Logic Express 7

Plug-In Reference

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Contents

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.

Given a fast enough computer, you could conceivably arrange and mix an entire song using several software instruments, such as Logic's ES1, or EXSP24, 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 96 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 and Audio Unit 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.

Basics

1

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

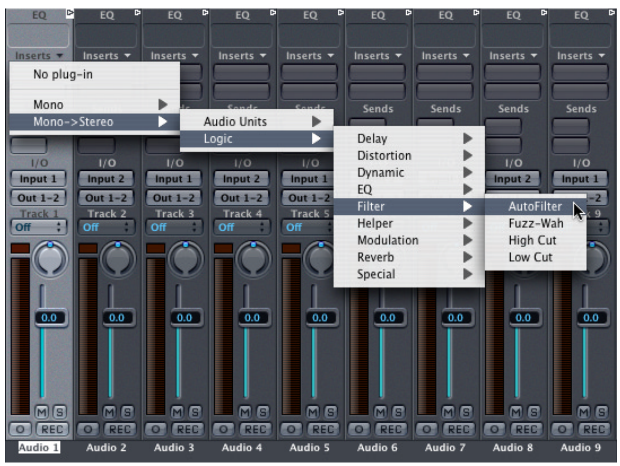
Plug-ins can be either; software instruments, which respond to MIDI note messages, or audio effects, which do not respond to MIDI note messages.

- 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 14. 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 EXSP24 instance was loaded on Audio Instrument channel 1, switching between these tracks would automatically update the Plug-in window to show the ES1/EXSP24, 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 16.

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.

001/ 011 Button

Some Logic plug-ins may have 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-in Settings

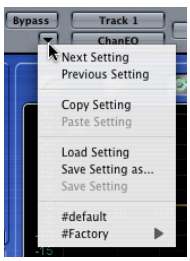
Logic's plug-ins ship with a library of ready-to-play preset sounds, known as Settings. These Settings can be found in the Logic > Plug-In Settings subfolder, following the installation procedure.

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.

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.

Instruments and Effects

2

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 11.). 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 4 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 24 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.

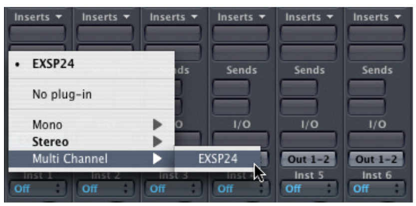
Note: Muting an Audio Instrument track in the Arrange does *not* reduce system load.

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 EXSP24 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 EXSP24 sampler and others) by ±100 cents.

Note: Some instruments do not recognize this remote command.

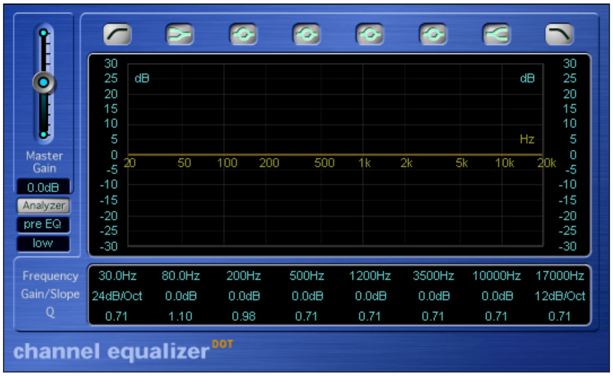
Equalizer

3

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, 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 four frequency bands.

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 4 is a highpass filter.

Note: The Q-parameter of band 1 and band 4 will have no effect when using a slope of 6 dB/Oct.

Bands 2 and3 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 1 to 4. 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.

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.

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.

Dynamic

4

This chapter introduces Logic's Dynamic plug-ins.

This includes the Compressor, Silver Compressor, Noise Gate, Silver Gate, Limiter, and Preset Multipressor 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.

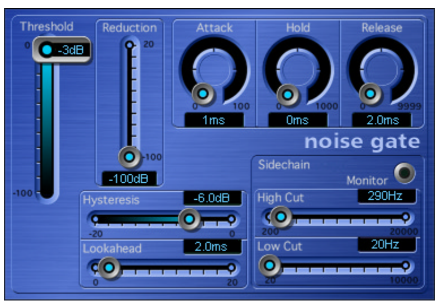
The *Output Clip* parameter limits (clips) the output to 0 dB, via the *OFF/SOFT/HARD* settings. This setting is only available if the 001/011 button is activated.

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.

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.

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).

Preset Multipressor

The Preset Multipressor is an easy-to-use variant of Logic Pro's Multipressor plug-in.

A multi-band compressor splits the incoming signal into different frequency bands before applying compression. These frequency bands are then compressed independently. Following compression, the frequency bands are mixed back together, and sent out of the plug-in.

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

The interface of the Preset Multipressor features 12 radio buttons that allow you to choose between settings optimized for various genres; the names of the presets are pretty much self-explanatory. Make use of the different presets and use your ears to determine which one best fits your needs.

Distortion

5

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

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

Guitar Amp 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 offers a range of Amplifier 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.

Four different amplifier models can be accessed via the *Model* radio buttons at the top.
- 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.

The entries in the pull-down menu at the top are Settings that refer to the simulated Amp models. Accordingly, the British Crunch, British Lead, American Clean, Crunch, and OverDrive select the appropriate Amp models, and apply new EQ and other settings. You can, however, combine any Amp model with any EQ or other parameter settings, as required.

At the top of the slider section, you will find the *Pre Gain* control, used to set the pre-amplification level of the input signal. This control has different effects, dependent on the selected Amp model. As an example: A maximum *Pre Gain* setting produces a powerful crunch sound when used in conjunction with the British Clean Amp model, but the same *Pre Gain* setting results in a heavy distortion—suitable for lead sounds—with the British or American Gain Amp models.

Directly below, you will find the *Low, Mid,* and *High* controls. Use of these sliders allows you to adjust the frequency ranges of the bass, mids, and treble as desired.

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

The *Master* slider 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.

The *Output Level* slider serves as a final level control for Guitar Amp's output. It 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

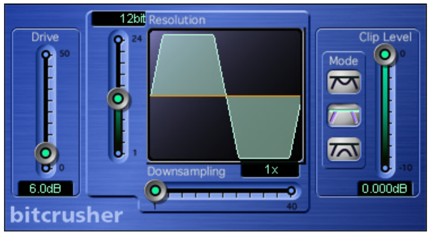
The Overdrive effect emulates the distortion of a field-effect transistor (FET). When saturated, FETs generate warmer sounding distortion than bipolar transistors.

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.

If you activate the 001/011 button, you have access to two more parameters: 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 you active the 001/011 button, the *Phase Reverse* parameter will be shown. 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.

Filter

6

This chapter covers Logic's filter effects.

The filter effects include the AutoFilter, Fuzz-Wah, Low/High Pass Filter, and Low/High Cut plug-ins.

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.

If you active the 001/011 button of the Autofilter plug-in, you will have access to the following five parameters:

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 Logic Pro 7 EVD6's Wah effect. Its parameters 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*, *Morl1*, *Morl2*.

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.

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*.

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*

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.

Delay 7

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 it's 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.

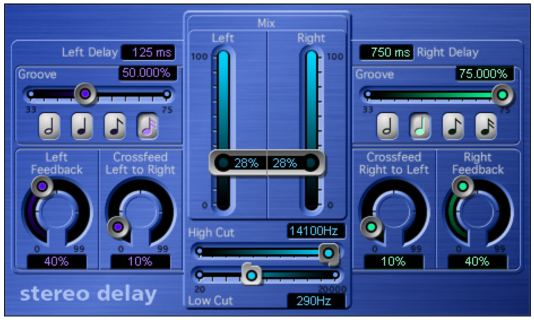
The *Mix* slider determines the balance between the original (dry) signal and effects (wet) signals. If you've inserted the Tape Delay in an individual track, you'll generally find that settings of up to 50% are desirable. If the Tape Delay is patched to the insert of a Bus channel, and you're routing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

If you active the 001/011 button in the plug-in header, three more parameters will be shown: 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.

Activating the 001/011 button in the plug-in header will display ten additional parameters.

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.

Modulation

8

This chapter introduces Logic's modulation effects.

This includes the Modulation Delay, Chorus, Flanger, Phaser, Tremolo, 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.

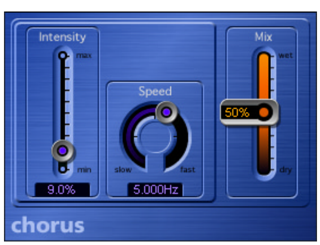
The *Feedback* slider determines the intensity at which the effect's signal feedback is routed to the input. If you're going for radical flanging effects, enter a high *Feedback* value. If simple doubling is what you're after, you won't want any feedback at all. The *Mix* slider determines the balance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

The *001/011 button* 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.

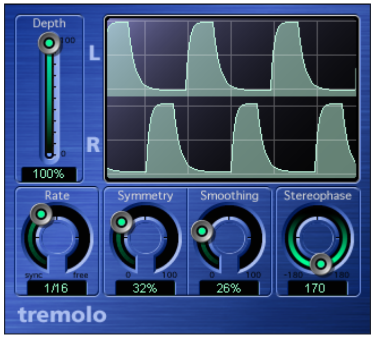
The modulation section offers two LFOs, featuring individually variable frequencies, and freely variable mix options (*LFO Mix*). Additionally, the frequency of LFO 1 can be modulated by the level of the input signal. Use the *Envelope Modulation* slider to set the desired modulation intensity. By staking 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.

If you activate the 001/011 button in the plug-in header, you will have access to the following six additional parameters:

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*.

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 (–180º) the balance wanders from left to right. When set to 180º, 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.

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.

Reverb

9

This chapter describes Logic's reverb effects.

This includes AVerb, SilverVerb, GoldVerb, and PlatinumVerb

AVerb

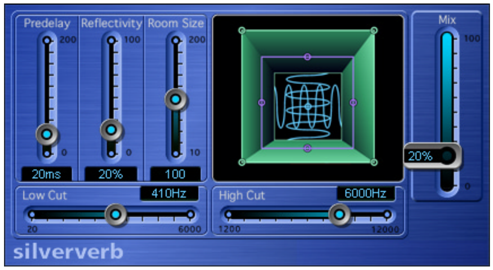
Although the AVerb is based on a simple reverb algorithm, it delivers remarkably good results.

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.

If you activate the 001/011 button in the plug-in header, four additional parameters will be available:

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

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.

Note: The Diffusion parameter is only available if you activate the 001/011 button in the plug-in header.

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 61. 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 001/011 button 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 63.

Special

10

This chapter introduces Logic's special plug-ins.

This includes the Pitch Shifter II and Denoiser plug-ins.

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.

If you activate the 001/011 button in the plug-in header, you can create your own presets, using the additional *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.

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.

Helper

11

This chapter introduces you to Logic's Helper plug-ins.

This includes the Tuner, Gain, and Levelmeter plug-ins.

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.

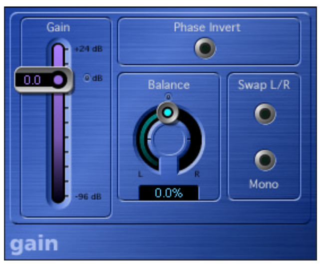
Use couldn't be simpler. There is a single tuning control at the bottom of the ET1 Tuner interface. To adjust, simply click and drag it to the desired pitch for A. By default, the ET1 Tuner is set to concert pitch A = 440 Hz.

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

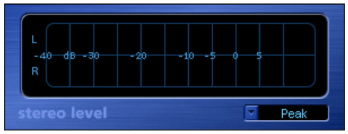
This button swaps the left and right output channels. It is placed after the Stereo Balance in the signal path.

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 → m, m → s and s → s configurations. In m → m and m → s modes, only one *Phase Invert* button is available. In the m → m version, the *Stereo Balance, Swap Left/Right* and *Mono* parameters are disabled.

Levelmeter



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.

The current peak values are displayed numerically (in dB increments), next to the Level Meter. The values are reset by clicking into the display.

The *Level Meter* plug-in is switchable between Peak and RMS characteristics.

Synthesizer Basics

12

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?

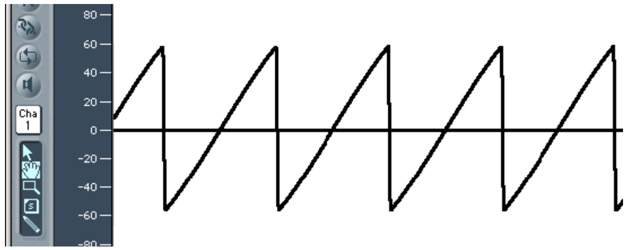
Before we start, synthesis in this context, is the (re)production of a sound which emulates, or synthesizes the sound of another instrument, a voice, helicopter, car, dog bark—in fact, any sound you can think of!

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.

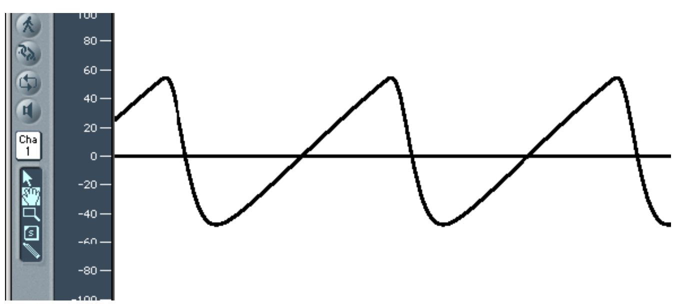
Cutoff and Resonance—Illustrated With a Sawtooth Wave



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.

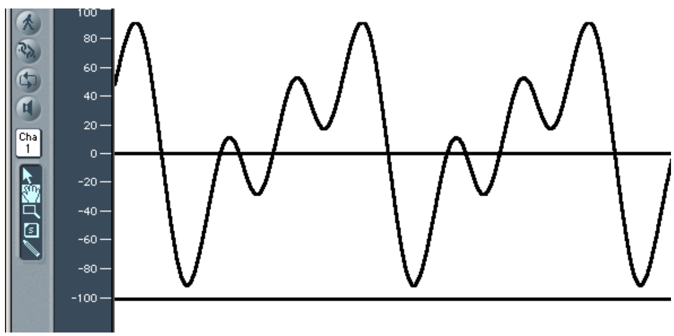


The wave length here is not really higher, but the zoom setting is.

Fourier Theorem and Harmonics

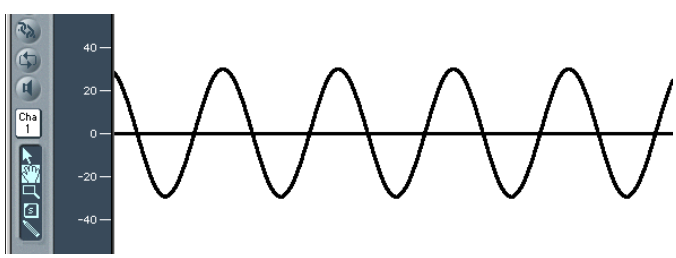
"Every periodic wave can be seen as the sum of sine waves with certain wave lengths and amplitudes, the wave lengths of which have harmonic relations (ratios of small numbers)". This is known as the Fourier theorem. Roughly translated into more musical terms, this means that any tone with a certain pitch 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:



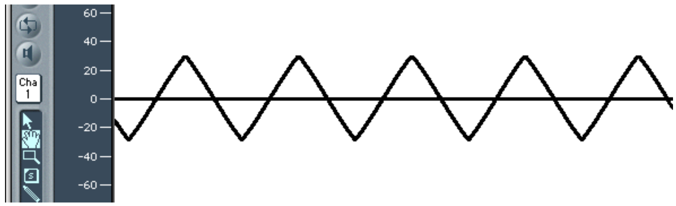
The effect of the resonating filter is comparable to a graphic equalizer with all faders higher than 660 Hz pulled all the way down, but with only 660 Hz (*Cutoff Frequency*) pushed to its maximum position (*resonance*). The faders for frequencies below 660 Hz remain in the middle (0 dB).

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

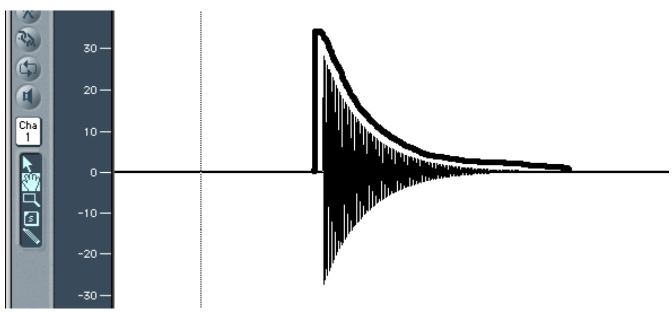
Waveforms (waves) are named sawtooth, square, pulse, or triangular because of their shape when displayed as an oscillogram (as in Logic's Sample Editor). This is the triangular wave:



The triangular wave has few harmonics—which is evident by the fact that is shaped more like a sine than a sawtooth wave. This wave contains only odd harmonics—which means no octaves.

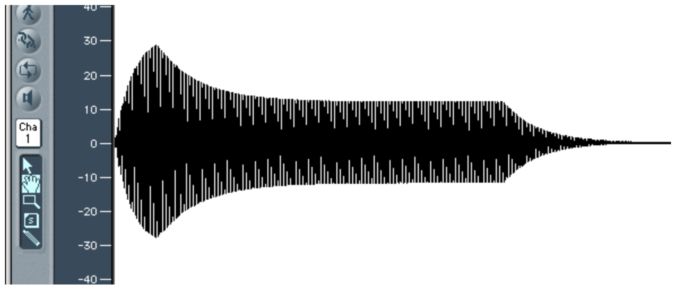
Envelopes

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 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 it's 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.

EFM 1

13

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 use 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 clock-wise 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.

ES M

14

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.

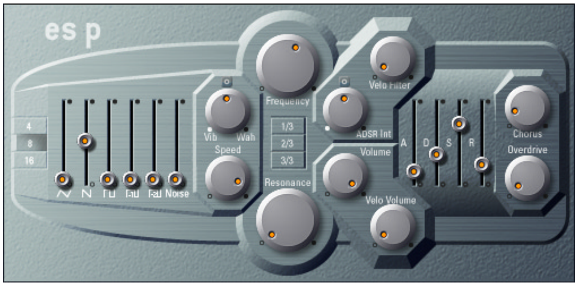
Overdrive
This parameter sets the overdrive/distortion level for the ES M output. Caution: The overdrive effect significantly increases the output level.

ES P

15

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.

ES E

16

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).

ES1

17

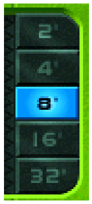
This chapter introduces Logic's virtual analog ES1 synthesizer.

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 99). 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*.

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.

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 101). 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.

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

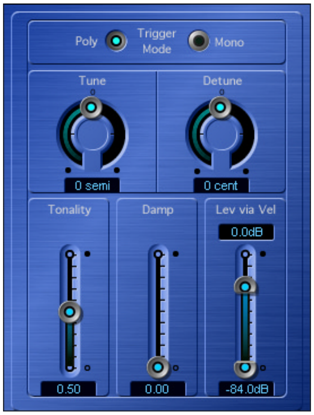
The ES1 offers classic stereo Chorus/Ensemble effects. There are four possible settings: *Off*, *C1*, *C2,* and *Ens*.

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.

KlopfGeist

18

KlopfGeist is an instrument that is optimized to provide a metronome click in Logic.



KlopfGeist is inserted on Audio Instrument channel 16 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 16. 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 16 *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 16 is fitted with a Ghost Buster facility—the button labeled *M*.

EXSP24

19

This chapter introduces Logic's EXSP24 sampler.

The EXSP24 sample player offers all of the playback 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 EXSP24 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.

The EXSP24 sample player does not allow new Sampler Instruments to be created, nor editing of existing EXS Instruments (apart from synthesis and modulation options). This functionality, including all sample editing, routing, and instrument creation options is only available in the full EXS24 sampler plug-in, found in Logic Pro.

The EXSP24 is compatible with the EXS24, AKAI S1000, and S3000, SampleCell, WAV, AIF(F), Gigasampler, and SoundFont2 sample formats, as well as the Vienna Library, allowing access to large and comprehensive sampler libraries.

The EXSP24 offers numerous synthesis options, enabling you to tailor sounds to meet your needs.

Last, but not least: as a highly optimized Logic instrument, the EXSP24 offers great performance, even on slower machines. The EXSP24'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 EXSP24 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 EXSP24, 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 EXSP24. 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 EXSP24 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 EXSP24'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.

File Organization

File Types and File Organization

The EXSP24 uses the following file types and hierarchical structures:

Audio File

A single sample on your hard disk. The EXSP24 is compatible with all audio file formats supported by Logic.

Sampler Instrument

A Sampler Instrument points to one or more audio files, and organizes them as multi samples or drum maps, respectively.

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.

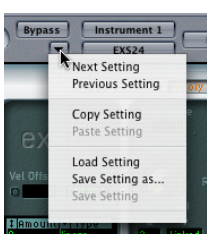
A Sampler Instrument is the file type that is loaded into the EXSP24 for playing. When you select a Sampler Instrument in the EXSP24's pull-down menu, the associated audio files are automatically located on the hard disk, and are subsequently loaded into your computer's RAM.

In order to be visible within the EXSP24'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 107.

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 EXSP24 is no exception. The Settings for the EXSP24 are stored in the *EXSP24* 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

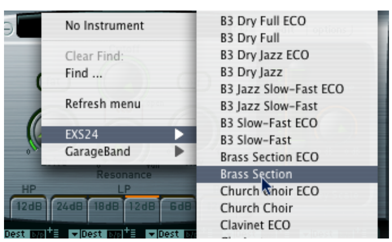
As your sample library grows, the list of Sampler Instruments will also expand. To aid you in keeping the list of Sampler Instruments manageable, the EXSP24 features a sophisticated, but easy to use method of file management.

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 EXSP24 Sampler Instruments into this newly created folder. Their menu structure will be reflected when clicking on the EXSP24 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 EXSP24 Instruments

This feature allows all EXSP24 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 EXSP24 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 EXSP24 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 EXSP24 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 EXSP24" 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 EXSP24 instance is opened; a new hierarchical menu item will appear within the EXSP24 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 EXSP24 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.

Searching for Sampler Instruments

As a further navigational enhancement, the EXSP24 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 EXSP24 is compatible with the EXS(P)24, AKAI S1000 and S3000, SampleCell, 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 107). The EXSP24 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 EXSP24 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 re-load 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 EXSP24 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.

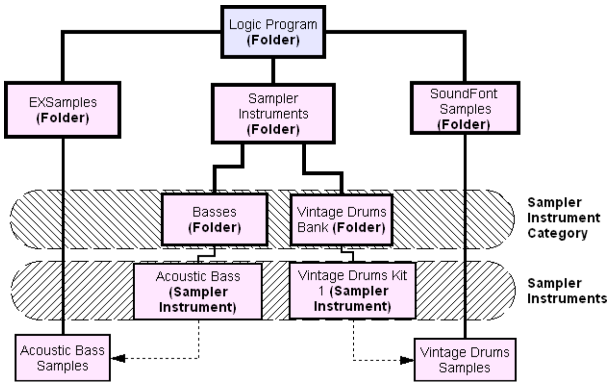
All sounds contained in the bank will automatically have an EXS Sampler Instrument file created and placed into the newly created *Bank* folder. The EXSP24 Sampler Instrument pull-down menu will automatically be updated to reflect the new folder hierarchy. All samples associated with the *Bank* will 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 EXSP24. 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 EXSP24.

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 EXSP24 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 EXSP24 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 EXSP24 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 EXSP24 folder hierarchy diagram above.

Converting AKAI Files

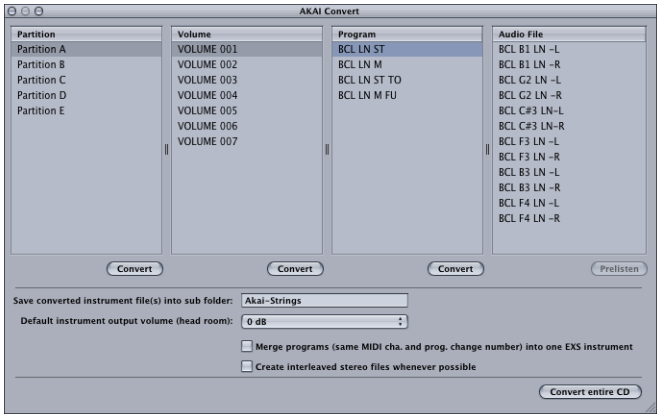
This section discusses the AKAI import procedure. The EXSP24 can import samples saved in the AKAI S1000 and S3000 sample formats. The AKAI Convert function can be used to import:

- 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 107.



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.

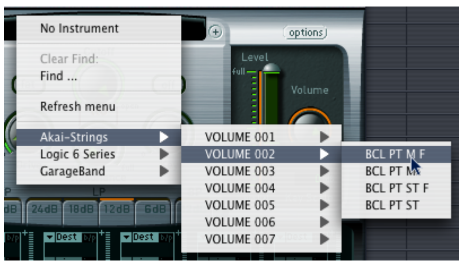
6 Any audio files imported will be stored within a folder which matches the name of the *Volume*. This folder is created within the *Logic > AKAI Samples* folder. The Sampler Instrument(s) created by the import procedure matches the Program name(s). It is placed inside the Sampler Instruments folder, or a sub folder as determined by the *Save converted instrument file(s) into sub folder* parameter discussed in "AKAI File Organization" on page 116.

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 EXSP24'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 116, 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 EXSP24 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 EXSP24. 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 EXSP24 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 EXSP24:

- 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 EXSP24 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 EXSP24 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 EXSP24 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.

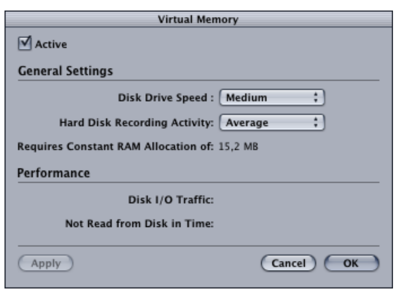
You will find plus (+) and minus (–) buttons to the left and right of the Instrument Load pull-down menu/display. These buttons allow you to browse to the next/previous Instrument (sound) of your sound library (if necessary, this will change folders in accordance with their order of appearance in the menu). Please note that the global *Next/Previous EXS Instrument* key and MIDI commands also perform the same function.

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 mkI 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 "EXSP24 mkI Modulation Paths" section, from page 131 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 115).
- *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 132 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 EXSP24 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 129).

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 EXSP24. 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 EXSP24 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.

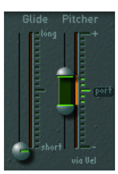
Remote

The *Remote* parameter allows you to easily pitch complete EXSP24 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 EXSP24 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.

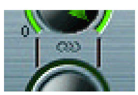
Cutoff

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 EXSP24. 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 EXSP24 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.

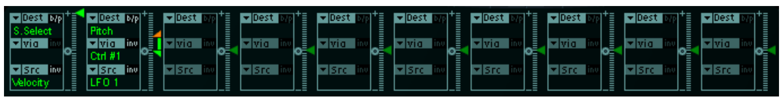
LFO 2 Rate

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

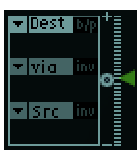
There is a third LFO available which always uses a triangular waveform. LFO 3 can oscillate freely between 0 and 35 Hz, or can be tempo synchronized in values between 32 bars and 1/128 triplets.

Modulation Matrix



The Modulation Matrix is the dark horizontal band that spans the center of the EXSP24 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 EXSP24 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.

EXSP24 mkI Modulation Paths

Many of the hard-wired modulation paths that were available as sliders on the original EXSP24 (mkI) are now part of the Modulation Matrix. In order to reconstitute the modulation slider configuration of the mkI version, click on the *options* button in the upper-right corner and choose *(Recall default EXS24 mkI settings)* from the pop-up menu. This will load the mkI 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 EXSP24 mkI, for example (see the complete list of sources and destinations below).

EXSP24 mkI Backward Compatibility

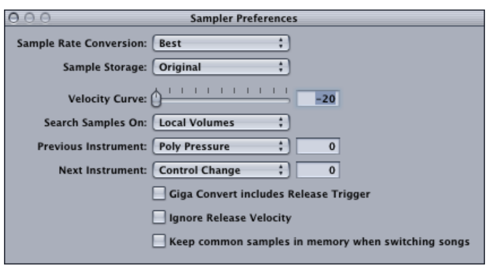
For technical reasons, the settings of the Modulation Matrix can not translate backwards to the EXSP24 mkI.

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
Relative Volume (auto adjust)	Key

Modulation Sources	Modulation Destinations
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.

Preferences



The Preferences window allows you to:

Choose the interpolation quality used by the EXSP24. 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 EXSP24 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 EXSP24 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 for24 Bit samples.

The *Velocity Curve* parameter determines the EXSP24'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 EXSP24 to find and load Sampler Instruments and files.

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 EXSP24 mkI, 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.

EXSP24 Key Commands

A number of key commands are available for the EXSP24 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.

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 EXSP24 is proof of this…

MIDI Controller List

Common	Mono Mode	71
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GarageBand Instruments

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GarageBand Instruments are automatically installed with Logic. You can insert them as per other software instruments.

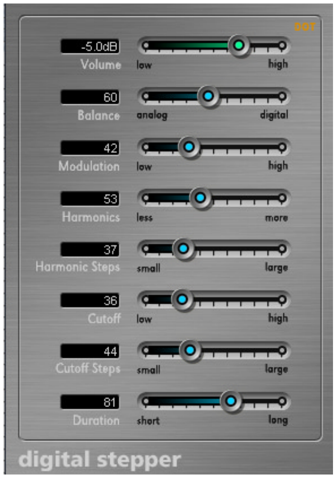
GarageBand Instruments are accessible from the *Stereo* > *Logic* > *GarageBand Instruments* sub-menu.

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 (Pro) 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 (EXSP24).

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 EXSP24, ES1 (or other equivalent Logic instrument) instance simultanously.

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!

Glossary

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.

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.

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.

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 Mac OS X 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.

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.

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.

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.

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 knows as 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 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 you 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.

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 EXSP 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.

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.

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.

PWM Pulse Width Modulation. Synthesizers often feature this facility, where a square waveform is deformed by adjusting it's 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.

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 EXSP24 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.

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.

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).

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.

transpositon 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.

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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