, as well as the Vienna Library.

Using EXS24 Files

We strongly recommend that you copy any EXS24 Sampler Instruments shipped on CD-ROM to your hard drives—for two reasons: firstly, to always have direct, immediate access to your Sampler Instruments without searching for and inserting CD-ROMs, and secondly, to be able to sort your Sampler Instruments according to your needs.

To copy an EXS24-format Sampler Instrument, along with its associated audio files, from CD-ROM to your hard drives:

- 1 Copy the Sampler Instrument files from the CD into the *Sampler Instruments* folder within the *Logic* folder.
- 2 Copy the associated samples from the CD into the *EXSamples* folder within the *Logic* folder.

Note: You can sort your Sampler Instruments to suit your own needs (see “File Organization” on page 526). The EXS24 file system is able to work with aliases for Sampler Instrument folders. Furthermore, a Sampler Instrument searches for, and finds, all samples it uses on all active hard drives—as long as you do not delete or rename the samples.

Using EXS24 Instruments directly from CD-ROM

Normally, the Sampler Instrument and associated samples (audio files) will be stored on your hard disks, but on occasion, you may wish, or need, to load an EXS Sampler Instrument from CD-ROM.

To use an EXS Sampler Instrument stored on CD-ROM:

- 1 Copy the Sampler Instrument file (*not* the associated samples) from the EXS format CD-ROM into the *Sampler Instruments* folder.

- 2 When the Sampler Instrument is loaded, ensure that the appropriate CD-ROM is in the computer's CD-ROM drive. If the appropriate CD-ROM (the one that contains the desired Sampler Instrument and its associated audio files) is in the drive, the EXS24 will automatically search for the associated samples on all local media. It will locate the CD-ROM and will load the Sampler Instrument.
- 3 If the CD-ROM is not present, you will be required to insert the appropriate disc and reload the Sampler Instrument.

Note: Aliases/shortcuts may only be used for files stored on hard disk, not on CD ROM.

Importing SoundFont2 Files

To make use of this functionality, simply copy or move your SoundFont2 files into the *Sampler Instruments* folder.

Select the file name in the EXS24 Sampler Instrument load flip-menu and the file will automatically be converted. An EXS Instrument file will be created in the *Sampler Instruments* folder which contains the original SoundFont2 file. The raw samples associated with the Sampler Instrument will be placed in a *SoundFont Samples* folder within the *Logic* program folder.

Should a SoundFont2 Bank file (a Bank contains multiple sounds—a General MIDI bank, for example) be loaded, it will create a *Bank* folder and also a *Samples* folder. These new folders will have the same name as the SoundFont2 Bank file, with the word “Bank” or “Samples” appended.

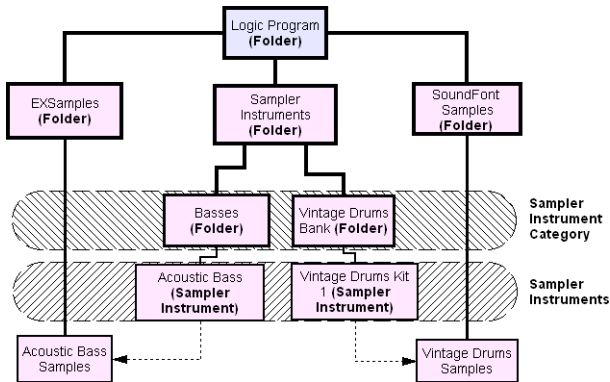
All sounds contained in the bank will automatically have an EXS Sampler Instrument file created and placed into the newly created *Bank* folder. The EXS24 Sampler Instrument pull-down menu will automatically be updated to reflect the new folder hierarchy. All samples associated with the *Bank* will automatically have a *Samples* folder created inside the *SoundFont Samples* folder which resides in the *Logic* program folder.

As an example, a SoundFont2 bank file named “Vintage Drums” is imported by the EXS24. It contains over 50 individual drum kits from several different vintage drum machines. A new folder named *Vintage Drums.Bank* will be created in the *Sampler Instruments* folder. A second folder named *Vintage Drums.Samples* will be created in the *SoundFont Samples* folder. Both of these folders are found in the main *Logic* program folder.

The Sampler Instrument pull-down menu hierarchy is updated and the original *Vintage Drums* entry is replaced with a *Vintage Drums.Bank* entry. This new entry is a folder that contains the individual Sampler Instruments, which can be selected and loaded as per usual.

Once conversion is complete, the original SoundFont2 source files can be freely deleted from the hard disks.

Note: You can store your imported Sampler Instruments in any folder on any of your computer's hard drives. To do so, you must create an alias pointing to this folder within the *Sampler Instruments* folder located in the main *Logic* program folder. Care should be taken when importing samples to ensure that when a song is loaded, the associated Sampler Instruments will be found. Sampler Instruments are only searched for in the *Sampler Instruments* folder (or an alias to it). Any Sampler Instruments stored in other locations will not be located, and must be loaded manually.



The folder hierarchy of the EXS24.

Importing SampleCell Files

The importation of SampleCell format files is as per that of SoundFont2 files. Simply copy or move your SampleCell files into the *Sampler Instruments* folder.

Select the file name in the EXS24 Sampler Instrument load flip-menu and the file will automatically be converted. An EXS Instrument file will be created in the *Sampler Instruments* folder which contains the original SampleCell file. The raw samples associated with the Sampler Instrument will be placed in a *SampleCell Samples* folder within the main *Logic* program folder.

Once conversion is complete, the original SampleCell source files can be freely deleted from the hard disks.

Should you import SampleCell or AKAI format Samples, they will appear as a *SampleCell Samples* or *AKAI Samples* folder on the same level as the *EXSamples*, *Sampler Instruments* and *SoundFont Samples* folders. Please refer to the EXS24 folder hierarchy diagram above.

Importing Giga Files

The importation of Giga format files is as per that of SoundFont2 files. Simply copy or move your Gigasampler files into the Sampler Instruments folder.

Select the file name in the EXS24 Sampler Instrument load flip-menu and the file will automatically be converted. An EXS Instrument will be created in the Sampler Instruments folder which contains the original Giga file. The raw samples associated with the Sampler Instrument will be replaced in a *Giga samples* folder within the main Logic program folder.

Once conversion is complete, the original Giga source file/s can be freely deleted from the hard disk.

Should you import Giga samples, they will appear as a *Giga samples* folder on the same level as the *EXSamples*, *Sampler Instruments* and *SoundFont Samples* folders. Please refer to the EXS24 folder hierarchy diagram above.

Converting ReCycle Files to EXS24 Instruments

ReCycle is a sample editing program from Propellerheads software. It slices sample material into small segments (slices) over time, and can generate a number of file types which can be read by Logic and the EXS24. This section covers the use of the various ReCycle format files with the EXS24.

The ReCycle file types supported are listed below:

Provenance	Mac OS File Type
Old ReCycle File	RCSO
Old ReCycle Export File	REX
New (2.0) ReCycle File	REX2

To convert ReCycle files into EXS24 Sampler Instruments, simply browse to the desired function in the *Instrument > ReCycle Convert* menu of the EXS24 Instrument Editor. The individual functions are outlined in the “Recycle Convert” section, from page 555 onwards.

Velocity Factor

The velocity factor determines how the loudness of each “slice” of the imported ReCycle file affects the velocity values of the MIDI note generated to trigger it.

- If a positive value (up to 100) is entered, louder slices will generate MIDI notes with higher velocity values.
- The use of negative values on louder slices will result in the generation of lower MIDI note velocities.

Converting AKAI Files

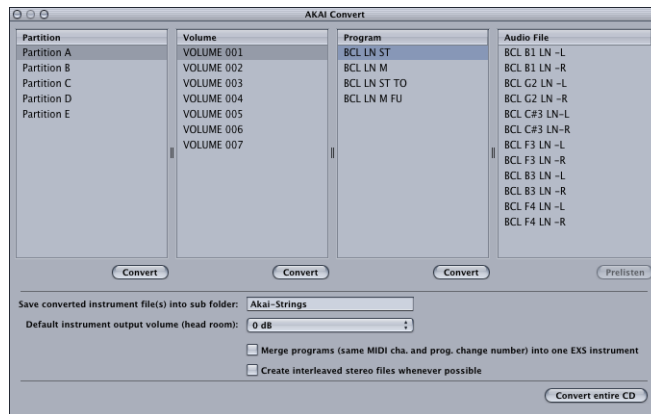
This section discusses the AKAI import procedure. The EXS24 can import samples saved in the AKAI S1000 and S3000 sample formats. The AKAI Convert function can be used to import:

- an entire AKAI format CD ROM
- an AKAI Partition
- an AKAI Volume
- an AKAI Program
- an Individual Audio File (sample)

These options have been provided to give you the most flexible and efficient method of dealing with your sample library. There may be a sample or two, or perhaps a particular drum kit which you would like to import from an AKAI CD-ROM.

Similarly, you may wish to import the contents of an entire CD-ROM in one simple operation, rather than spend the time dealing with individual Partitions, Volumes, Programs, and Audio Files.

This way, you can load and audition all of an AKAI CD-ROMs programs and files within Logic. Later, at your convenience, you can make use of your operating system's file management utilities to remove or reorganize your imported AKAI sounds, as discussed in "File Organization" on page 526.



To convert AKAI files

- 1 Select *Options > AKAI Convert*. This will launch a window similar to that shown above, with the text "Waiting for AKAI CD" spread across the four columns.
- 2 Insert an AKAI format sample disc into your CD-ROM drive and the AKAI Import window will commence reading the data. Following the reading of the CD-ROM, the display will update to show the contents of the CD-ROM. The *Partition* column will display information, with Partition A, Partition B (and so on) entries listed.

Note: Reading of a CD-ROM may take some time, dependent on the amount of sample data and file structure of the disc. In addition, the speed of the CD-ROM mechanism, bus speed, memory, and other factors can affect performance.

- 3 To view the contents of the Partitions, click once on the appropriate entry with the mouse button. This will display the *Volume* information contained within the *Partition*.
- 4 To continue through the architecture of the CD ROM, click on the *Volume* entries to view any *Programs* contained therein, and on the *Program* entries, to view the raw *audio files* (samples).
- 5 Once you have made your selection of Partition, Volume or Program, click on the *Convert* button beneath the appropriate column. The selected Partition, Volume or Program will be imported along with all associated audio files.
- 6 Any audio files imported will be stored within a folder which matches the name of the *Volume*. This folder is created within the *Logic > AKAI Samples* folder. The Sampler Instrument(s) created by the import procedure matches the Program name(s). It is placed inside the Sampler Instruments folder, or a sub folder as determined by the *Save converted instrument file(s) into sub folder* parameter discussed in “AKAI File Organization” on page 536.

Note: Should you wish to convert an entire AKAI CD ROM, click on the *Convert entire CD* button found to the lower right of the AKAI Sample Import window.

Sub-folders named after the Volume are created when converting a partition. If a Volume only contains one program, no sub-folder is created. Sub-folders named after the Partition are created when converting more than one Partition.

AKAI File Organization

In the following graphic, the *AKAI-Strings* folder contains several *Volumes*, which contain *Programs*.



The VOLUME 002 folder contains four patches—BCL PT M F, BCL PT M, BCL PT ST F, and BCL PT ST. The according Sampler Instruments are stored in the *Sampler Instruments > Akai-Strings > VOLUME 002* folder.

The audio files associated with these Sampler Instruments appear in the *AKAI Samples > VOLUME 002* folder.

When a Program is imported, these programs appear as Program.EXS in the Sampler Instruments folder.

Sampler Instrument management works with AKAI samples imported from CD ROM, in the same fashion as with other sample formats. Given the different file structures used by many AKAI format discs, however, you should take care to follow these guidelines.

- Create a shortcut to any folder on your hard disk/s which contains your AKAI sample library (or where you wish to store it). Name the shortcut “AKAI Samples” and all converted AKAI CDs/samples will automatically be saved in this destination folder. The “AKAI Samples” shortcut must be placed within the Sampler Instruments folder.
- If converting an entire CD ROM, you can create a shortcut with the sample CD’s name—“Dance MegaSynth” for example. This can be placed in the Sampler Instruments folder directly, or as a sub-folder within the AKAI Samples folder. The advantage with the second method is that all imported AKAI Instruments will be placed under the AKAI Samples sub-menu within the EXS24’s load window flip-menu.

Note: Assuming that an entire CD has been converted, you will find an *AKAI Samples* folder (which actually contains the raw sample data) and several *Partition* folders within the destination folder. The Partitions may contain several folders which bear the name of the imported instruments. The .EXS files (the EXS Instruments) may be contained in either the *Instrument* or *Partition* folders.

Additional AKAI Convert Parameters

Within the AKAI Convert window, you will find additional parameters listed below the four gray column areas. We will discuss these in their order of appearance.

Save converted instrument file(s) into sub folder.

Entering a name into this parameter field is achieved by clicking once with the mouse and typing in the desired name, followed by pressing *Return* or *Enter* respectively. In the example shown within “AKAI File Organization” on page 536, an *AKAI-Strings* folder was created.

All imported Volumes and Programs will automatically be added to this menu, and folder structure, until the name is changed. This facility may be useful, particularly when importing an entire CD, to create a folder name which reflects the CD-ROM’s name. Alternately, you may wish to use a category name, such as *Strings*. This way, any imported Programs or Volumes will be added to the *Strings* category.

Note: If an existing category name is used, the imported Sampler Instrument will be *added* to the folder/menu. It will *not* create a new menu entry/folder of that name.

Default instrument output volume (head room)

This parameter is extremely useful for many AKAI CD-ROMs. Please select this option before converting a CD-ROM.

- For drum CDs, select a headroom value of — 3 up to zero dB.
- For piano/string/pad CDs, a headroom value of — 9 dB is recommended, or the sound may/will clip with polyphonic use of these types of instruments.
- In cases where you're not sure of which headroom value to select, choose – 6 dB (average).

Merge programs (same MIDI cha. and prog. change number) into one EXS instrument

This parameter is *Off* by default. Its use is dependent on the structure of program material on the CD-ROM being imported.

To explain, many CD-ROMs created for AKAI samplers may feature several programs that contain single velocity layers for an instrument. AKAI samplers require the loading of an entire volume, or all necessary single programs, to be able to hear/play all velocity layers. All of these single programs are automatically assigned to the same MIDI channel and also react to the same MIDI program change number.

The EXS24 AKAI Conversion intelligently checks for these settings, and will build a single EXS Sampler Instrument out of multiple single programs. In general, this type of behavior is desirable with these types of CDs. When importing samples of this type, this option should be set to ON.

The same is true for drum CD-ROMs where single programs contain one instrument from a complete drum kit (kick/snare/hi-hat and so on as separate entities) You'll probably want these single AKAI programs to be merged into a single EXS Sampler Instrument as a full drum kit.

There are, however, a number of AKAI CD-ROMs where a single program of an AKAI Volume contains the entire instrument, and where other programs in the same Volume have the same MIDI channel and MIDI program change number preset. On this type of CD-ROM, use of the merge programs parameter is not desirable, and the option should be set to *OFF*.

Create interleaved stereo files whenever possible

This option should always be left enabled, as interleaved files offer better performance within the EXS24. When executing an AKAI conversion, some audio files are created as split stereo *and* as interleaved stereo files.

The detection of when it is possible to build an interleaved file is based on information stored with both the AKAI Program and audio files. Both the left and right files must have the same settings; otherwise they can not be used to create an interleaved file/multiple interleaved files.

Prelisten Function

The AKAI Import window features a *Prelisten* button, which is found below the Audio Files column. This facility allows you to individually audition AKAI audio files before deciding whether or not to import them.

To prelisten an audio file:

- 1 Select an individual file (sample) within the Audio Files column:
- 2 Press *Prelisten*. This will start playback of the selected audio file and the Prelisten button will update, with the word “Stop” appearing on the face of the button.
- 3 The selected audio file will loop continuously until you press the *Stop* button.

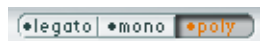
Vienna Library

The EXS24 features an additional interface for the Vienna Symphonic Library—Performance Set. The Performance Tool software provided by VSL needs to be installed to allow access to this interface. For details please refer to the VSL documentation.

Plug-in Window Parameters



Legato/Mono/Poly Buttons



These switches determine the number of voices used by the EXS24:

- When *Poly* is selected, the maximum number of voices is set via the numeric field alongside the *Poly* button. To change the value, click and hold with your mouse, and drag up or down to increase/decrease polyphony.
- When *Mono* or *Legato* is selected, the EXS24 is monophonic, and uses only one voice.
- In *Legato* mode, Glide is only active on tied notes. Envelopes are not retriggered when tied notes are played (single trigger).

In *Mono* mode, Glide is always active and the envelopes are retriggered by every note played (multi trigger).

Voices

This parameter determines the number of voices (polyphony) that the EXS24 is supposed to play. The *used* field indicates the number of voices that are actually used. If both fields tend to show the same value most of the time (probably causing a noticeable number of samples to drop out), you should set a higher *voices* value.

Unison Mode

This mode plays multiple EXS24 voices when each key is triggered:



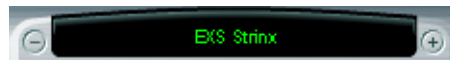
- In *Poly* mode, two voices per note.
- In *Mono* or *Legato* mode, you can adjust the number of voices per note with the *voices* parameter (this value is limited to 8—which is more than enough for fat unison sounds!)

The voices are equally distributed in the panorama field and are symmetrically detuned, dependent on the *Random* knob value.

Note: The number of voices actually used per note increases with the number of layered sample zones.

Sampler Instrument Selection Pull-Down Menu

This menu allows the selection and loading of a Sampler Instrument into your computer's RAM. In order to appear within this list, a Sampler Instrument must reside in the *Sampler Instruments* subfolder of Logic's program folder.



You will find plus (+) and minus (-) buttons to the left and right of the Instrument Load pull-down menu/display. These buttons allow you to browse to the next/previous Instrument (sound) of your sound library (if necessary, this will change folders in accordance with their order of appearance in the menu). Please note that the global *Next/Previous EXS Instrument* key and MIDI commands also perform the same function.

Edit Button

This button to the right of the Sampler Instrument selection pull-down menu opens the currently loaded Sampler Instrument in the EXS24 Instrument Editor. If none is loaded, the Sample Editor will open, allowing the creation of a new Sampler Instrument.

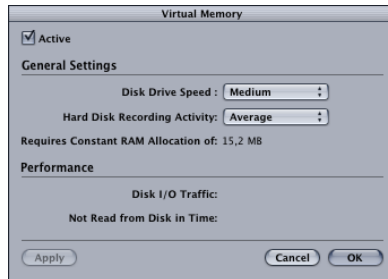


Options Button

Clicking the *Options* button launches a menu that offers the following options:

- *Recall default EXS24 settings* recalls a neutral setting for all parameters in the Plug-in window.
- *Recall settings from instrument* command manually recalls the original parameter settings of the loaded Sampler Instrument. This parameter is extremely useful if you've been over zealous with your tweaking.
- *Save settings to instrument* parameter stores the current settings of the Plug-in window into the Instrument file. When the Instrument is reloaded, these settings are restored in the Plug-in window.
- *Delete Settings from instrument* removes the stored settings from the Instrument.
- *Rename instrument* allows the renaming of the currently opened Sampler Instrument. When invoked, a file dialog box will open. This will overwrite the existing Instrument name.
- *Save instrument as* allows the storage of the currently opened Sampler Instrument under a different name. When invoked, a file dialog box will open.
- *Delete instrument* will delete the opened Sampler Instrument.
- *(Recall default EXS24 mkl settings)* does almost the same as the first entry, but the settings for the former version of the EXS are recalled for the selected Instrument, especially the former modulation paths (see "EXS24 mkl Modulation Paths" section, from page 552 onwards).
- *Extract MIDI-Region(s) from Recycle Instrument* allows you to extract the Regions contained in a Recycle Instrument. If no Recycle Instrument is selected, this option is not active.
- *AKAI Convert* launches the AKAI Convert window (see "To convert AKAI files" on page 535). This menu option accelerates working with AKAI samples, as you do not need to open the EXS24 Instrument Editor.
- *SoundFont Convert*
SampleCell Convert
DLS Convert
Giga Convert each will launch a dialog with instructions on performing these conversions. In order to play back long Gigasampler audio files, the Virtual Sample Memory option should be active (see below).
- *Preferences* opens a window with preferences for each Sampler Instrument (see "Preferences" section, from page 558 onwards).

- *Virtual Memory* opens a settings window for the EXS virtual memory functions. Virtual memory allows samples of almost unlimited length to be played back using streams that are fed directly from the hard disk. Switch off this option if you have enough RAM for your current work.



The *Active* checkbox switches virtual memory on or off. In the *General Settings*, you can set the *Disk Drive Speed* and the *Hard Disk Recording Activity*. The *Requires Constant RAM allocation of* field displays the memory usage required by the two parameters mentioned above. The *Performance* section contains two fields that show the current *Disk I/O Traffic* and the data *Not Read from Disk in Time*. Should these values rise to high levels you should change the *General Settings* to free up additional RAM for virtual memory use. The *Cancel* button rejects any changes made in the window.

Hold Pedal and Crossfades

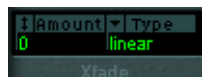
Hold via

This parameter determines the modulation source used to trigger the sustain pedal function (hold all currently played notes, and ignore their note off messages until the modulation source's value falls below 64). The default is controller number 64 (MIDI standard). You can change it if there are reasons to prevent Sustain from using CC 64, or if you wish to trigger Sustain with another modulation source.



Crossfade (Xfade)

Xfade allows you to crossfade between layered sample Zones with adjacent *Select Range* settings (*Select Range* was labeled *Velocity Range* in earlier versions). Please read the “Sample Select” section which follows.



Crossfades are controlled by two parameters:

Amount

This is the range of velocity (or other modulation source) values in which the crossfade takes place. The Select Range setting of all Zones will be expanded by this value, with the crossfade taking place in the expanded area. When the *Amount* parameter is set to 0, the EXS24 will switch between sample Zones in exactly the same fashion as earlier versions (Velocity Switching).

Type

You can choose between three different fade types for the velocity crossfade:

- dB linear
- linear (gain linear)
- equal power

Sample Select

This is just another modulation *Destination*—but it is important to know a little bit more about it. By default, Sample Select is controlled by velocity (via the default Velocity to Sample Select modulation path). The velocity value determines which of the layered Zones with different Select Range settings (velocity layers) is heard. You can also use modulation sources other than velocity—even multiple sources (in multiple modulation paths)!

If planning to do so, however, please keep in mind the fact that all sources (except Velocity and Key) cause all velocity layers to run simultaneously—using up as many voices as there are layered Zones. This also happens in cases where the Zones are not audible at the current control level.

If a continuous controller (such as the modulation wheel) is chosen, you can step through the velocity layers during playback. This is where the *XFade* parameter becomes important, as it allows smooth transitions between velocity split points.

Keep in mind that you can also combine velocity and modulation wheel control by using the Modulation Matrix (see “Modulation Matrix” on page 550).

Pitch Parameters



Tune

Offsets the pitch of the sample(s) in semitones by up to ± 2 octaves. The middle position of the knob (which can be set by clicking the small 0 button) leaves the pitch unaltered.

Transpose

This parameter allows you to transpose the EXS24. In contrast to the *Coarse Tune* parameter, *Transpose* not only affects the pitch, but also moves the Zones in accordance with the *Transpose* setting.

Random

This rotary knob controls the amount of random detuning which will apply to every played note. *Random* ranges from 0 to ± 50 cents.



You can use *Random* (detune) to simulate the tuning drift of analog synthesizers. This parameter can also be effective in emulating a natural feel for some stringed instruments.

Fine

Allows the EXS24 to be fine-tuned.

Pitch Bend Up (▲)

The amount of pitch bend (in semitones) that can be introduced by moving the pitch bend wheel to its maximum position.

Pitch Bend Down (▼)

The amount of pitch bend (in semitones) that can be introduced by moving the pitch bend wheel to its minimum position. When *Linked* is selected, the *Pitch Bend Up* value is used.

Remote

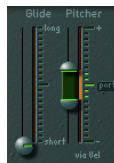
The *Remote* parameter allows you to easily pitch complete EXS24 Instruments in real-time. To do so, set the *Remote* parameter to the key of your MIDI Keyboard that you would like to use as the original pitch. All keys in a range of ± 1 octave around this key will now pitch the entire Instrument. This two octave range is similar to the Pitch Bend function, but is quantized to semitones.



Please note that the 2 octaves of remote keys don't actually trigger the instrument—they are used exclusively for semitone tuning.

Glide

The effect of this slider depends on the setting of the *Pitcher* slider: When *Pitcher* is centered, *Glide* determines the time it takes for the pitch to slide from one note to another (portamento). When the *Pitcher* parameter is set to a value above its centered value, *Glide* determines the time it takes for the pitch to glide down from this higher value back to its normal value. When *Pitcher* is set to a value below the centered value, the pitch glides from this lower setting back up to the normal value.



Pitcher

The *Pitcher* slider works in conjunction with the *Glide* slider: When the *Pitcher* is centered (which can be set by clicking the small *Port(amento)* button), *Glide* determines the portamento time. When *Pitcher* is set to a higher or lower value, a pitch envelope is activated. In this scenario, *Glide* determines the time it takes for the pitch to glide from the higher/lower *Pitcher* setting back to the original value. The *Pitcher* parameter can be modulated by velocity: the upper half of the slider determines the setting for maximum velocity, the lower half for minimum velocity. By clicking and dragging in the area between the two slider segments you can move both simultaneously.

Please note that the upper half of the *Pitcher* slider can be set above the center position, and the lower half below the center position. When the *Pitcher* sliders are set in this fashion, lower velocity values cause the pitch to rise from the lower setting back to the original note pitch, while higher values cause it to fall from the higher setting back to the original value. In other words: the polarity of the pitch envelope can be changed according to velocity values.

When both halves of the pitcher slider are set below or above the centered position, either a low or high velocity will slide up/down to the original pitch. Dependent on the position of the upper/lower halves of the slider in relation to the center position, the time required for the slide up/down to the original note pitch can be adjusted independently for both soft/hard velocities.

Filter Parameters



Filter On/Off Switch

This button switches the filter section on or off. Please note that the knobs and buttons in the silver panel area and the Filter Envelope are only active when the filter is turned on. When the filter section is turned off, the EXS24 is far less CPU-intensive.

Lowpass (LP)

The *Lowpass Filter* offers four different settings for its cutoff steepness: 24 dB (4 pole), 18 dB (3 pole), 12 dB (2 pole), and 6 dB (1 pole). The 24 dB setting can be used for drastic sweep effects, such as cutting off all but a few notes, or for the creation of ultra-deep bass sounds with just the necessary amount of overtones. The slope setting of 6 dB per octave is very useful in cases where you want a slightly “warmer” sound, without drastic filter effects—to smooth “overly bright” samples, for example. The two remaining values may be used for any purposes.

Fat (Fatness)

The Fatness mode is separate from the slope setting, and can be used with all available slope values. Fatness preserves the bass frequency response, even when high *Resonance* settings are used. Please note that this only applies to Lowpass filters. *Fatness* is non-functional when used in conjunction with the *High* or *Bandpass* filters.



Highpass (HP)

The *Highpass Filter* is a 2 pole (12 dB/Oct.) design. A Highpass filter reduces the level of frequencies that fall below the cutoff frequency. It is useful for situations where you would like to suppress the bass and bass drum in a sample, for example, or for creating classic highpass filter sweeps.

Bandpass (BP)

The *Bandpass Filter* is a 2 pole (12 dB/Oct.) design. A Bandpass filter only allows the frequency bands directly surrounding the cutoff frequency to pass. Frequencies which fall outside these boundaries will be cut.

Drive

This knob allows the filter input to be overdriven. Turning *Drive* up leads to a more dense and saturated signal, with additional harmonics being introduced/becoming audible.

Cutoff

The cutoff frequency of the lowpass filter. As you turn this knob to the left, an increasing number of high frequencies are filtered from the signal. The *Cutoff* value also serves as the starting point for any modulation involving the filter.

Resonance

Turning up *Resonance* leads to an emphasis of the frequency area surrounding the frequency defined by the *Cutoff* parameter. Very high *Resonance* values introduce self oscillation, and cause the filter to produce a sound (a sine wave) on its own.

Simultaneous Control of Cutoff and Resonance



By clicking and dragging on the chain symbol located between the *Cutoff* and the *Resonance* knobs, you can control both parameters simultaneously: vertical mouse movements alter *Cutoff*, and horizontal mouse movements affect *Resonance* values.

Key

This knob defines the amount of filter cutoff frequency as determined by note number. When *Key* is fully turned to the left, the cutoff frequency is not affected by the note number, and is identical for all notes played. When *Key* is set fully right, the cutoff frequency follows the note number 1:1—if you play one octave higher, *Cutoff* is also shifted by one octave. This parameter is very useful in avoiding overly filtered high notes.

Volume and Pan Parameters

Level via Vel

Controls the volume of the sound. The *Level* parameter can be modulated by velocity: the upper half of the slider determines the volume for maximum velocity, the lower half for minimum velocity. By clicking and dragging in the area between the two slider segments, you can move both simultaneously.

Volume

The main volume parameter for the EXS24. Move this knob to find the right balance between avoiding distortion and getting the best (highest) resolution in the channel fader and the *Level via Vel* slider.



Key Scale

This parameter modulates the sound's level by note number (position on the keyboard). Negative values increase the level of lower notes. Positive values increase the level of higher notes.

Amp Envelope (ENV 2)

This is an ADSR envelope generator for controlling the sound's level over time. It offers *Attack*, *Decay*, *Sustain*, and *Release* parameters.



The attack time can be reduced by velocity: the upper half of the slider determines the time for minimum velocity, the lower half for maximum velocity. By clicking and dragging in-between the two slider segments, both can be moved simultaneously.

LFO Parameters



LFO 1 EG

This knob allows LFO 1 to be faded out (*Decay* area) or faded in (*Delay* area). In the centered position (which can be set by clicking on the small 0 button), the LFO intensity is constant.

LFO 1 Rate

This is the frequency of LFO 1. It can be set in note values (left area), or in Hertz (right area). In the centered position (which can be set by clicking on the small 0 button), the LFO is halted and generates a constant modulation value at full level (*DC* = Direct Current).

This allows you to perform a nice trick: Set up an LFO to modulate, say, *Cutoff*, with the modulation wheel controlling the LFO's intensity. Then set the LFO's rate to DC. As the LFO's modulation intensity is controlled via the modulation wheel, you can now make use of the modulation wheel to manually open the filter.

Waveform for LFO 1 and LFO 2

These two switches allow the selection of the waveform type used by LFO 1 and LFO 2. A selection of Triangle, falling and rising Sawtooth, Square up and Square down, a random stepped waveform, and a smoothed random waveform is available for each LFO.

LFO 1 is a polyphonic LFO with key synchronization. This means that when LFO 1 is used, each voice of the EXS24 has its own discrete LFO. When a note is played, the LFO corresponding to that voice starts its cycle. This scheme means that the LFO cycles of each voice played are not synchronous, and operate independently of each other. This opens up a range of modulation possibilities. As an example—the LFO of one voice could generate the maximum modulation value, while the LFO assigned to another voice could output its minimum value. This extremely flexible approach can result in some very lively modulations.

In contrast, LFO 2 is a monophonic LFO without key synchronization. This means that LFO 2 runs continuously, and is not restarted by the triggering of a new note. All voices are modulated by the sole LFO, so the degree of modulation at any given time is the same for all voices. This results in a rather synthetic-sounding modulation.

Use these different characteristics to tailor the sound to your needs.

LFO 2 Rate

The frequency of LFO 2. It can be set in note values (left area), or in Hertz (right area). In the centered position (which can be set by clicking on the small 0 button), the LFO is halted, and generates a constant modulation value with full level (DC = Direct Current). Again, don't overlook this feature if you want to control an LFO-modulated parameter directly via the Modulation Matrix (see following section).

LFO 3 Rate

There is a third LFO available which always uses a triangular waveform. LFO 3 can oscillate freely between 0 and 35 Hz, or can be tempo synchronized in values between 32 bars and 1/128 triplets.

Modulation Matrix



The Modulation Matrix is the dark horizontal band that spans the center of the EXS24 interface. It consists of ten modulation paths, each linking a modulation *Source* with a modulation *Destination* (the sound parameter that you want to modulate). This is similar to the use of patch cords on modular synthesizers, but with the additional option of control over the modulation amount via another modulation source.

Creating a new modulation path is easy:



- first, choose the Destination (*Dest*)
- then choose the Source (*Src*)

The green triangular fader on the right side of each modulation path allows you to set the modulation depth with a bipolar range (positive or negative value).

In this example, the LFO 1 *Speed* is modulated by channel pressure (aftertouch) messages of a MIDI keyboard.



You have the option of inserting another modulation source in the middle slot labeled *via*. In this scenario, the green triangular fader will divide, allowing you to set a range for modulation depth. The size of the modulation range depends on the possible values allowed by the *via* modulation source.

In our example, the key number of the MIDI keyboard (*Key*) determines how strongly channel pressure controls the *Speed* of LFO1. More experienced users would read the picture like this: pressure to LFO1 Speed via key number.



Inverting Sources

You can also invert the direction of the source's effect on modulation depth by clicking the *inv* button (right of the word *Src* or *via*), depending on which of the sources you would like to invert.

In this example, we inverted the *via* modulation source. You can see how the green and orange triangles have swapped positions. The orange triangle always marks the modulation depth for the maximum value of the *via* source, while the green triangle always marks the modulation depth if the *via* source is at its minimum value. They are reversed by inverting the modulation.



Bypassing Modulation Paths

You can temporarily disable the entire modulation path with the *b/p* button, found alongside the word *Dest*.



In our example, both modulation sources—*Pressure* and *Key*—are disconnected from the *LFO1 Speed* modulation destination. Clicking the *b/p* button a second time reconnects the modulation path, restoring the old modulation depth settings.

Second Order Modulations

The EXS24 also allows the use of second order modulation destinations (such as envelope times, LFO speeds and so on)—functionally outperforming many analog synthesizers. To explain:

- The same *source* can be used as often as desired to control different *destinations*.
- The same *destination* can be controlled by different *sources*. The different input values are accumulated.

EXS24 mkl Modulation Paths

Many of the hard-wired modulation paths that were available as sliders on the original EXS24 (mkl) are now part of the Modulation Matrix. In order to reconstitute the modulation slider configuration of the mkl version, click on the *options* button in the upper-right corner and choose (*Recall default EXS24 mkl settings*) from the pop-up menu. This will load the mkl modulation paths into the Modulation Matrix, as follows:

- Velocity to Sample Select
- LFO 1 to Pitch via ModWheel (= Ctrl#1)
- Velocity to Sample Start (inv)
- LFO 2 to Filter Cutoff via ModWheel
- Velocity to Filter Cutoff
- Envelope 1 to Filter Cutoff via Velocity
- LFO 2 to Pan via ModWheel

You can, of course, freely alter the settings of these modulation paths. To exchange modulation sources with sources that were not available in EXS24 mkl, for example (see the complete list of sources and destinations below).

EXS24 mkl Backward Compatibility

For technical reasons, the settings of the Modulation Matrix can not translate backwards to the EXS24 mkl. If you are a sound designer and want your products to be compatible for Logic 4.x users, we recommend the use of Logic version 5.3 (Mac OS X: 5.4) with the EXS24 mkl.

Modulation Sources	Modulation Destinations
Sample Select	Side Chain (level)
Sample Start	Maximum
Glide Time	Env1
Pitch	Env2 (Amp)
Filter Drive	LFO 1
Filter Cutoff	LFO 2
Filter Resonance	LFO 3
Volume	Release Velocity
Pan	Pressure
Relative Volume	Pitch Bend

Modulation Sources	Modulation Destinations
Relative Volume (auto adjust)	Key
LFO1 Dcy./Dly	Velocity
LFO1 Speed	Control Nr. 1
LFO2 Speed	...
LFO3 Speed	Control Nr. 120
Env1 Attack	
Env1 Decay	
Env1 Release	
Time	
Env2 Attack (Amp)	
Env2 Decay (Amp)	
Env2 Release (Amp)	
Hold	

Note: Controllers 7 and 10 are marked as (not available). Logic uses these controllers for volume and pan automation of the audio object. Controller 11 is marked as (Expression). It has a fixed connection to this functionality, but it can also be used to control other modulation sources.

Multiple Outputs

The EXS24 mkII offers five stereo pairs and six individual mono outputs. The first ten EXS24 outputs are always configured as stereo pairs. Outputs 11 through 16 are mono outputs.

The main output pair (1–2) of the EXS24 is assigned to its Audio Instrument channel. Additional outputs, starting with output 3, are accessed via the Aux channels. Please refer to the paragraph on Aux channels below.

Individual output assignments may be set for *Groups*, or *Zones*, within an EXS24 Instrument. If zones are assigned to their individual outputs, any existing group outputs for the zones will be ignored. All output assignments are stored with the EXS24 Sampler Instrument.

If the EXS24 is inserted as a regular (not multi channel) mono or stereo plug-in, all multiple output assignments stored with a Sampler Instrument are automatically routed to the main stereo output pair, or mono output, respectively.

The multiple outputs of the EXS24 are managed intelligently: Multiple outputs accessed via the Aux channels are automatically subtracted from the main output. Conversely multiple outputs that are not accessed via Aux channels are automatically routed to the main stereo output pair.

Aux Channels for Instrument Plug-ins with Multiple Outputs

By default, all outputs of a “Multi Channel” Instrument plug-in are summed and routed to the stereo output of the respective Audio Instrument channel at first. Signals of individual outputs (starting with output 3) are automatically subtracted from the stereo output sum after they have been assigned to Aux channel inputs.

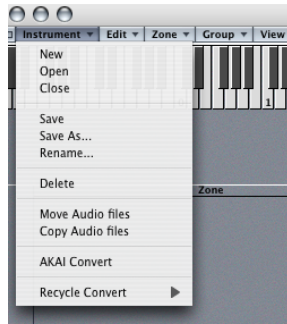
Outputs 1 and 2 are always assigned to the respective Audio Instrument. This assignment is fixed and can’t be changed.

Note: Additional instrument outputs are only available to plug-ins inserted into an Audio Instrument channel via the *Multi Channel* sub menu. Aux channel Input sources—being sent from suitable Instrument plug-ins—start from output 3.

Instrument Editor Parameters

Instrument Menu

New



Creates a new, empty Sampler Instrument. In order to load a sample, you will need to create a new Zone (see below).

Open

Allows an existing Sampler Instrument to be loaded into the Instrument Editor.

Close

Closes the currently opened Sampler Instrument. If you have made any changes to the Instrument, you will be asked if you would like to save them.

Save

Saves the currently loaded/edited Sampler Instrument. When you create a new Instrument and save it for the first time, you will be asked to give it a name. If you have edited an existing Sampler Instrument and save it via this command, the existing file name is used and the old version is overwritten.

Save As...

This command also saves the currently loaded/edited Sampler Instrument. When *Save As...* is used, you will be prompted to give the file a name. Use this command when you want to save a copy of an edited Sampler Instrument, rather than overwriting the original version.

Rename

This command allows you to rename the loaded Sampler Instrument. The renamed version replaces the previous version on the hard disk.

Delete

This deletes the currently opened Sampler Instrument.

Move Audio files

Moves the audio files of the selected Sampler Instrument into a desired folder location. Use of this option will launch a standard operating system file navigation/browse utility. You may browse to an existing folder or enter a new name, as desired. If no folder name is entered, a new folder will be created which matches the Instrument name, and all audio files will be moved into this folder.

Copy Audio files

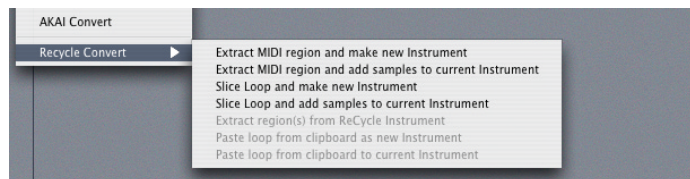
Operation is as per the *Move Audiofiles* function, but files are duplicated, rather than moved, to the nominated folder. This facility should be used as part of your working methods when creating Logic songs, as discussed in “Saving of Project-related EXS24 Instruments” on page 528.

AKAI Convert

Opens a window that allows you to convert files from an AKAI format CD ROM (see “To convert AKAI files” on page 535).

Recycle Convert

There are different options here (please also read “Converting ReCycle Files to EXS24 Instruments” on page 534):



Extract MIDI region and make new Instrument

Use of this menu option will launch a file selection dialog, allowing the selection of a ReCycle file.

- Following selection, a new EXS24 Instrument with a name which matches that of the ReCycle loop will be created. Should an EXS24 Instrument of that name already exist, a # sign and a number will be appended, ensuring that the filename is unique within the *Sampler Instruments* folder.
- You will be prompted to enter a velocity factor (see page 534). This should be left at the default value of zero in most cases.
- Press the OK button (after entering a value, if applicable). The EXS24 will generate a number of Zones (one for each “slice” of the imported ReCycle file) and one Group to which the Zones are assigned.
- In addition, a MIDI Region is generated on the currently selected track, at the current song position (rounded to whole bars). This MIDI Region is used to trigger the imported “slices” at the timing defined by the ReCycle file. You can generate new MIDI Regions at any time from the imported EXS24 Instrument (see below), so feel free to modify or delete it.

Extract MIDI region and add samples to current Instrument

This option allows the addition of a ReCycle loop to any EXS24 Instrument currently opened in the Instrument Editor. This permits the use of several different loops in a single Sampler Instrument, which can be recorded and played on a single Audio Instrument track. If no Instrument is open in the Instrument Editor, this function behaves identically to the *Extract MIDI region and make new instrument* function.

Slice Loop and make new Instrument/Slice Loop and add samples to current Instrument

The first option creates a new EXS24 Sampler Instrument while the second option adds the Zones (from the sliced loop) to the currently active Sampler Instrument.

The Recycle slices are rendered into a single audio file (the whole Recycle loop), played back at the current Logic song tempo. Rather than importing and playing back a single discrete Recycle slice, each Zone will play back the Recycle loop to its very end, starting with the slice points originally assigned to the respective Zones. To explain: the lowest Zone will play back the entire loop, and the highest zone will only play the last slice of the loop.

This option allows for loop trigger techniques often used in “old school” Drum’n’Bass tracks, with the sample loop start point determined by playing the respective notes on the keyboard.

Extract Region(s) from ReCycle Instrument

This option generates MIDI Regions from EXS24 Instruments which were originally converted from ReCycle file(s). The MIDI Regions are created on the currently selected track, at the current song position (rounded to bars). A single MIDI Regions is generated for each imported ReCycle loop in the currently open Instrument. If no imported ReCycle loop exists in the currently open Instrument, this menu option is disabled. This function will also ask for a velocity factor (see page 534).

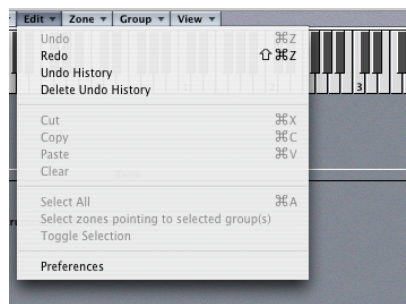
Paste loop from clipboard as new Instrument

Functionally identical to the *Extract MIDI region and make new instrument* option with the contents of the Clipboard used, rather than an existing ReCycle file. A loop must have been copied into the Clipboard via ReCycle's *Copy loop* facility for this to function.

Paste loop from clipboard to current Instrument

Functionally identical to the *Extract MIDI region and add samples to current instrument* option with the contents of the Clipboard used, rather than an existing ReCycle file. A loop must have been copied into the Clipboard via ReCycle's *Copy loop* facility for this to function.

Edit Menu



Undo

Allows the most recent change to the Sampler Instrument to be undone.

Redo

Undoes the last Undo command.

Undo History...

Opens the Undo History. Here you can undo or redo arbitrary operation steps. More information can be found in your Logic manual.

Delete Undo History

Deletes the Undo History. More information can be found in your Logic manual.

Cut, Copy, Paste

The standard commands for cutting, copying, and pasting values. In addition to values you may also cut, copy, and paste selected Zones and Groups.

Note: When multiple Zones and Groups are cut, copied, or pasted simultaneously, the Group assignments of the Zones are retained.

Clear

Deletes the currently selected Zone or Group. Clear can be undone with the *Undo* command.

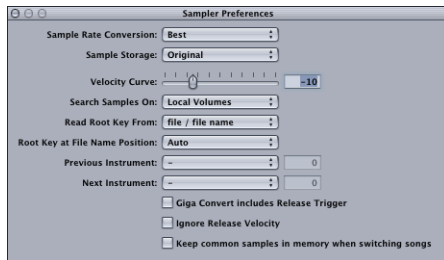
Select All

Selects all Zones and Groups of the loaded Sampler Instrument.

Select Zones pointing to selected group(s)

This command automatically selects all Zones that point to one or multiple selected Groups.

Preferences



The Preferences window allows you to:

Choose the interpolation quality used by the EXS24. When *Sample Rate Conversion* is set to *Best*, the highest possible sound quality is maintained when transposing samples. It should be noted that this option requires additional CPU cycles over the *Normal* setting, which will be adequate in most cases.

Select the format in which the EXS24 handles the loaded sample data via the *Sample Storage* parameter. When set to *Original*, the samples are loaded into RAM at their original bit depth, and are converted to Logic's internal 32 Bit floating point format on playback. When *32 Bit Float* is selected, the samples are stored and loaded in this format. This eliminates the need for any realtime conversion, meaning that the EXS24 can handle the sample data more efficiently and can play back more voices simultaneously. It should be noted that this requires twice as much RAM for 16 bit samples, and a third more RAM for 24 Bit samples.

The *Velocity Curve* parameter determines the EXS24's responsiveness to velocity values received from your MIDI keyboard. Negative values increase the response to soft key strikes, and positive values decrease it.

The *Search Samples On* parameter determines the location that instruments samples should be searched in. You may either choose the drives normally used by the operating system or external SCSI, FireWire or USB drives, accessible directly or over a network.

Drives can be selected individually, or grouped as follows:

- *Local Volumes* internal storage media (hard disks and CD ROM mechanisms) attached to or installed in the computer directly.
- *External Volumes* storage media accessible over a network.
- *All Volumes* both internal and network media are scanned for appropriate data.

Note: Selecting External or All Volumes may result in a dramatic increase in the time required by the EXS24 to find and load Sampler Instruments and files.

The following two preferences are particularly useful when used in conjunction with the function discussed in the "Load Multiple Samples" section, from page 561 onwards.

Read Root Key From:

- *file/filename*—initially reads information about the root key from the file itself (in the header of the AIFF or WAVE file) when loading an audio file into a zone. If no information of this type exists in the file header, a smart analysis of the filename may detect a root key. If this second method doesn't provide any useful results, C3 will be used as the default root key in the zone.
- *filename/file*—as above, but vice versa, with the filename read first, and the header read second.
- *filename only*—reads from the filename only. If no root key information exists, C3 will automatically be assigned to the zone as the root key.
- *file only*—reads from the file header only. If no root key information exists, C3 will automatically be assigned to the zone as the root key.

Root Key at File Name Position

When loading an audio file into a zone this option is used for the analysis of a root key in the file name. Possible options in the pull-down menu are "Auto", or numerical values from 1 to 30.

- *Auto* is the recommended value. It provides a smart analysis of numbers and keys in the file name. A number in the file name can be recognized, regardless of its format—"60" or "060" are both valid. Other valid numbers can range between 21 and 127. Numerical values outside of these are generally just version numbers. A key number is also a valid possibility for this use—"C3", "C 3", "C_3", "A-1", "A -1" or "#C3", "C#3", for example. The possible range is "C-2" up to "G8".

Note: There may be cases where a sound designer has used multiple numbers in a filename, which is common with loops, with one value being used to indicate tempo—“loop60-100.wav”, for example. In this situation, it isn’t clear which, if either of the numbers, indicates a root key or something else: 60 or 100 could indicate the file number in a collection, tempo, root key, and so on. You can set a value of “8” to read the root key at position (letter/character) eight of the filename—namely the 100 (E6). Alternately, setting a value of “5” will select the 60 (C3) as the root key position.

Previous/Next Instrument

Previous/Next Instrument and the number fields alongside determine which MIDI event type (and data value) will be used for selection of the previous or next Instrument: *Note, Poly Pressure, Control, Program, Channel Pressure, Pitch Bend*.

In the number field (depending on the event type), either the Note Number or the value of the first data byte can be entered. When *Control* is selected, the number field determines the Controller number.

Giga Convert includes Release Trigger

Determines whether or not the Release Trigger function of the Gigasampler format will be performed by the EXS. This is important if you want to stay compatible with the EXS24 mkl, which doesn’t offer this functionality.

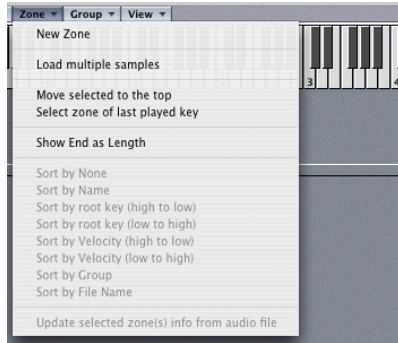
Ignore Release Velocity

This option also refers to the Release Trigger function and should always be set to on for this purpose. Regardless of whether or not your keyboard is able to send Release Velocity, you would want your samples played by the Release Trigger function to be louder or softer than the original Sample, or at the same volume, regardless of the initial velocity. When playing with Release Trigger, you would want the Release Velocity value to have the same value as the Initial Velocity value. To accomplish this, you can switch off Release Velocity.

Keep common samples in memory when switching songs

Determines whether or not the samples commonly used by two open song files are reloaded when switching between songs.

Zone Menu



New Zone

Creates a new Zone in the currently loaded Sampler Instrument.

Load Multiple Samples

Allows several samples to be loaded in one operation. The Instrument Editor reads the key note from samples, and places the samples into new Zones. These Zones are automatically created. The key note is located in the middle of the Zone, with Zone borders being determined by the Zone borders of neighboring Zones. Zones with identical key notes will be layered. A root key is automatically determined by use of the settings described in the “Preferences” section, from page 558 onwards.

Three automatic mapping functions are available when loading multiple samples:

- *Auto map (default)* uses the key note information (root key) stored with the audio files and places the samples (as zones) over the keyboard range. The number of keys that constitute a zone is intelligently determined by the placement of neighboring zones.
- *Drums* uses the key note information (root key) stored with the audio files. Each zone contains a single key on the keyboard determined by the key note information.
- *Chromatic* ignores all key note information (root key) of the audio files and places the samples on the keyboard in chromatic order, starting at C1.

Move selected to the top

When this option is activated, the parameter window of a selected Zone is moved to the top of the onscreen Zone listing, and is automatically opened.

Select zone of last played key

When active, this menu option allows you to switch between Zones by pressing a key on the onscreen keyboard, or via a connected MIDI keyboard.

Show End as Length

When this option is activated, the sample length is shown in the Zone’s parameter window, rather than the end point value.

Sort by...

These commands determine the order in which the Zone windows are displayed:

Sort by None: The Zones are shown in the order in which they were created.

Sort by Name: The Zones are sorted alphabetically.

Sort by root key (high to low): The Zones are sorted according to their *Root Key* settings; the higher key notes are displayed at the top of the list.

Sort by root key (low to high): The Zones are sorted according to their *Root Key* settings; the lower key notes are displayed at the top of the list.

Sort by Velocity (high to low): The Zones are sorted according to their *Velocity Range* settings; the higher velocity ranges are displayed at the top of the list.

Sort by Velocity (low to high): The Zones are sorted according to their *Velocity Range* settings; the lower velocity ranges are displayed at the top of the list.

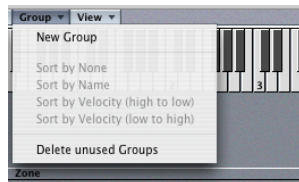
Sort by Group: The Zones are shown in the order of their Group assignment(s).

Sort by File Name: The Zones are sorted alphabetically according to the names of their associated audio files.

Update selected zone(s) info from audio file

This option is used when editing loops in an external sample editor. It reads loop settings from the audio file and updates the settings of the Zone accordingly. In addition, this feature also validates the loop's length and start position.

Group Menu



New Group

Creates a new Group.

Sort by...

These commands determine the order in which the Group windows are displayed:

Sort by None: The Groups are shown in the order in which they were created.

Sort by Name: The Groups are sorted alphabetically.

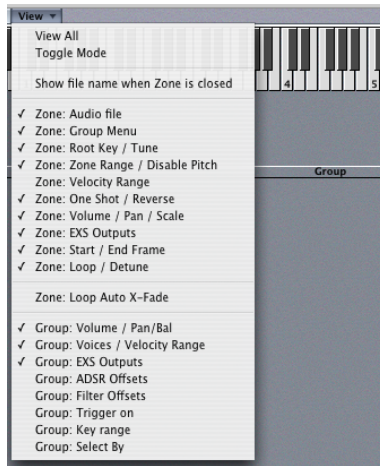
Sort by Velocity (high to low): The Groups are sorted according to their *Velocity Range* settings; the higher velocity ranges are displayed at the top of the list.

Sort by Velocity (low to high): The Groups are sorted according to their *Velocity Range* settings; the lower velocity ranges are displayed at the top of the list.

Delete unused Groups

This command deletes all Groups that do not have a Zone assignment. This deletion can be undone with the *Undo* command.

View Menu



This menu offers display options for the Zone and Group windows:

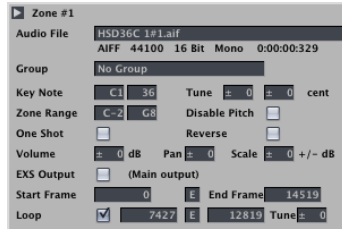
- *View All* shows all available parameters. When this option is activated, you can manually prevent certain parameters from being displayed by deselecting the corresponding option from the menu.
- When *Toggle Mode* is on, only one parameter will be displayed at a time. You can select the desired parameter by selecting the corresponding option from the menu.
- When *Show filename when Zone is closed* is activated, the names of loaded audio files are shown in the title bars of closed Zone windows.

The following menu entries allow you to select the parameters you wish to be displayed.

Note: When working with the Zone and Group windows, it can be useful to close or open all windows at once. To do so, click on one of the triangles while pressing Option.

Note: To set a switch parameter for a number of (selected) Zones simultaneously, hold Command when clicking the switch.

Zone Parameters



Zone Name

New Zones are numbered consecutively. Double-clicking on a Zone's number allows you to enter a name instead.

Audio File

A file selection dialog can be opened by clicking on the gray field next to the *Audio File* label. This will allow you to select and load a sample into a Zone. When loaded, the sample's name is displayed in the gray field. Additional information about the sample is displayed beneath its name (sample format, sampling frequency, bit depth, mono/stereo status and sample length).

Group

Group allows you to assign a Zone to an existing Group. When *No Group* is selected, the Zone is only affected by the parameters of the Plug-in window.

Key Note/Tune

Key Note allows you to determine the note at which the sample will sound with its original pitch. The *Key Note* is displayed as both a note name and a numerical value. The *Tune* and *cent* fields allow coarse and fine-tuning of the sample in semitone and cent increments.

Zone Range/Disable Pitch

The two *Zone Range* parameters allow you to define a key range for the Zone. When *Disable Pitch* is activated, the sample is always played at its original pitch, regardless of the note number.

Velocity Range

Activation of the *Select Range* option, in conjunction with the use of the two *Velocity Range* fields allows you to define a velocity range for the Zone.



One Shot/Reverse

Activating *One Shot* causes the Zone to ignore the length of notes used to trigger the sample—the sample is always played to the end. This option is useful for drum samples, where you often don't want the MIDI note length to affect sample playback. *Reverse* plays the sample from its end to its beginning. This option works non-destructively, leaving the audio data in the sample unchanged.

Volume/Pan/Scale

Volume adjusts the volume of the Zone.

Pan adjusts the pan position of the Zone. This parameter only works when the EXS24 is used in stereo.

Scale—Negative *Scale* values make notes lower than the note position defined by the *Key Note* sound louder than higher ones; positive values have the opposite effect. Use this parameter for balancing the volume of a sample across the selected key range.

EXS Output

This parameter determines the outputs used by the Zone. Choices include the main outputs, and paired channels 3 and 4, 5 and 6, 7 and 8, 9 and 10, or individual outputs 11 through to 16. This allows individual zones to be routed independently to Aux channels in a multi channel EXS instance.

Start/End Frame

The *Start Frame* and *End Frame* parameters set the sample's start and end points, respectively. Clicking on the small *E* button between the two values will launch Logic's Sample Editor, allowing you to set the start and end points graphically.



Loop

The loop parameters become visible when this option is activated, and the sample will loop when sustained MIDI notes are received.

Loop Start, Loop End—You can define discrete loop start and end points in these fields, allowing you to cycle (loop) a portion of the audio file. Data entry is via the mouse as slider, or by double-clicking and directly typing in a value. Clicking on the small *E* button between the two values will launch Logic's Sample Editor, allowing you to set the loop start and end points graphically: Loop Start is represented by the LS marker and Loop End, by the LE marker.

Tune—This parameter allows the tuning of the looped portion of the audio file to be different to that of the non-looped portion. It is adjustable in cent increments (± 50 cents).

Auto Crossfade

In a crossfaded loop, there is no hard “cut” between the loop end and loop start points. Rather, the loop end and start points are crossfaded for a smooth transition. This is especially convenient with samples that are hard to loop, and would normally produce clicks at the transition point—the “join” in the loop.

The *Auto Crossfade* field allows a pre-determined value (in milliseconds) to be used as a default when the option is enabled. The higher the value, the longer the crossfade, and the smoother the transition between the loop end and start points.

EqPower—*EqPower* allows you to enable an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This will fade out/fade in the joined portions of a loop at an equal volume level.

Note: The “perfect” settings for the crossfade parameters depend on the sample material. A loop which cycles reasonably smoothly is the best starting point for a perfect crossfade loop, but a crossfaded loop does not always sound better. Just experiment a little with the parameters, and you’ll soon find out how, when, and where they work best.

Group Parameters



Volume/Pan/Bal

Volume—Adjusts the volume of the Group, and therefore the volume of all assigned Zones, simultaneously.

Pan/Bal—Adjusts the pan position of the Group (stereo balance for stereo samples), and the pan position of all assigned Zones simultaneously.

EXS Output

This parameter determines the outputs used by the Group. Choices include the main outputs, and paired channels 3 and 4, 5 and 6, 7 and 8, 9 and 10, or individual outputs 11 through to 16. This allows individual Groups to be routed independently to Aux channels in a multi channel EXS instance.

Voices/Velocity Range

Voices—Determines how many voices the Group is allowed to use. This parameter is discussed in the “Multiple Zones and Groups” section, from page 525 onwards.

Select Range—The two *Velocity Range* parameters allow you to set up a velocity range for the Group. The settings made here override the settings in the Zones, if necessary: When a Zone's velocity range is larger than that allowed by the Group setting, the Zone's velocity range is limited by the Group setting.

Trigger on

Key Down (default)—Zones pointing to this group are triggered on key down.

Key Release—Zones pointing to this group are triggered on key release. This allows you to trigger additional samples on key release. This is useful for emulating organ key clicks, for example.

Select by

If this option is enabled, you can define a specific key. Whenever the defined key is pressed, zones pointing to this group are played by ensuing MIDI note data, and other groups (selected by a different key) are not played. The defined key is not played itself, it simply acts as a remote.

This feature also works with other MIDI events (Controllers, Bends), and Groups.

As you select the different menu options, the display will update to allow the entry of the desired continuous controller number (CC 19, for example) and range, or Group selection, as appropriate.

Key Range

The *Key Range* allows you to define a key range within the Group that will be affected by this functionality.

ADSR Offsets

The amp(litude) envelope settings from the Plug-in window can be offset separately for each Group by these parameters. Each time parameter has an offset range of ± 9999 ms, the sustain level can be varied by $\pm 50\%$. The offset fields are for *Attack*, *Decay*, *Sustain*, and *Release*—from left to right.

Filter Offsets

The *Cutoff Offset* setting of the Plug-in window can be offset separately for each Group ($\pm 50\%$).

The *Resonance Offset* setting of the Plug-in window can be offset separately for each Group ($\pm 50\%$).

EXS24 Key Commands

A number of key commands are available for the EXS24 which accelerate editing in Logic, and provide additional functionality. They are found in the Key Commands window.

These key commands have no default keyboard assignments, so you will need to create them, should you wish to take advantage of these shortcuts and facilities. Please consult your Logic reference manual for information on accessing the Key Commands window and on the assignment of keyboard shortcuts to functions.

Previous Instrument

Selects the previous Instrument (when multiple Instruments are opened for editing) allowing you to quickly switch between several Instruments. An Instrument must be selected for this function to work.

Next Instrument

Selects the following Instrument (when multiple Instruments are opened for editing) allowing you to quickly switch between several Instruments.

The *Next/Previous Plug-In Setting* or *EXS Instrument* key commands are also available. They perform the same functions as above, but only in the topmost window. If the EXS is the topped window, the key command will select the next/previous Instrument. If another Plug-in window is selected, the key command will select the next/previous plug-in Setting file.

Select zones pointing to selected group(s)

Following the selection of a Group, use of this function will select all associated Zones, ensuring that any edits made are only performed on Zones associated with the selected Group.

Previous Zone/Group

Selects the previous Zone/Group (when multiple Groups/Zones exist) allowing you to quickly switch between them. A Zone or Group must be selected for this function to work.

Next Zone/Group

Selects the following Zone/Group (when multiple Groups/Zones exist) allowing you to quickly switch between them. A Zone or Group must be selected for this function to work.

New Zone

Creates a new Zone.

New Group

Creates a new Group.

View: All/Toggle Mode

Toggles between viewing all parameters in Zones and Groups and a limited view which displays the Audio File name in Zones and the Volume/Pan parameters in Groups.

View: Next Zone Parameter

This key command is designed to aid in the adjustment of the same parameters in each Zone. It limits the Zone display(s) to individual parameters (or rows of parameters) and steps through them from top to bottom. This allows you to adjust the appropriate parameters of all Zones in a Group more easily as working on a reduced set of parameters is simpler. It should be noted that if all Zone parameters are visible before the Next Zone Parameter Key Command is invoked, use of the function will switch to a reduced view mode.

View: Next Group Parameter

Operation and functionality is as per the Next Zone Parameter, for Group Parameters.

Move Audiofiles

Moves the audio files of the selected Instrument into a desired folder location. Use of this option will launch a standard operating system file navigation/browse utility. You may browse to an existing folder or enter a new name, as desired. If no folder name is entered, a new folder will be created which matches the Instrument name, and all audiofiles will be moved into this folder.

Copy Audiofiles

Operation is as per the *Move Audiofiles* function, but files are duplicated, rather than moved, to the nominated folder. This facility should be used as part of your working methods when creating Logic songs, as discussed in "Saving of Project-related EXS24 Instruments" on page 528.

Move audiofiles of all instruments...

Moves the audio files of all Sampler Instruments in the Sampler Instruments folder to the target directory of your choice. In the target location, folders for the audio files associated with these Sampler Instruments are created.

Backup/Copy audiofiles of all instruments...

Copies the audio files of all Sampler Instruments in the Sampler Instruments folder to the target directory of your choice. In the target location, folders for the audio files associated with these Sampler Instruments are created. In addition, the Sampler Instrument files themselves are also copied.

Backup audiofiles of all USED and ACTIVE instruments of current song...

Copies the audio files of all (active) Sampler Instruments used by the current song to the target directory of your choice. Folders for the audio files associated with these Sampler Instruments are created in the target location. All used Sampler Instrument files are also copied.

Move audiofiles of all USED and ACTIVE instruments of current song...

Moves the audio files of all (active) Sampler Instruments used by the current song to the target directory of your choice. Folders for the audio files associated with these Sampler Instruments are created in the target location.

A Brief History of Sampling

The idea of an instrument that could change its sound at any time, and that could imitate any other instrument, dates back centuries. By the 15th century, organ builders had managed to simulate violins, flutes, trumpets, and even human-like sounds with their instruments. Some years later, organs were perfected that could imitate birdsong.

Following the inception of film sound, several instruments were built that used film for the storage and playback of sound. Motion picture sound was based on the concept of recording sound onto the film itself as a separate track. Changes in brightness were read via an opto-electrical mechanism, and sound was replayed. This meant that sound was transferred to light and graphics in the widest sense. Creative musicians of the time began to scratch these films manually, to draw waves on them, and to film gearwheels and other things in order to produce interesting sounds from these images.

The immediate next of kin to today's samplers, however, was the Mellotron. This was a very bulky keyboard instrument that used a separate tape recording of an acoustic instrument for each and every key. Pressing a key started the playback of the corresponding tape; after releasing the key, the tape was drawn back by a spring. Due to the very complicated electro-mechanical mechanism used by the Mellotron, it was a very heavy and frequently unreliable keyboard instrument.

Compared to this, the first digital samplers at the beginning of the eighties seemed ultra-modern, but from today's point of view they did not offer much for their five or six digit price tag: a few seconds of sampling time, and sound quality that is surpassed by today's speaking toys. Nevertheless, early samplers like the Fairlight CMI and E-mu's Emulator are considered legendary. They had a great impact on music and on the development of electronic musical instruments in the following years.

Nowadays, hardware samplers all sound good and are comparatively affordable. However, this is not the end of development for samplers. With computers getting faster and faster, it is now possible to build a fully-fledged sampler entirely in software, making hardware samplers unnecessary. Your EXS24 is proof of this...

MIDI Controller List

Common	Mono Mode	71
	Voices	72
	Start Fixed	73
	Start via Vel	74
	Time via Key	11
	Attack Curve	112
Pitch	• Pitch Bend (up)	9
	• Pitch Bend (down)	70
	• Transpose	5
	Coarse Tune	76
	• Fine Tune	77
	Glide	78
	Pitcher	79
	Pitcher via Vel	80
	Modulation LFO	81
	Mod. Depth Fixed	82
	Mod. Depth Wheel	83
Filter	Filter (on/off)	84
	Filter Type	85
	Filter LFO	86
	Filter LFO Fixed	87
	Filter LFO Wheel	88
	Filter Cutoff	89
	Filter Resonance	90
	Filter Drive	91
	Filter via Key	92
	• Filter via Vel	93
	Filter ADSR Fixed	94
	Filter ADSR via Vel	95
	Filter Attack	106
	Filter Att. via Vel	107
	Filter Decay	108
	Filter Sustain	109
	Filter Release	110

Volume	• Output Volume	67
	• Key Scale +/-	68
	Level Fixed	96
	Level via Vel	97
	Tremolo/Pan LFO	98
	Pan Modulation	99
	Tremolo	100
	Amp Attack	113
	Amp Att. via Vel	114
	Amp Decay	115
	Amp Sustain	116
	Amp Release	117
LFOs	LFO 1 Dec./Delay	101

GarageBand Instruments are automatically installed with Logic. You can insert them as per other software instruments.

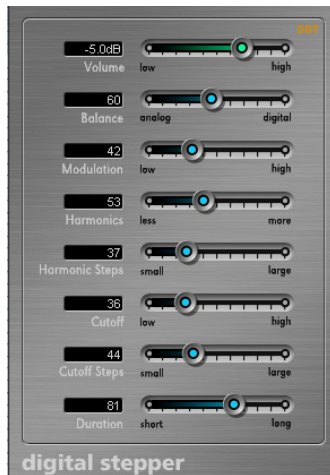
GarageBand Instruments are accessible from the *Stereo > Logic > GarageBand Instruments* sub-menu.

About GarageBand Instruments

GarageBand Instruments are software instrument plug-ins that are used in Apple's *GarageBand* application. Their inclusion makes the importing of GarageBand files into Logic a trouble-free experience.

GarageBand Instruments are actually small variations of equivalent Logic instrument plug-ins. In the case of synthesizer sounds, the ES2 is the "big equivalent" of the GarageBand Instrument. In the case of organ sounds, the EVB3's is the big brother, in the case of electric piano sounds, it's the EVP88 and so on with Clavinet (EVD6) or other sounds (EXS24).

The interface of GarageBand Instruments consists of a simple silver panel that contains a number of parameter sliders and associated value fields. As an example, here is the Digital Stepper instrument:



Many of these parameters are macro parameters, which address specific, useful parameters in the EXS24, ES1 (or other equivalent Logic instrument) instance simultaneously.

This has two main benefits:

- as the GarageBand plug-in are smaller, they load faster than the equivalent software instrument and use less processing power
- limitation to a few, but powerful parameters makes use of the instruments very easy. Play around with the parameters to see how easy it is to get spectacular sounds!

The *Macro* parameter sliders available to each GarageBand Instrument are different. This is because the Logic Instrument's parameters they address may be different or because there's no need to include an organ's *Drawbars* parameter on a GarageBand Piano Instrument, for example—at least not unless you've been getting creative with your Steinway in the garden shed!

The External Instrument plug-in provides a simplified way of handling an external MIDI sound source if you're feeding its audio output directly into your audio interface.

This facility allows you to use one Arrange track for both MIDI recording and audio mixing with effect plug-ins.

The External Instrument plug-in can be inserted in Audio Instrument channels (Mono/Stereo > Logic > External) in place of a software instrument.

The plug-in features a *MIDI Destination* pull-down menu, that allows you to choose one of the MIDI Instrument Objects in your Environment. The *Input* field allows you to select one (or a pair of) audio inputs (connected to the audio output(s) of the external sound source). *Input Volume* determines the incoming signal level.

Obviously, this facility is not as useful if the audio interface you are using with Logic is not equipped with multiple physical inputs.

Additional things to bear in mind:

You might still have the corresponding Input Objects (Input 1, for example) in your mixer. Any effects inserted into this Input channel are placed before the effects inserted in the External Instrument channel (in the signal chain).

If using multitimbral MIDI sound sources, please be aware that each External Instrument requires a separate audio output.

Freezing an External Instrument track cannot happen faster than realtime, as per any Bounce operations where MIDI hardware is involved.

AAF Abbreviation for Advanced Authoring Format. This file format, typically used for data exchange with Digidesign ProTools software, can be imported and exported by Logic. It allows multiple audio tracks to be imported, with reference to tracks and Region position, volume automation included.

AD converter or ADC Short for analog/digital converter; a device that converts an analog signal to a digital signal.

aftertouch MIDI data-type generated by pressure on keys after they have been struck. There are two types: Channel aftertouch, the value of which is measured by a full length keyboard sensor. It affects all played notes. Polyphonic aftertouch (rare) is individually measured and transmitted for each key. Aftertouch is also known as pressure.

AIFF Abbreviation for Audio Interchange File Format. A cross-platform file format supported by a large number of digital video and audio editing applications. AIFF audio can use a variety of bit depths, but the two most commonly used are 16 bit and 24 bit.

AKAI Common sample format that the EXS24 is compatible with.

alias A pointer to a MIDI Region in the Arrange window. An alias does not contain any data. It simply points to the data of the original MIDI Region. You can create an alias by Shift-Option-dragging the original MIDI Region to a new location. An alias can not be edited directly. Any change to the original Region will be reflected in the alias.

aliasing A digital artefact that occurs when the sample material contains frequencies higher than one-half of the sample rate.

allpass filter A filter that allows all frequencies to pass, providing only phase shift or phase delay without appreciably changing the amplitude characteristic.

amplifier Device which controls the level of a signal.

amplitude This term is used to describe the amount of a signal. If you have an audio signal, amplitude refers to the sound's volume, measured in decibels (dB).

analog signal A description of data that consists of a constantly varying voltage level, that represents audio information. Analog signals must be digitized, or captured, for use in Logic. Compare with digital.

Arrange window The heart of Logic. The primary working window of the program where Audio and MIDI Regions are edited and moved to create a song arrangement.

attack Start phase of a sonic event. Also part of an envelope (see envelope).

attenuate To lower an audio signal's level.

Audio Configuration window Logic window that provides an overview of all audio routing. Allows the copying of the entire audio configuration between Logic songs, and assists in renaming tasks. You can open the Audio Configuration window by choosing Audio > Audio Configuration.

audio file Any digital recording of sound, stored on your hard drive. You can store audio files in the AIFF, WAV and Sound Designer II formats. All recorded and bounced WAV files are in Broadcast Wave format.

Audio Instrument Logic supports the use of software based instruments. Software instrument plug-ins are inserted into Audio Instrument Objects. Software instrument recording takes place on Audio Instrument tracks in the Arrange window. Playback of these tracks is routed via the Audio Instrument Object.

audio interface Device needed to get sound into and out of your computer. An audio interface converts digital audio from your computer to analog waves that speakers can broadcast, or, in the other direction, an audio interface converts analog waves into digital audio your computer can work with.

Audio Mixer An Environment layer that shows all Audio Objects of a song. The Audio Mixer is used for mixing multiple audio channels in real time. The Audio Mixer is also known as Environment Mixer.

Audio Object Audio Objects are found in Logic's Environment. They are the "building blocks" of the Audio Mixer. When expanded, Audio Objects look just like channel strips. Audio Objects tell Logic where to send audio signals. The following Audio Object types are available: Audio Track Object, Input Object, Audio Instrument Object, Bus Object, Output Object, Master Object, Aux Object.

Audio Region Chosen area of an audio file which is registered in the Audio window for use in the song and, can be placed on audio tracks in the Arrange window, just like a MIDI Region can be placed on MIDI tracks. Audio Regions are aliases (or pointers) to portions of audio files. They can be as short as a single sample, or as long as the audio file itself. You can use all of Logic's tools to edit Audio Regions. Editing is non destructive on the original audio file, as the Region is only an alias of the audio file. See also Region and MIDI Region.

audio track A track in Logic's Arrange window that is used for playback, recording and editing of Audio Regions.

Audio Track Object Audio Object in the Environment's Audio layer. Used to playback audio tracks in Logic's Arrange window. All data on the audio track is routed to the Audio Object, that was assigned in the Arrange window's Track List menu.

Audio Units (AU) Audio Units is the standard format for real-time plug-ins running on Mac OS X. It can be used for audio effects and software instruments. The Audio Unit format is part of the Mac OS X operating system. Once installed, Audio Unit plug-ins can be accessed by all programs simultaneously. Logic supports all Audio Unit format plug-ins.

Audio window Logic window used for a number of audio file handling and conversion tasks.

Autoload Song Song with your favorite settings and preferences. It loads automatically when you launch Logic, and serves as a starting point for your songs and projects.

automation Automation is the ability to record, edit, and play back the movements of all knobs, controls and buttons, including volume faders and pan, EQ, and Aux send controls plus almost all effect and instrument plug-in parameters.

Aux Object Audio Object in the Environment's Audio layer. Aux Objects are similar to the Bus Objects, but are more flexible. Unlike Bus Objects Aux Objects also have Sends of their own, allowing you to form complex signal paths

bandpass filter This filter only allows the frequency band centered around the cutoff frequency to pass, while frequencies that lie further away (the lows and highs) are filtered out. A sound that contains lots of mid range frequencies is the result. Also see filter.

band rejection filter This filter cuts the frequency band centered around the cutoff frequency, while allowing the frequencies that lie further away to pass. The mid range frequencies will become softer and the lows and highs remain unchanged.

bar In musical notation, a bar is a measure that contains a specified number of beats, and establishes the rhythmic structure of a musical piece.

Bar Ruler Ruler found at the top of the Arrange, Matrix, Hyper and Score windows. It displays musical time units including bars, measures, beats and beat divisions. It is used to set and display the song position, the cycle and autodrop locators, as well as markers.

beat A musical time interval: "the beat is the regular rhythmic pulse in a composition that people tap their feet to." Usually a quarter note.

Beat Mapping track Component of the Global tracks that helps to make a rhythmically meaningful display of recordings that do not correspond to a strict tempo throughout. It does this by redefining the bar positions of existing musical events, without changing their absolute time position, thereby preserving the audible result with its original timing.

beats per minute See bpm.

bit depth The number of bits a digital recording or digital device uses. The number of bits in each sample determines the theoretical maximum dynamic range of the audio data, regardless of sample rate. Also known as bit resolution, word length or bit rate.

bit rate See bit depth

bit resolution See bit depth

blue noise Highpass-filtered white noise, sounds like tape hiss.

boosting The act of raising an audio level.

bounce To process recorded or streamed MIDI and/or Audio Regions with any applied effects, such as delay or compression, combining them into one audio file. In Logic, you can choose between Realtime and Offline bouncing. Offline bouncing is faster, but doesn't allow you to apply live automation or record real time audio input.

Bounce button You can bounce the output of any Output Object to an audio file by clicking the Object's Bounce button. See also bounce.

bpm Abbreviation for beats per minute, a measure of the tempo of musical piece. As an example: 120 bpm means that in one minute, there will be 120 musical beats (quarter notes).

bus The term bus is used to describe a send/return routing scheme for audio channels. In Logic, effects can be sent to/from Bus Objects for processing or submixing tasks.

Bus Object Audio Object in the Environment's Audio layer. Usually used to route the signal of an individual send bus to Output Objects. See also bus.

bypass To deactivate a plug-in. Bypassed plug-ins do not drain system resources. In Logic you can bypass a plug-in by either clicking its Bypass button in the plug-in window or by Option-clicking on the appropriate plug-in slot.

cable In Logic the term cable is used to describe the virtual cables that represents the MIDI connection between two Environment Objects.

carrier In FM synthesis, the carrier is the equivalent of an analog synthesizer oscillator that is producing a sine wave. The carrier frequency is modulated by the modulator.

Catch button The button in the Transport bar featuring the running man icon. Activate this button (blue) to turn on automatic horizontal scrolling during playback. This ensures that the current playback position is always visible.

Catch function A window function that makes the currently displayed song section reflect the current song position. Also see Catch button.

CD Audio Short for Compact Disc—Audio; current standard for stereo music CDs: 44.1 kHz sampling rate and 16 bit depth.

cent A tuning subdivision of a semitone. There are one hundred cents in a semitone. Many of Logic's software instruments contain a Fine parameter that allows sounds to be tuned in cent steps.

channel strip A channel strip is a virtual representation of a channel strip on a mixing console. Each channel strip contains a number of similar controls, such as a Mute button, Volume fader, Pan/Balance knob, Output selector and Bus and/or Insert slots.

Channel Strip setting Logic allows the routing of a channel strip, including all inserted effects or instruments (plus their settings) to be saved and recalled. This simplifies the task of recreating complex serial effect routings between channels or songs.

checkbox A small box. You click a checkbox to select or deselect an option.

chorus effect Effect achieved by layering two identical sounds with a delay and slightly modulating the delay time of one or both of the sounds. This makes the audio signal routed through the effect sound thicker and richer, giving the illusion of multiple voices.

click Metronome, or metronome sound.

Clipboard The Clipboard is an invisible area of memory, into which you cut or copy selected objects, using the Edit menu. From there, you can paste these objects to different positions. Logic's Clipboard spans all songs, allowing it to be used to exchange objects between songs.

clipping (in digital recording) Feeding too much signal through a channel strip, thereby exceeding the limit of what can be accurately reproduced results in a distorted sound known as clipping. Logic's Audio Objects feature a clip detector, which indicates signal level peaks above 0 dB.

comb filter effect A short delay of feedback that emphasizes specific harmonics in a signal is generally termed a comb filter. The name is derived from the appearance of a frequency spectrum graphic, which resembles the teeth of a comb.

compressor An effect that restricts the dynamic range of an audio signal.

controller MIDI data type. As examples; sliders, pedals or standard parameters like volume and panning. The type of command is encoded in the first data byte, the value in the second data byte.

Controls view All Logic plug-ins (and Audio Units) offer a non-graphical alternative to the Editor views of effect and instrument parameters. The Controls view is accessed via the Controls pull-down menu at the top of each plug-in window. This view is provided to allow access to additional parameters and to use less onscreen space.

Core Audio Standardized audio driver system for all Macintosh computers running version 10.2 or higher. Core Audio is an integral part of Mac OS X, allowing access to all audio interfaces that are Core Audio compatible. Logic is compatible with any audio hardware that offers Core Audio drivers.

Core MIDI Standardized MIDI driver system for all Macintosh computers running Mac OS X version 10.2 or higher. Core MIDI is an integral part of Mac OS X, allowing the connection of all MIDI devices that are Core MIDI compatible.

cutoff frequency Frequency at which the audio signal passing through a low or highpass filter is attenuated by 3 dB.

Cycle function A function in Logic which constantly repeats the area between the Locator positions. To turn on Cycle mode, click the Cycle button in the Transport window. The cycle function is useful for composing a part of a song or editing events, as examples. The Cycle area is shown as a green stripe in the top part of the Bar Ruler.

DA converter or DAC Short for digital/analog converter; a device that changes an analog signal into a digital signal.

DAW Acronym for Digital Audio Workstation. A computer used for recording, mixing and producing audio files.

dB Abbreviation for decibels, a unit of measurement that describes the relationships of voltage levels, intensity or power, particularly in audio systems.

decay An envelope parameter that determines the time it takes for a signal to fall from the maximum attack level to the sustain level. See envelope.

Deesser A signal processor that removes hissing or sibilance in audio signals.

default The preset parameter value.

delay In the Environment, an Object that can create a series of repeats. In the Arrange window, a Region parameter which can delay or advance a selected Region by a given number of milliseconds. Delay is also an effect process that delays the incoming audio signal, resulting in subtle chorusing effects through to endless repeats of the signal.

destructive Destructive audio processing means that the actual data of an audio file is changed, as opposed to just editing peripheral or playback parameters.

dialog A window containing a query or message. It must be cancelled or replied to before it will disappear and allow you to continue.

digital A description of data that is stored or transmitted as a sequence of ones and zeros. Most commonly, refers to binary data represented using electronic or electromagnetic signals. All files used in Logic are digital. Also see analog for comparison.

disclosure triangle A small triangle you click to show or hide details in the user interface.

distortion The effect produced when the limit of what can be accurately reproduced in a digital signal is surpassed, resulting in a sharp, crackling sound.

distributed audio processing See Logic Node application.

drag & drop Grabbing objects with the mouse, moving them, and releasing the mouse button.

driver Drivers are software programs that enable various pieces of hardware and software to be recognized by other programs in a computer, and also to have the appropriate data routed to them in a format they can understand. In Logic, you can use the Preferences > Audio > Drivers panel to select and configure your audio hardware drivers. If you do not have the proper driver installed, your computer may not recognize or work properly with a given piece of hardware.

DSP (digital signal processing) In Logic, the mathematical processing of digital information to modify a signal. An example is the Insert slot of channel strips, which assigns DSP effects such as dynamic compression and delay to a channel's signal.

DTDM Mixer Logic Pro supports a number of Digidesign hardware devices via Direct TDM. The DTDM mixer is created in Logic's Environment window, and allows the use of Logic "native" effects and instruments with suitable Digidesign hardware.

dynamics Refers to changes in volume or other aspects of a piece of music over time.

dynamic range The dynamic range of a sound system is the difference in level between the highest signal peak that can be reproduced by the system (or device in the system) and the amplitude of the highest spectral component of the noise floor. The dynamic range is the difference between the loudest and softest signals that the system can reproduce. It is measured in decibels (dB). See decibels.

editor Window for editing MIDI or audio data. Logic offers the Hyper, Matrix and Score editors for MIDI event data, and the Sample Editor for audio data.

Editor view Almost all Logic plug-ins (and Audio Units) offer a graphical view of effect and instrument parameters. The Editor view is used by default, but can be accessed via the Editor pull-down menu at the top of each plug-in window, should the Controls view be visible.

effect A type of software algorithm that lets you alter the sound of a track in a variety of ways. Logic includes a set of EQ, dynamics, time-based, modulation and distortion effects in Logic's native and Audio Unit plug-in formats.

envelope The envelope is the variation that a sound exhibits over time, an envelope basically determines how a sound starts, continues and disappears. Synthesizer envelopes usually consist of Attack, Decay, Sustain and Release phases.

Environment The Environment is Logic's brain: it graphically reflects the relationships between hardware devices outside your computer and virtual devices within your computer. Beyond basic input and output handling, the Environment can be used to process MIDI data in real-time, and can even be used to create processing "machines", such as virtual rhythm generators and step sequencers or complex synthesizer editors.

Environment layer A place in the Environment, used to organize Objects and making usage easier. Objects of the same type (Audio Objects, for example) are generally placed on the same layer.

Environment Mixer See Audio Mixer

EQ Shortened form of equalizer. Equalizers are used to boost or cut frequencies in an audio signal. There are several types available in Logic.

equalization See EQ

Eraser A tool used for deleting items. Click a selected item to delete it. All other currently selected items are also deleted.

ESB TDM The ESB TDM connects your TDM hardware with Logic's audio engine. This allows your computer's CPU to perform processes in Logic's native mixer, including audio track playback, the use of software-based instruments and effect plug-ins.

event Individual MIDI command, such as a note on command. Continuous controller movements (modulation wheel, for example) produce a quick succession of individual events with absolute values.

export To create a version of a file, such as a Logic song, in a different format that can be distributed and used by other applications.

filter effect Filters are effects you can apply to Audio or MIDI Regions (when streamed or recorded as audio). They are designed to reduce a signal's energy at a specific frequency. A true filter always acts as a subtractive device, and doesn't add anything to the signal. The names of the individual filters illustrate their function. As an example: A Low Pass filter allows frequencies that are lower than the cutoff frequency to pass.

Filter button Buttons in the Event List/Track Mixer, that allow you to hide/show specific event types/channel strip types.

filter slope The filter slope is the steepness, or severity, of filter attenuation. As examples, a filter slope of 6 dB per octave would sound much softer than a filter slope of 12 dB per octave.

flanger The flanger effect is similar to the chorus effect, where a slightly delayed signal (which is shorter than that of the Chorus) is fed back into the delay line input. Flanging makes a sound thicker, and slightly "out of phase".

float window Window with special status which always "floats" on the surface above all other windows, but can only be operated with the mouse. Any Logic window can be opened as a float window by holding down Option while opening it.

frame Unit of time. A second in the SMPTE standard is divided into frames that correspond to a single still image in a file or video.

Freeze function The Freeze function performs individual offline bounce processes for each "frozen" track, saving almost 100% of the CPU power used for software instruments and effect plug-ins. All plug-ins of a track (including software instrument plug-ins, if applicable, along with all related automation data) are rendered into a "Freeze file." You can use the Freeze function on individual Audio or Audio Instrument tracks.

frequency The number of times a sound signal vibrates each second, measured in cycles per second, or Hertz (Hz).

grab (an object) Positioning the mouse cursor over an object, then pressing and holding the mouse button down.

help tag A small text window that appears when the mouse cursor is placed over an interface element that indicates the name or value. When editing operations such as moving or cutting a Region are performed, a larger help tag will display the current position of the Region or function—in realtime.

Hermodé Tuning A microtonal tuning system that can be used on all Logic and Audio Unit software instruments. Hermodé Tuning can make your software instruments sound harmonically richer by fine tuning thirds, fifths and sevenths to specific intervals—in cents.

hierarchical menu Structured menus where choosing an individual entry opens a submenu.

high cut filter A high cut filter is essentially a lowpass filter that offers no slope or resonance controls.

highpass filter A highpass filter allows frequencies above the cutoff frequency to pass. A highpass filter that offers no slope or resonance controls is generally known as a low cut filter.

icon Small graphic symbol. In Logic, an icon may be assigned to each track.

importing The process of bringing files of various types into a Logic project or song. Imported files can be created in another application, captured from another device, or brought in from another Logic project.

Input Object Audio Object in the Environment's Audio Layer. The Input Object represents the physical inputs of your audio interface and helps managing audio from your audio interface into Logic.

Insert slot A point on Logic's Mixers where you can patch in (insert) an effect plug-in. All audio channel types in Logic's Track and Audio Mixers (except the Master Object) offer effect insert slots.

interface 1) A hardware component such as a MIDI or audio device that allows Logic to "interface" (connect) with the outside world. You need an audio or MIDI interface to get sound/MIDI into and out of your computer. Also see audio interface. 2) A term that is used to describe Logic's graphical elements that can be interacted with. An example would be the Arrange window, where graphical interface elements such as Regions are interacted with to create an arrangement, within the overall Arrange interface.

key The scale used in a piece of music, centered around a specific pitch. The specified pitch is called the root of the key.

key command Function which can be executed by pressing a specific key (or key combination) on your computer keyboard or MIDI controller.

latency You may notice a delay between playing your keyboard and hearing the sound. This is a form of latency. A variety of factors contribute to latency including audio interface, audio and MIDI drivers. One factor under your control, however, is the I/O buffer size, which is set in the Audio > Audio Hardware & Drivers preferences.

legato Method of musical performance that smoothly connects one note to the next.

level meter A meter that lets you monitor audio output levels from your computer. You use the level meters in Logic when recording, arranging and editing audio files.

Link mode Link mode is activated by clicking the Link button. It determines the relationships between windows. An editing window in Link mode shows the same contents as the top window.

Link button Button featuring the chain link icon in the top left corner of most Logic windows. It controls the linking between different windows.

local menu Menu in a window that only contains functions that are relevant to that particular window.

Local Off mode Operating mode on a MIDI keyboard where the keyboard does not directly play its own integrated sound generator. This is useful when using it as a master keyboard in a MIDI setup with Logic.

Locators Lower two sets of numbers, displayed to the right of the Transport buttons in the Transport window. The number on top is the left Locator; the number on bottom the right Locator. The Left and Right Locators control the time-range which Logic's playback will cycle or skip during playback. The Locators also can be used to define the editing area for certain functions.

Logic Setup Assistant A utility that guides you through the process of setting up your Macintosh, audio and MIDI hardware to work with Logic. The Logic Setup Assistant can be run at any time by choosing the Preferences > Start Logic Setup Assistant menu item.

loop An audio clip that contains recurring rhythmic musical elements or elements suitable for repetition. Logic also supports Apple Loops.

Loop function Loop is a Region parameter in Logic that creates "loop repetitions" for an Audio or MIDI Region. These repetitions will repeat until the song end point, or until another Region or folder (whichever comes first) is encountered on the same track in the Arrange window.

LFO Abbreviation for Low Frequency Oscillator. An oscillator that delivers modulation signals below the audio frequency range—in the bandwidth that falls between 0.1 and 20 Hz, and sometimes as high as 50 Hz or 400 Hz.

lowcut filter A low cut filter is essentially a highpass filter that offers no slope or resonance controls.

lowpass filter The lowpass filter defines the maximum frequency that can pass through without being affected, thus controlling the brightness of the sound. Every signal above this frequency will be cut. The higher the cutoff frequency, the higher the frequencies that can pass through. A lowpass filter that offers no slope or resonance controls is a high cut filter.

marker Markers serve three purposes in Logic: They mark time-positions in the Arrange window. They hold text notes and they delimit song settings. Markers can be placed in the Marker track, or they can be placed in the Bar Ruler. Markers are generally used for indicating and navigating to different song sections.

main menu bar The bar at the top of the computer screen, offering global functions such as opening, saving, exporting or importing songs. It does not offer access to local functions.

merge Mix, or combine, two or more MIDI events or Regions into a single Region.

metronome A part of Logic that produces a sound that taps out the beat. It can set by click-holding the Metronome button in the Transport bar.

MIDI Abbreviation for Musical Instrument Digital Interface. Standardized, asynchronous, serial and event-oriented interface for electronic musical instruments. MIDI is an industry standard that allows devices such as synthesizers and computers to communicate with each other. It controls a musical note's pitch, length, and volume, among other characteristics.

MIDI channel A MIDI channel is a "tube" for MIDI data, which flows through MIDI ports in channels. Up to 16 separate MIDI channels can pass through a port simultaneously.

MIDI message A message transmitted via MIDI consisting of one status byte and none, one, two or many data bytes (with system exclusive commands). See event.

MIDI Multi mode Multi-timbral operating mode on a MIDI sound module where different sounds can be controlled polyphonically on different MIDI channels. A Multi mode sound module behaves like several polyphonic sound modules. General MIDI describes a 16-part multi mode (the ability to control 16 different parts individually). Most modern sound generators support multi mode. In Logic, multi mode sound modules are addressed via Multi Instrument Objects.

MIDI Region Data container for MIDI events which is shown in the Arrange window as a named horizontal beam. In earlier Logic versions MIDI Regions were called sequences.

mixing The process of shaping the overall sound of a song by adjusting the volume levels, pan positions, adding EQ and other effects, and using automation to dynamically alter aspects of the song.

modifier key Computer keyboard keys used in conjunction with alphabetical keys to change functionality. Modifier keys include; Control, Shift, Option and Command.

modulation Generally, a slight, continuously varying change. Logic's effects and synthesizers contain a number of modulators.

modulation amount The strength, or intensity, of modulation.

modulation matrix The EXS 24 and other Logic instruments contain a grid that allows you to modulate a number of target parameters with a number of modulators. This grid is referred to as the modulation matrix.

modulation path A modulation path determines which target parameter will be modulated by a specific modulator (modulation source).

modulation wheel A MIDI controller found on most MIDI keyboards.

mono Short for monophonic sound reproduction. The process of mixing audio channels into a single track, using equal amounts of the left and right audio channel signals. Compare with stereo.

MP3 Abbreviation for MPEG-2 Audio Layer 3. A compressed audio file format, frequently used to distribute audio files over the Internet.

MS stereo recordings Short for middle-side stereo. Two microphones are stacked on a stand or suspended from the ceiling so that they are positioned as closely together as possible. One microphone delivers the middle signal, the other the side signal. If you want to use MS stereo recordings in Logic, you have to decode them—this can for example be done with the Direction Mixer plug-in. Also see XY stereo recording.

MTC See MIDI Time Code.

Multi Instrument Object An Object in Logic's Environment that represents a multi timbral hardware or software device that reacts to MIDI. The Multi Instrument Object is essentially 16 Instrument Objects rolled into a single package. Each of these, called sub-channels (or part-instrument), has a fixed MIDI channel and shares the same port. All other parameters can be set individually. The purpose of Multi Instrument Object is to address multi-channel MIDI devices, which receive MIDI data and play different sounds on separate MIDI channels.

multitimbral This term describes an instrument or other device that can play different sounds at the same time, using several MIDI channels at the same time.

Multi Trigger mode This term is associated with synthesizers such as the ES 1. In this mode, a synthesizer envelope usually is retriggered by every note played.

mute Switch off an Audio Object or track's audio output. You can mute a track by clicking the Track Mute button in the Track List. The output of an Audio Object is disabled by clicking the Mute button at the bottom of the channel strip.

nodes Positions in Hyper Draw and automation tracks that mark the positions where data manipulation begins or ends. Occasionally referred to as points.

normalize This function applies the current Parameter box settings to the selected MIDI events (by altering the actual events themselves), and clears the Parameter settings. When it comes to audio, a different “Normalize” function raises the volume of a recorded audio file to the maximum digital level without altering the dynamic content.

notch filter This filter type cuts the frequency band directly surrounding the cutoff frequency and allows all other frequencies to pass.

note number Pitch of a MIDI note, controlled by the first data byte of a MIDI note event.

Object If capitalized, the term Object is used to refer to the graphical representation of all elements in Logic’s Environment. These elements can be used to create and process MIDI data in real-time, and can even be used to create processing “machines”, such as virtual rhythm generators or step sequencers. Examples for Objects are Instruments, Multi Instruments, Faders, Arpeggiators and others. In the Environment’s Audio layer you also find Objects used to process audio data. These Objects are correspondingly named Audio Objects. Also see Audio Objects.

Object Parameter box The Object Parameter box displays the properties of any selected Environment Object. In the Arrange window this Parameter box is located below the Toolbox and displays the properties of the selected Track’s Object.

OMF Abbreviation for Open Media Framework, also known as OMFI—Open Media Framework Interchange. This file format, typically used for data exchange with Digidesign ProTools software, can be imported and exported by Logic. The OMF file format only supports the exchange of audio data (audio media and the usage of this audio media in a song). MIDI and automation data will simply be ignored when using Logic’s export function. **option** 1) Alternative function, often in the form of a checkbox, sometimes also available as a menu entry. 2) Modifier key, in Windows terminology this key is also known as Alt key.

oscillator A synthesizer oscillator generates an alternating current, using a selection of waveforms which contain different amounts of harmonics.

Output Object Audio Object in Logic’s Environment controlling the output level and pan/balance for each output on your audio interface. They are assigned to a specific hardware output in their Object Parameter box.

pan, pan position The placement of mono audio signals in the stereo field, by setting different levels on both sides.

Parameter box Field on the left side of Logic’s windows used to adjust the parameters of the selected Regions or Objects.

peak 1) The highest level in an audio signal 2) portions of a digital audio signal that exceed 0 dB, resulting in clipping. You can use Logic's level meter facilities to locate peaks and remove or avoid clipping. The Search Peak command in the Sample Editor's Functions menu searches for the sample bit with the greatest amplitude value in the currently selected Audio Region.

pink noise A harmonic noise type that contains more energy in the lower frequency range.

pitch The perceived highness or lowness of a musical sound. Corresponds to the frequency of the sound wave.

pitch bend message MIDI message transmitted by a keyboard's pitch bend wheel.

playback Playing an Audio or MIDI Region or an entire arrangement, allowing you to hear it.

plug-in Software application that enhances the functionality of the main program (in this case, Logic). Logic's plug-ins are typically software instruments or effects.

Plug-in window A window that launches when a plug-in is inserted, or the Insert/Instrument slot is double-clicked. Allows you to interact with the plug-in parameters.

post fader Sends in analog mixers are positioned either before (pre) or after (post) the fader. Post fader means positioned after the fader in the signal flow, with the level of a signal going to the Send changing along with the fader movements.

pre fader Sends in analog mixers are positioned either before (pre) or after (post) the fader. Pre fader means positioned before the fader in the signal flow, so the level of a signal routed pre fader to a Send remains constant, regardless of any fader movements.

Preferences window A window that is accessed via the Logic > Preferences menu. All Logic preferences can be set in this window.

preset Set of plug-in parameter values that can be loaded, saved, copied or pasted via the Settings menu in the Plug-in window header. See setting and Settings menu.

pressure See aftertouch

project In Logic, the top-level folder that holds all media associated with a song, including audio files, Sampler Instruments and samples, Video and Settings of various kinds.

Project Manager A window that allows you to manage all media and file types that Logic can read/use. You can access the Project Manager by choosing Windows > Project Manager.

PWM Pulse Width Modulation. Synthesizers often feature this facility, where a square waveform is deformed by adjusting its pulse width. A square waveform usually sounds hollow, and woody, whereas a pulse width modulated square wave sounds more reedy and nasal.

Q factor A term generally associated with equalizers. The Q factor is the “quality” factor of the equalization, and is used to select a narrower or broader frequency range within the overall sonic spectrum of the incoming signal.

quantization Time-correction of note positions by moving them to the nearest point on a selectable grid. When quantization is applied to any selected event or Region, Logic will move all note events non-destructively to align perfectly with the nearest grid position.

RAM Abbreviation for random-access memory. A computer’s memory capacity, measured in megabytes (MB), which determines the amount of data the computer can process and temporarily store at any given moment.

real-time effects Effects that can be applied to Regions in real time, without requiring rendering before playback. Real-time effects can be played back with any Macintosh computer qualified to run Logic.

ReCycle ReCycle is the name of an application from software manufacturer Propellerheads, which mainly serves as an editing and production tool for loops (repeatedly looped audio samples). ReCycle uses specific file formats (.REX) which can be imported by Logic.

Region Regions can be found in the tracks of the Arrange window: They are rectangular beams that act as containers for audio or MIDI data. There are three different types of Regions: Audio Regions, MIDI Regions and Folder Regions. Also see: Audio Region, MIDI Region and Folder.

Region Parameter box Box in the upper left corner of the Arrange window, used to non-destructively set the playback parameters for individual Regions, including; quantization, transposition, velocity, compression and delay. These parameters do not alter the stored data. Rather, they affect how the events are played back.

resonance A term generally associated with filters, particularly those of synthesizers. Resonance emphasizes the frequency range surrounding the cutoff frequency. See cutoff frequency.

reverb Reverb(eration) is the sound of a space. More specifically, the reflections of soundwaves within a space. As an example, a handclap in a cathedral will reverberate for a long time as sound waves bounce off the stone surfaces within a very large space. A handclap in a broom closet will hardly reverberate at all. This is because the time it takes for the soundwaves to reach the walls and bounce back to your ears is very short, so the “reverb” effect will probably not even be heard.

RMS Root Mean Square. A measurement of the effective audio signal average. Used in Logic’s analysis tools.

root note The central note of a musical scale or key, which all other notes are related to.

routing Generally refers to the way audio is sent through processing units. Also often used to describe specific input and output assignments.

sample A digital recording of a sound at a particular instant in time.

Sample Editor Logic’s Sample Editor allows stereo or mono audio files to be destructively cut, reversed, shortened, changed in gain and processed in a number of other ways. It allows editing of individual samples within an audio file consisting of thousands or millions of samples. The Sample Editor also provides access to a number of special sample processing tools, collectively known as the Digital Factory.

sampler Device used for sampling. In Logic, this generally refers to the EXS24 software-based sampler.

sample rate When an analog audio signal is converted to a digital signal, this term refers to the number of times per second the audio file is sampled. Logic can record and edit audio at sample rates ranging from 44.1 kHz (44,100 times per second) up to 192 kHz (192,000 times per second).

sampling The process of converting analog audio into digital information. The sample rate of an audio stream specifies the number of samples that are captured per second (see sample rate). Higher sample rates yield higher quality audio.

saturation A term most commonly associated with a slight tape distortion or the characteristics of tube amplifiers. It basically describes a very high gain level that causes a slight distortion of the incoming signal, resulting in a warm, rounded sound.

scroll bar and scroll box Gray beam at the edge of a window. A movable box inside the beam is used to select the displayed song section in the window.

self-oscillation Self-oscillation is a typical characteristic of analog filter circuits. It occurs when the filter, at high resonance values, feeds back into itself and begins to oscillate at its natural frequency.

semitone Smallest interval between two pitches in the standard diatonic scale, equal to a half tone. Correspondingly a semitone is also called half step or half tone.

send Abbreviation for auxiliary sends. An output on an audio device used for routing a controlled amount of the signal to another device. Sends are for example often used to send several signals to the same effect, which is rather advisable for computationally-intensive effects such as reverb.

sequencer A sequencer is a computer application that allows you to record both digital audio and MIDI data and blend the sounds together in a software mixing console. There are editing tools that let you control every aspect of the production, down to the finest details. Effect processors and software instruments are increasingly being added to these applications. Modern sequencers such as Logic now can fulfill many, if not all, functions that were only possible in the most expensive and well-equipped recording studios a decade ago.

setting 1) A parameter value. 2) A set of plug-in parameter values that can be loaded, saved, copied or pasted via the Settings menu. A plug-in setting is also known as preset. Also see preset and Settings menu.

Settings menu Accessible via the Arrow button found in the gray header at the top of all plug-in windows. Allows you to save, load, copy and paste settings: the parameter values of effects and software instruments.

shelving EQ EQ type that allows reducing or increasing the frequency range above or below the specified frequency.

side chain A side chain is effectively an alternate input signal—usually routed into an effect—that is used to control an effect parameter. As an example, you could use a side chained track containing a drum loop to act as the control signal for a Gate inserted on a sustained pad track, creating a rhythmic gating effect of the pad sound.

Single Trigger mode This term is associated with synthesizers such as the ES 1. In this mode, envelopes are not retriggered when tied (legato) notes are played.

Snap menu A pull-down menu found at the top of all linear editing windows in Logic. Selection of items in the menu will cause events or Regions to “snap” to the specified value when moved.

software instrument Logic’s software counterpart to a real sound source or a sampler or synthesizer module.

solo A way to temporarily highlight one or more tracks or Regions or events, allowing them to be heard in isolation.

Solo tool Click-holding on individual Regions or events with the Solo tool temporarily allows them to be heard in isolation. All other objects are muted.

song Main Logic file, containing all MIDI events and parameter settings (including mixer automation data) plus information about the audio files to be played.

Song Settings The Song Settings, accessible from the File menu, are a collection of program settings that are specific to the current song. These are different to the global preferences that affect all Logic songs (see preference).

stereo Short for stereophonic sound reproduction of two different audio channels. Compare with mono.

Sustain pedal A momentary footswitch that is connected to MIDI keyboards. It transmits MIDI controller number 64, which is recorded and played back by Logic.

synthesizer A device (hardware or software) that is used to generate sounds. The word is derived from early attempts with mechanical and electronic machines to emulate (or synthesize) the sounds of musical instruments, voices, birdsong and so on. Logic features several software synthesizers, including; the ES1, ES2, EFM 1, ES E, ES P and ES M.

tempo The playback speed of a piece of music, measured in beats per minute. Logic allows you to create and edit tempo changes in the Tempo track.

time signature Two numerals separated by a diagonal bar that appear at the beginning of a song. Common time signatures are 4/4 and 2/4. The first number denotes the number of notes in a measure, or bar. The second number denotes a unit of time for each beat. In a 2/4 signature, each bar has two beats; each beat is a quarter note long.

timing Measure of the ability to play notes at the right time. Timing can also refer to synchronization between events, Regions and devices.

toggle To switch between two states such as on or off (applies to windows, parameter values and so on).

Touch Tracks An Environment Object that allows MIDI Regions to be assigned to, and triggered by, individual MIDI note events. This enables you to assign a number of musical phrases to different MIDI keyboard keys, and trigger (or record) them in real-time, making the process of arrangement faster and more intuitive.

track A horizontal row in the Arrange window that contains either Audio or MIDI Regions that can be played back over time. Each track has a specified destination that data is routed to. Logic allows hundreds of tracks to be used in a song.

Track List Situated to the left of the Arrange window's working area. Displays the Objects assigned to various tracks as well as the Track buttons.

Track Mixer Adaptive Mixer which automatically configures itself to show every audio and MIDI track, in the order that they appear in the Arrange window or in an open Folder. If you move the controls on the Track Mixer while recording, automation data is stored in the relevant tracks as MIDI controller information.

transient Position in an audio recording where the signal becomes a lot louder—over a short time span (a signal “spike”, in other words). As this is typical for drum recordings, transients can be used to indicate where beats occur in an audio signal.

Transport window Window used to control recording and playback functions. The Transport window offers Record, Pause, Play, Stop and Rewind/Forward buttons plus other functions. You can also configure a fixed Transport window in the Arrange and Matrix windows by selecting View > Transport. This Transport window variation is named Transport field. The term Transport bar refers to both the Transport window and Transport field.

transposition Transposition is changing the pitch of a Audio or MIDI Region or event by a number of semitones.

Undo function Function which reverses the previous editing operation.

velocity Force at which a MIDI note is struck; controlled by the second data byte of a note event.

virtual memory Area of the hard disk used as an extension of RAM memory by the computer. The disadvantage is its very slow access time, in comparison to physical RAM.

WAV, WAVE The primary audio file format used by Windows-compatible computers. In Logic, all recorded and bounced WAV files are in Broadcast Wave format, which include a high-resolution timestamp.

waveform A visual representation of an audio signal.

wet/dry mix Refers to the ratio of a signal that effects have been added to (wet), and the original, unprocessed signal (dry).

white noise Noise type that consists of all frequencies (an infinite number) sounding simultaneously, at the same intensity, in a given frequency band. Its name is analogous to white light, which consists of a mixture of all optical wavelengths (all rainbow colors). Sonically, white noise falls between the sound of the consonant F and breaking waves (surf). Synthesis of wind and seashore noises, or electronic snare drum sounds, requires the use of white noise.

window class Status of the window as a float window or a normal window. Float windows always “float” in the foreground and can not be hidden by normal windows. Also see float window.

word length See bit depth.

XY stereo recordings Two cardioid microphones aligned so that they are directed to the left and right of the sound source. Also see MS stereo recordings.

zero crossing A point in an audio file where the waveform crosses the zero amplitude axis. If you cut an audio file at a zero crossing there will be no click at the cut point.

zoom An action that enlarges (zooms in on) or shrinks (zooms out from) the display in a Logic window. The Magnifying Glass in the Toolbox, and the Zoom controls found in the lower left and upper right corners of windows, are both used for zooming tasks. Also see Zoom control and zoom level.

Zoom control The control that appears at the bottom left and top right of some windows, such as the Arrange. The Zoom control slider allows you to navigate through the entire length of the currently displayed song. The lines on the left and right of the slider can be clicked to zoom in and out by a fixed percentage.

zoom level The amount that a window's contents (tracks, Regions and Objects, for example) are magnified. Zooming in to a high level allows you to make more precise edits. Conversely, you can zoom all the way out to see the entire song and work on very large sections.

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