

Logic Studio Instruments and Effects









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Introduction to the Logic Studio Plug-ins

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

These include innovative synthesizers, high quality effect plug-ins, a powerful sampler, and authentic recreations of vintage instruments.

This manual will introduce you to the individual effects and instruments—and their parameters. All plug-in parameters are discussed in detail. The instrument chapters include a number of tutorials that will help you to make the most of your new instruments. Using plug-ins is much easier if you are familiar with the basic functions of the individual applications included in Logic Studio. Information about these can be found in the respective user manuals.

This manual covers all plug-ins available in the Logic Studio applications. Not all of them are available in all individual applications, however. Please refer to the following tables to see which plug-ins are available in each application.

Logic Pro Effects and Instruments

The following tables outline the effects and instruments included with Logic Pro.

Effect category	Included effects
Amp Modeling	Bass Amp (p. 19)Guitar Amp Pro (p. 21)
Delay	 Delay Designer (p. 28) Echo (p. 46) Sample Delay (p. 46) Stereo Delay (p. 47) Tape Delay (p. 48)
Distortion	 Bitcrusher (p. 52) Clip Distortion (p. 53) Distortion (p. 54) Distortion II (p. 55) Overdrive (p. 56) Phase Distortion (p. 57)

Effect category	Included effects
Dynamic	 Adaptive Limiter (p. 61) Compressor (p. 62) DeEsser (p. 65) Ducker (p. 67) Enveloper (p. 68) Expander (p. 70) Limiter (p. 71) Multipressor (p. 72) Noise Gate (p. 76) Silver Compressor (p. 78) Silver Gate (p. 79) Surround Compressor (p. 79)
EQ	 Channel EQ (p. 84) DJ EQ (p. 88) Fat EQ (p. 88) Linear Phase EQ (p. 90) Match EQ (p. 91) Single Band EQs (p. 96) Silver EQ (p. 97)
Filter	 AutoFilter (p. 100) EVOC 20 Filterbank (p. 103) EVOC 20 TrackOscillator (p. 107) Fuzz-Wah (p. 116) Spectral Gate (p. 118)
Imaging	 Binaural Post-Processing Plug-in (p. 121) Direction Mixer (p. 123) Stereo Spread (p. 125)
Metering	 BPM Counter (p. 128) Correlation Meter (p. 128) Level Meter (p. 129) MultiMeter (p. 129) Surround MultiMeter (p. 133) Tuner (p. 134)
Modulation	 Chorus (p. 136) Ensemble (p. 136) Flanger (p. 137) Microphaser (p. 138) Modulation Delay (p. 138) Phaser (p. 140) Ringshifter (p. 142) Rotor Cabinet (p. 147) Scanner Vibrato (p. 148) Spreader (p. 149) Tremolo (p. 150)
Pitch	 Pitch Correction (p. 151) Pitch Shifter II (p. 155) Vocal Transformer (p. 156)

Effect category	Included effects
Reverb	 AVerb (p. 160) EnVerb (p. 161) GoldVerb (p. 162) PlatinumVerb (p. 164) SilverVerb (p. 167) Convolution Reverb: Space Designer (p. 169)
Specialized	 Denoiser (p. 192) Enhance Timing (p. 193) Exciter (p. 194) Grooveshifter (p. 195) Speech Enhancer (p. 196) SubBass (p. 197)
Utility	 Down Mixer (p. 199) Gain (p. 200) I/O (p. 201) Test Oscillator (p. 202)

The following table outlines the instruments included with Logic Pro.

Instrument category	Included instruments
Synthesizer	 EFM1 (p. 229) ES E (p. 235) ES M (p. 237) ES P (p. 239) ES1 (p. 241) ES2 (p. 249) Klopfgeist (p. 453) Sculpture (p. 455)
Drum synthesizer	Ultrabeat (p. 553)
Software sampler	EXS24 mkll (p. 391)
Vocoder synthesizer	EVOC 20 PolySynth (p. 205)
Vintage instruments	EVB3 (p. 323)EVD6 (p. 359)EVP88 (p. 379)
Utility	External Instrument (p. 451)
GarageBand instruments	Analog Basic, Analog Mono, Analog Pad, Analog Swirl, Analog Sync, Bass, Digital Basic, Digital Mono, Digital Stepper, Drum Kits, Electric Clavinet, Electric Piano, Guitar, Horns, Hybrid Basic, Hybrid Morph, Piano, Sound Effects, Strings, Tonewheel Organ, Tuned Percussion, Voice, Woodwind (see "GarageBand Instruments" on page 615)

Soundtrack Pro Effects

The following table outlines the effects included with Soundtrack Pro.

Note: Effects included in Soundtrack Pro do not feature the extended parameters that are covered in this document.

Effect category	Included effects
Delay	Stereo Delay (p. 47)Tape Delay (p. 48)
Distortion	 Bitcrusher (p. 52) Clip Distortion (p. 53) Distortion (p. 54) Distortion II (p. 55) Overdrive (p. 56) Phase Distortion (p. 57)
Dynamic	 Adaptive Limiter (p. 61) Compressor (p. 62) DeEsser (p. 65) Enveloper (p. 68) Expander (p. 70) Limiter (p. 71) Multipressor (p. 72) Noise Gate (p. 76) Surround Compressor (p. 79)
EQ	 Channel EQ (p. 84) Fat EQ (p. 88) Linear Phase EQ (p. 90) Match EQ (p. 91) Single Band EQs (p. 96)
Filter	AutoFilter (p. 100)Spectral Gate (p. 118)
Imaging	Direction Mixer (p. 123)Stereo Spread (p. 125)
Metering	 Correlation Meter (p. 128) MultiMeter (p. 129) Surround MultiMeter (p. 133) Tuner (p. 134)
Modulation	 Chorus (p. 136) Ensemble (p. 136) Flanger (p. 137) Modulation Delay (p. 138) Phaser (p. 140) Ringshifter (p. 142) Scanner Vibrato (p. 148) Tremolo (p. 150)
Pitch	Pitch Shifter II (p. 155)Vocal Transformer (p. 156)

Effect category	Included effects			
Reverb	 PlatinumVerb (p. 164) Soundtrack Pro Reverb (p. 168) Convolution Reverb: Space Designer (p. 169) 			
Specialized	 Denoiser (p. 192) Exciter (p. 194) SubBass (p. 197) 			
Utility	 Gain (p. 200) Multichannel Gain (p. 202) Test Oscillator (p. 202) 			

WaveBurner Effects

The following table outlines the effects included with WaveBurner.

Note: WaveBurner does not provide a project tempo, and therefore, does not support tempo-based effect parameters (such as sync).

Effect category	Included effects
Amp Modeling	Bass Amp (p. 19)Guitar Amp Pro (p. 21)
Delay	 Delay Designer (p. 28) Sample Delay (p. 46) Stereo Delay (p. 47) Tape Delay (p. 48)
Distortion	 Bitcrusher (p. 52) Clip Distortion (p. 53) Distortion (p. 54) Distortion II (p. 55) Overdrive (p. 56) Phase Distortion (p. 57)
Dynamic	 Adaptive Limiter (p. 61) Compressor (p. 62) DeEsser (p. 65) Enveloper (p. 68) Expander (p. 70) Limiter (p. 71) Multipressor (p. 72) Noise Gate (p. 76) Silver Compressor (p. 78) Silver Gate (p. 79)
EQ	 Channel EQ (p. 84) DJ EQ (p. 88) Fat EQ (p. 88) Linear Phase EQ (p. 90) Match EQ (p. 91) Single Band EQs (p. 96) Silver EQ (p. 97)

Effect category	Included effects
Filter	 AutoFilter (p. 100) EVOC 20 Filterbank (p. 103) Fuzz-Wah (p. 116) Spectral Gate (p. 118)
Imaging	Direction Mixer (p. 123)Stereo Spread (p. 125)
Metering	 BPM Counter (p. 128) Correlation Meter (p. 128) Level Meter (p. 129) MultiMeter (p. 129) Tuner (p. 134)
Modulation	 Chorus (p. 136) Ensemble (p. 136) Flanger (p. 137) Microphaser (p. 138) Modulation Delay (p. 138) Phaser (p. 140) Ringshifter (p. 142) Rotor Cabinet (p. 147) Scanner Vibrato (p. 148) Spreader (p. 149) Tremolo (p. 150)
Pitch	Pitch Correction (p. 151)Pitch Shifter II (p. 155)Vocal Transformer (p. 156)
Reverb	 AVerb (p. 160) EnVerb (p. 161) GoldVerb (p. 162) PlatinumVerb (p. 164) SilverVerb (p. 167) Convolution Reverb: Space Designer (p. 169)
Specialized	 Denoiser (p. 192) Exciter (p. 194) Speech Enhancer (p. 196) SubBass (p. 197)
Utility	Gain (p. 200)

MainStage Instruments and Effects

The following tables outline the effects and instruments included with MainStage.

Note: MainStage is a live application, and therefore, does not include effect plug-ins that introduce a noticeable amount of latency. A further exclusion is the EXS24 mkll Instrument Editor.

Effect category	Included effects			
Amp Modeling	Bass Amp (p. 19)Guitar Amp Pro (p. 21)			
Delay	 Delay Designer (p. 28) Echo (p. 46) Sample Delay (p. 46) Stereo Delay (p. 47) Tape Delay (p. 48) 			
Distortion	 Bitcrusher (p. 52) Clip Distortion (p. 53) Distortion (p. 54) Distortion II (p. 55) Overdrive (p. 56) Phase Distortion (p. 57) 			
Dynamic	 Compressor (p. 62) DeEsser (p. 65) Ducker (p. 67) Enveloper (p. 68) Expander (p. 70) Limiter (p. 71) Multipressor (p. 72) Noise Gate (p. 76) Silver Compressor (p. 78) Silver Gate (p. 79) 			
EQ	 Channel EQ (p. 84) DJ EQ (p. 88) Fat EQ (p. 88) Single Band EQs (p. 96) Silver EQ (p. 97) 			
Filter	 AutoFilter (p. 100) EVOC 20 Filterbank (p. 103) EVOC 20 TrackOscillator (p. 107) Fuzz-Wah (p. 116) 			
Imaging	Direction Mixer (p. 123) Stereo Spread (p. 125)			
Metering	 BPM Counter (p. 128) Correlation Meter (p. 128) Level Meter (p. 129) MultiMeter (p. 129) Tuner (p. 134) 			

Effect category	Included effects
Modulation	 Chorus (p. 136) Ensemble (p. 136) Flanger (p. 137) Microphaser (p. 138) Modulation Delay (p. 138) Phaser (p. 140) Ringshifter (p. 142) Rotor Cabinet (p. 147) Scanner Vibrato (p. 148) Spreader (p. 149) Tremolo (p. 150)
Pitch	Pitch Shifter II (p. 155)
Reverb	 AVerb (p. 160) EnVerb (p. 161) GoldVerb (p. 162) PlatinumVerb (p. 164) SilverVerb (p. 167) Convolution Reverb: Space Designer (p. 169)
Specialized	Exciter (p. 194)SubBass (p. 197)
Utility	 Gain (p. 200) Test Oscillator (p. 202)

The following table outlines the instruments included with MainStage. \\

Instrument category	Included instruments
Synthesizer	 EFM1 (p. 229) ES E (p. 235) ES M (p. 237) ES P (p. 239) ES1 (p. 241) ES2 (p. 249) Klopfgeist (p. 453) Sculpture (p. 455)
Drum synthesizer	Ultrabeat (p. 553)
Software sampler	EXS24 mkll (p. 391)
Vocoder synthesizer	EVOC 20 PolySynth (p. 205)
Vintage instruments	EVB3 (p. 323)EVD6 (p. 359)EVP88 (p. 379)
GarageBand instruments	Analog Basic, Analog Mono, Analog Pad, Analog Swirl, Analog Sync, Bass, Digital Basic, Digital Mono, Digital Stepper, Drum Kits, Electric Clavinet, Electric Piano, Guitar, Horns, Hybrid Basic, Hybrid Morph, Piano, Sound Effects, Strings, Tonewheel Organ, Tuned Percussion, Voice, Woodwind (see "GarageBand Instruments" on page 615)

Amp Modeling 1

You can add the sound of a guitar and bass amplifier to your audio recordings and software instruments.

Using a method known as component modeling, both the sound and functionality of musical instrument amplifiers, particularly those used with electric guitar and bass, can be emulated as an effect. These effects recreate the sound of both tube and solid state amplifiers, and feature a full set of controls, including pre-gain and tone controls for bass, midrange, and treble, as well as output level. They allow you to select from a variety of familiar amp models.

The following sections describe the individual plug-ins included with Logic Studio.

- "Bass Amp" on page 19
- "Guitar Amp Pro" on page 21

Bass Amp

The Bass Amp simulates the sound of several famous bass amplifiers. You can process bass guitar signals directly within Logic Pro and reproduce the sound of high-quality bass guitar amplification systems.

You can also use the Bass Amp for experimental sound design. You may freely use the plug-in on other instruments, as desired—applying the sonic character of a bass amp to a vocal or drum part, for example.

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Bass Amp Parameters



• *Model pop-up menu:* Choose from among nine different amplifier models. The choices are:

American Basic 1970s-era American bass amp, equipped with eight 10-is suited for blues and rock recordings. American Deep Based on the American Basic amp, but with strong lower (from 500 Hz on) emphasis. Well suited for reggae and public american Scoop Based on American Basic amp, but combines the frequency of the American Deep and American Bright, with both 1 500 Hz and upper mid (from 4.5 Hz) frequencies amples	er-mid frequency pop recordings.
American Scoop Based on American Basic amp, but combines the freque of the American Deep and American Bright, with both I	oop recordings.
of the American Deep and American Bright, with both I	ency characteristics
500 Hz) and upper mid (from 4.5 kHz) frequencies empl for funk and fusion recordings.	ow mid (from
American Bright Based on the American Basic amp, this model massively upper-mids (from 4.5 kHz upwards).	emphasizes the
New American Basic 1980s-era American bass amp, well suited for blues and	rock recordings.
New American Bright Based on the New American Basic amp, this model strong the frequency range above 2 kHz. Well suited for rock and the frequency range above 2 kHz.	• .
Top Class DI Warm Famous DI box simulation, well suited for reggae and positive Mids, in the broad frequency range between 500 and 50 emphasized.	
Top Class DI Deep Based on the Top Class DI Warm amp, this model is well and fusion its mid frequency range is strongest around.	
Top Class DI Mid Based on the Top Class DI Warm amp, this model feature linear frequency range, with no frequencies emphasized blues, rock, and jazz recordings.	

- Pre Gain slider: Sets the pre-amplification level of the input signal.
- Bass, Mid, and Treble sliders: Adjusts the bass, mid, and treble levels.

- Mid Frequency slider: Sets the center frequency of the mid band (between 200 Hz and 3000 Hz).
- Output Level slider: Sets the final output level for the Bass Amp.

Guitar Amp Pro

The Guitar Amp Pro can emulate the sound of a variety famous guitar amplifiers and the cabinets/speakers used with them. You can process guitar signals directly within Logic Pro, allowing you to reproduce the sound of high-quality guitar amp systems.

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

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

The Guitar Amp Pro window is organized into four main sections.



- The Amp section has parameters for choosing the Amp, Speaker, and EQ model, and a set of tone, gain, and level controls.
- The Effects (FX) section is where you control the built-in guitar effects. Below the FX section is the final output control.

- The Microphone Position section is where you set the position of the microphone on the speaker.
- The Microphone Type section is where you choose which type of microphone captures the amp's sound.

Amp Section

• Amp pop-up menu: Choose the amp model you want to use. The choices are:

Model	Description		
UK Combo 30W	Neutral sounding amp, well suited for clean or crunchy rhythm parts.		
UK Top 50W	Quite aggressive in the high frequency range, well suited for classical rock sounds.		
US Combo 40W	Clean sounding Amp model, well suited for funk and jazz sounds.		
US Hot Combo 40W	Emphasizes the high mids of the frequency range, making this model ideal for solo sounds.		
US Hot Top 100W	This Amp produces very fat sounds, even at low <i>Master</i> settings, than result in broad sounds with a lot of "oomph."		
Custom 50W	With the Presence parameter set to 0, this Amp model is well suited for smooth fusion lead sounds.		
British Clean	Simulates the classic British Class A combos used continuously since the 1960s for rock music, without any significant modification. This model is ideally suited for clean or crunchy rhythm parts.		
British Gain	Emulates the sound of a British tube head, and is synonymous with rocking, powerful rhythm parts and lead guitars with a rich sustain.		
American Clean	Emulates the traditional full tube combos used for clean and crunchy sounds.		
American Gain	Emulates a modern Hi-Gain head, making it suitable for distorted rhythm and lead parts.		
Clean Tube Amp	Emulates a tube amp model with very low gain (distortion only when using very high input levels or Gain/Master settings).		

• Speaker pop-up menu: Choose one of the 15 speaker models. The choices are:

Speaker type	Description			
UK 1x12 open back	Classic open enclosure with one 12" speaker, neutral, well-balanced, multifunctional.			
UK 2x12 open back	Classic open enclosure with two 12" speaker, neutral, well-balanced, multifunctional.			
UK 2x12 closed	Loads of resonance in the low frequency range, therefore well suited for Combos: crunchy sounds are also possible with low Bass control settings.			
UK 4x12 closed slanted	when used in combination with off-center miking, you will get an interesting mid frequency range; therefore this model works well when combined with High Gain amps.			

Speaker type	Description	
US 1x10 open back	Not much resonance in the low frequency range. Suitable for use with (blues) harmonicas.	
US 1x12 open back 1	Open enclosure of an American lead combo with a single 12" speaker.	
US 1x12 open back 2	Open enclosure of an American clean/crunch combo with a single 12" speaker.	
US 1x12 open back 3	Open enclosure of another American clean/crunch combo with a single 12" speaker.	
US broad range	Cabinet simulation of a classic electric piano speaker.	
Analog simulation	Internal speaker simulation of a well-known British 19" tube preamplifier.	
UK 1x12	A British Class A tube open back with a single 12" speaker.	
UK 4x12	Classic closed enclosure with four 12" speakers (black series), suitable for Rock.	
US 1x12 open back	Open enclosure of an American lead combo with a single 12" speaker.	
US 1x12 bass reflex	Closed bass reflex cabinet with a single 12" speaker.	
DI Box	This option allows you to bypass the speaker simulation section.	

- *EQ pop-up menu:* Choose one of the four EQ models. The choices are: British 1, British 2, American, and Modern EQ.
- Amp–Speaker Link button: Links the Amp and Speaker menus so that when you change the amp model, the speaker associated with that amp is loaded automatically.
- Amp-EQ Link button: Links the Amp and EQ menus so that when you change the
 amp model, the EQ model associated with that amp is loaded automatically.
 Each amp model has a speaker and EQ model associated with it. Together, the amp,
 speaker, and EQ combined recreate a well-known guitar sound. However, you can
 freely combine any speaker or EQ model with any amp by turning off the two Link
 buttons.
- Gain knob: Sets the amount of pre-amplification applied to the input signal. This
 control has different effects, dependent on which Amp model is selected. For
 example, when using the British Clean amp model, the maximum Gain setting
 produces a powerful crunch sound. When using the British Gain or Modern Gain
 amps, the same Gain setting produces heavy distortion, suitable for lead solos.
- Bass, Mids, and Treble knobs: Adjusts the frequency ranges of the EQ models, similar to the tone knobs on a hardware guitar amplifier.
- *Presence knob:* Adjusts the high frequency range. The Presence parameter affects only the output (Master) stage of Guitar Amp Pro.

Master knob: Sets the output volume of the amplifier (going to the speaker).
 Typically, for tube amplifiers, increasing the Master level produces a more compressed and saturated sound, resulting in a more distorted and powerful (louder) signal. High settings can produce an extremely loud output. In Guitar Amp Pro, the Master parameter modifies the sonic character, and the final output level is set using the Output parameter below the FX section. (see below for information).

Effects Section

The Effects section contains Reverb, Tremolo, and Vibrato effects. You can choose either Tremolo (which modulates the amplitude or volume of the sound) or Vibrato (which modulates the pitch), and use Reverb together with either one, or separately.

Before you can use or adjust an effect, you must first turn it on by clicking its On button (with a power on icon). The On button lights when the effect is turned on. The FX and Reverb On buttons are located to the left of the controls for each effect.

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

FX Parameters

- FX pop-up menu: Choose either Tremolo or Vibrato from the menu.
- *Depth knob:* Sets the intensity of the modulation.
- *Speed knob*: Sets the speed of the modulation (in Hz). Lower settings produce a smooth and floating sound, while higher settings produce a rotor-like effect.
- Sync button: When turned on, the Speed is synchronized to the project tempo. When Sync is activate, adjusting the Speed parameter lets you select different musical note values. Set the Speed parameter to the desired value, and whichever effect you have chosen will be perfectly synchronized to the project tempo.

Reverb Parameters

- Reverb pop-up menu: Choose one of the three types of spring reverb.
- Level knob: Sets the amount of reverb applied to the pre-amplified amp signal.

Microphone Position and Microphone Type Sections

After choosing a speaker from the Speaker menu, you can set the type of microphone emulated, and where the microphone is placed in relation to the speaker.

Microphone Position Parameters

- Centered button: When selected, places the microphone in the center of the speaker cone, also called on-axis. This placement produces a fuller, more powerful sound, suitable for blues or jazz guitar tones.
- Off-Center button: When selected, places the microphone on the edge of the speaker, also referred to as off-axis. This placement produces signal a tone that is brighter and sharper, but also thinner, suitable for cutting rock or rhythm and blues guitar tones.

When you select either button, the graphic speaker display reflects the current setting.

Microphone Type Parameters

- Condenser button: When selected, emulates the sound of a studio condenser microphone. The sound of condenser microphones is fine, transparent, and well balanced.
- Dynamic button: When selected, emulates the sound of a dynamic cardioid microphone. This microphone type sounds brighter and more cutting, compared to the Condenser model. At the same time, the lower Mids are less pronounced, making this model more suitable for miking rock guitar tones.

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

Output

Below the Effects section is the Output slider, which serves as the final level control for Guitar Amp Pro output. The Output parameter can be thought of as a "behind the cabinet" volume control, and is used to set the level that is fed into the following plugin slots on the channel or into the channel output.

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

Delay 2

Delay effects store the input signal—and hold it for a short time—before sending it to the effect input or output.

Most delays allow you to feed a percentage of the delayed signal back to the input, creating a repeating echo effect. Each subsequent repeat is a little quieter than the previous one.

The delay time can often be synchronized to the project tempo by matching the grid resolution of the project, usually in note values or milliseconds.

You can use delays for:

- Doubling individual sounds, making it sound like a group of instruments playing the same melody.
- Creating echo effects, placing the sound in a large "space."
- Enhancing the stereo position of tracks in a mix.

Delay effects are generally used as channel insert or bussed effects. They are rarely used on an overall mix (in an output channel), unless you're trying to achieve a special effect, such as an "other worldly" mix.

This chapter describes the delay effects included with Logic Studio:

- Delay Designer (see below).
- Echo (see "Echo" on page 46).
- Sample Delay (see "Sample Delay" on page 46).
- Stereo Delay (see "Stereo Delay" on page 47).
- Tape Delay (see "Tape Delay" on page 48).

Delay Designer

Delay Designer is a *multi-tap* delay. Each *tap* is an independent delay. Unlike simple delay effects that only offer one or two delays (or taps), Delay Designer offers you up to 26 individual taps. In other words, you can think of Delay Designer as 26 separate delay processors—in one effect unit.

Delay Designer provides control over the following aspects of each tap:

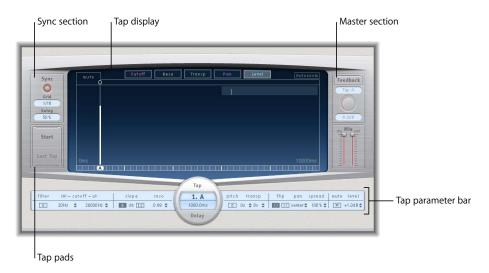
- Level and pan position
- Highpass and lowpass filters
- Pitch transposition (up or down)

Further effect-wide parameters include synchronization, quantization, feedback, and so on.

As the name implies, Delay Designer offers significant sound design potential. You can use it for everything from a basic echo effect, through to an audio pattern sequencer. You can create complex, evolving, moving rhythms by synchronizing the placement of taps—coupled with judicious use of pitch transposing and filtering. Alternately, you can set up numerous taps as "repeats" of other taps, much as you would use the feedback control of a simple delay—but with individual control over each repeat.

You can use Delay Designer on channels with mono, stereo, or surround inputs and/or outputs. See "Working With Delay Designer in Surround" for details on using it in surround channels.

The Delay Designer interface consists of five main sections:



- *Tap display:* This blue "view screen" display features a graphic representation of all taps. You can see, and edit, the parameters of each tap in this area. See "The Tap Display" section of this chapter for a more detailed look.
- Tap parameter bar: Offers a numeric overview of the current parameter settings for the selected tap. You can view and edit the parameters of each tap in this area. See "The Tap Parameter Bar" later in this chapter.
- *Sync section:* You can set all Delay Designer synchronization and quantization parameters in this section. See "Syncing Delay Taps" for more information.
- *Tap pads:* You can use these two pads to create taps in Delay Designer. See "Creating and Deleting Taps".
- Master section: This area contains the global Mix and Feedback parameters. See "The Master Section" for details.

The Tap Display

You can see—and interact with—taps in the Tap display. The display is divided into a number of sections:



- View buttons: Determine the parameter or parameters represented in the Tap display.
- Autozoom: When engaged, the main display is zoomed out, making all taps visible.
 Turn Autozoom off, if you want to zoom the display (by dragging vertically on the overview display) to view specific taps.
- Overview display: Shows all the taps in the time range.
- Toggle buttons: Click to toggle (switch) the parameters of a particular tap. The parameter being toggled is chosen with the view buttons. The label at the left of the toggle bar always indicates the parameter being toggled. See "Using the Toggle Buttons to Edit Tap Parameters" for more information.
- *Main display:* Offers a visual representation of each tap as a shaded line. Each tap contains a bright bar (or dot for stereo panning) that indicates the value of the parameter. You can directly edit tap parameters with the mouse in the main display area. See "Editing Taps" for more details.

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• *Identification bar:* Includes an identification letter for each tap, along with handles that allow you to move the selected tap backwards or forwards in time.

The View Buttons

The view buttons determine which parameter is represented in the main display.



- *Cutoff:* When clicked, the taps in the main display will show the highpass and lowpass filter cutoff frequencies.
- Reso: When clicked, the main display shows the filter resonance value of each tap.
- Transp: Click to show the pitch transposition of each tap in the main display area.
- Pan: Click to show the pan parameter of each tap in the main display.
 - For mono to stereo channels, each tap will contain a line showing its pan position.
 - For stereo to stereo channels, each tap will contain a dot showing its stereo balance. A line (extending outwards from the dot) indicates its stereo spread.
 - For surround channels, each tap will contain a line representing its surround angle (see "Working With Delay Designer in Surround" for details).
- Level: Click to show the relative volume level of each tap in the main display.

The Overview Display

You can use the Overview display to zoom and navigate the main display area:

To zoom the main display, do one of the following:

 Click-hold on the highlighted section (bright rectangle) of the overview display, and drag up or down.



 Click-hold on the highlighted bars—to the left or right of the bright rectangle—and drag to the left or right.



Note: The Autozoom button needs to be turned *off* for this to work. When you zoom in on a small group of taps, the overview display continues to show all taps. The area shown in the Tap display is indicated by the bright rectangle.

To move to different sections of the Tap display:

Click-hold the bright rectangle and drag to the left or right.

The zoomed view in the main display will update as you drag.

Creating and Deleting Taps

You can create new delay taps in three different ways: the identification bar, by using the Tap pads, or by copying existing taps.

To create taps in the identification bar:

Click at the desired position.



To create taps with the Tap pad:

1 Click the upper Start pad.

Note: Whenever you click the Start pad, it will automatically erase all existing taps. Given this behavior, once you have created your initial taps, you will want to create subsequent taps by clicking in the identification bar.

The upper pad label will change to Tap, and a red tap recording bar will appear in the strip below the view buttons.



2 Click the Tap button to record new taps on the fly.

New taps are created (at the exact moments in time) of each click, adopting the rhythm of your click pattern.

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3 To finish creating taps, click the Last Tap button.

This adds the final tap, ending tap recording, and assigning the last tap as the *feedback* tap (see "The Master Section" for an explanation of the feedback tap).

Note: If you do not click the Last Tap button, tap recording automatically stops after ten seconds, or when the 26th tap is created, whichever comes first.

To copy taps in the identification bar:

Option-drag a selection of one or more taps to the desired position.

The delay time of copied taps is set to the drag position.

Tap Creation Suggestions

The fastest way to create multiple taps is to use the Tap pads. If you have a specific rhythm in mind, you might find it easier to tap out your rhythm on dedicated hardware controller buttons, instead of using mouse clicks. If you have a MIDI controller, you can assign the Tap pads to buttons on your device. See the Control Surfaces Support manual for information on assigning controllers.

Whenever you click the Start Tap pad, it will automatically erase all existing taps. Given this behavior, once you have created your initial taps, you will want to create subsequent taps by clicking in the identification bar.

Once a tap has been created, you can freely adjust its position. See "Moving Taps" for details.

Identifying Taps

Taps are assigned letters—based on their order of creation. The first tap to be created is assigned as Tap A, the second tap is assigned as Tap B, and so on. Once assigned, each tap will always be identified by the same letter, even as taps are moved in time, and therefore re-ordered. As an example, if you initially create three taps, they will be named Tap A, Tap B, and Tap C. If you then change the delay time of Tap B, so that it precedes Tap A, it will still be called Tap A.

The identification bar shows the letter of each visible tap. The Tap Delay field of the Tap parameter bar displays the letter of the currently selected tap, or the letter of the tap being edited when multiple taps are selected (see "Selecting Taps" for details).

Deleting Taps

To delete a tap, simply select it and press the Delete or Backspace key. You can also select a tap in the identification bar and drag it down, below the Tap display.



These methods also work when more than one tap is selected.

Finally, you can right-click or Control-click *on any tap* in the Delay Designer interface, and choose the Delete All Taps command from the shortcut menu to delete all taps.

Selecting Taps

There will always be at least one selected tap. You can easily distinguish selected taps by color—the toggle bar icons, and identification bar letters of selected taps are white.



To select a tap, do one of the following:

- Click on a tap in the main display.
- Click on the desired tap letter in the identification bar.

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• Click the downward pointing arrow in the Tap field of the Tap parameter bar, and choose the desired tap letter from the menu.





You can select the next or previous tap by clicking the arrow buttons next to the left of the Tap name.

To select multiple taps, do one of the following:

- Click-drag across the background of the main display to rubber band select multiple taps.
- Shift-click specific taps in the Tap display to select multiple non-adjacent taps.

Moving Taps

You can move a tap backwards or forwards in time.

Note: When you move a tap, you are actually editing its delay time.

To move a tap in time:

 Select the tap in the identification bar, and drag it forward in time (left) or backward in time (right).

Note: Editing the Delay Time parameter in the Tap Delay field of the Tap parameter bar also moves a tap in time. See "The Tap Parameter Bar" and "Editing Taps" for more details on the Tap Delay field and editing taps.

The Tap Parameter Bar

The Tap parameter bar shows the current numeric values for every parameter of the selected tap. You can directly edit these parameters in the Tap parameter bar.



The parameters shown are:

- Filter On/Off button: Enables or disables the highpass and lowpass filters for the selected tap.
- HP Cutoff LP: You can view, and set, the cutoff frequencies (in Hz) for the highpass and lowpass filters here.
- Slope: Determines how steep the highpass and lowpass filters will be. Click the 6 dB button for a gentler filter slope, or the 12 dB button for a steeper, more pronounced filtering effect. You cannot set the slope of the highpass and lowpass filters independently.
- Reso: Sets the amount of filter resonance for both filters.
- *Tap Delay field:* This displays both the number and name (top), and delay time (bottom) of the selected tap.
- *Pitch On/Off button:* Enables or disables pitch transposition for the selected tap.
- *Transpose:* Use the first field to set the amount of pitch transposition in semitones, and the second field to fine tune each semitone step in cents (1/100th of a semitone).
- Flip: Swaps the left and right side of the stereo or surround image. In other words, clicking this button reverses the tap position from left to right, or vice versa. As an example, if a tap is set to 55% left, clicking the flip button will swap it to 55% right.
- Pan: The Pan parameter controls the pan position for mono input signals, stereo balance for stereo input signals, and surround angle when used in surround configurations. The pan parameter displays a percentage between 100% (full left) and -100% (full right), which represents the pan position or balance of the tap. A value of 0% represents the center panorama position. When used in surround, a surround panner replaces the percentage representation. See "Working With Delay Designer in Surround" for more information.
- *Spread:* When a stereo to stereo or stereo to surround instance of Delay Designer is used, this parameter allows you to set the width of the stereo spread for the selected tap.
- *Mute*: Clicking this button mutes or unmutes the selected tap.
- Level: Determines the output level for the selected tap.

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Editing Taps

You can edit taps both graphically, using the main Tap display, and numerically, using the Tap parameter bar. All tap edits are reflected both graphically and numerically.

Editing Taps in the Tap Parameter Bar

You can edit every parameter in the Tap parameter bar using standard click, or click-drag techniques.



To edit a parameter in the Tap parameter bar:

- Click on a button or up/down arrow to enable, disable, or alter a parameter value.
- Drag a parameter value up or down to change it.

If you have multiple taps selected in the Tap display, the values of all selected taps will be increased or decreased. These changes are relative to other taps.

Option-clicking on a parameter resets it to the default setting. If multiple taps are selected, Option-clicking a parameter of one tap will reset that parameter to its default value for all selected taps.

Editing Parameters in the Tap Display

You can graphically edit any tap parameter that is represented as a vertical line in the main Tap display.

To edit a tap parameter in the Tap display:

- 1 Click the view button of the parameter you want to edit.
- 2 Click-drag vertically on the bright line of the tap you wish to edit (or on one of the selected taps, if multiple taps are selected).



If you have multiple taps selected, the values of all selected taps will be increased or decreased relative to other taps.

You can also set the value of multiple taps by Command-dragging horizontally and vertically across several taps in the Tap display. As you do so, the parameter value changes to match the mouse position as you drag across the taps. Put another way, Command-dragging across several taps allows you to "draw" in values, much like drawing in controllers with the Pencil tool in the Hyper Editor.



You can also hold down the Command key, and click the Tap display before dragging. This will result in a line trailing behind the pointer. The values of the taps are aligned along the line when you click the Tap display again.



Option-clicking a tap resets the chosen parameter to its default setting. If multiple taps are selected, Option-clicking one tap will reset that parameter to its default value for all selected taps.

Editing the Filter Cutoff Parameters in the Tap Display

While the steps outlined above apply for most graphically editable parameters, the Cutoff and Pan parameters work in a slightly different fashion.



In the Cutoff view, each tap actually shows two parameters—highpass and lowpass filter cutoff frequency. The filter cutoff values can be adjusted independently by dragging the specific cutoff frequency line (upper line is lowpass, lower line is highpass), or both cutoff frequencies can be adjusted by dragging between them.

When the highpass filter cutoff frequency value is lower than that of the lowpass cutoff frequency, *only one line is shown*. This line represents the frequency band that passes through the filters (in other words, the filters act as a bandpass filter). In this configuration, the two filters operate *serially*, meaning the tap first passes through one filter, then the other.

If the highpass filter's cutoff frequency value is above that of the lowpass filter cutoff frequency, the filter switches from serial operation to *parallel* operation, meaning the tap passes through both filters simultaneously. In this case, the space between the two cutoff frequencies represents the frequency band being rejected (in other words, the filters act as a band-reject filter).

Editing the Pan Parameter in the Tap Display

The way that the Pan parameter is represented in the Pan view is entirely dependent on the input channel configuration of Delay Designer.



In mono input/stereo output configurations, all taps are initially panned to the center. To edit the pan position, click-drag (vertically) *from the center* of the tap—in the direction you wish to pan the tap or taps. A white line will extend from the center in the direction you have dragged, reflecting the pan position of the tap (or taps, in multiple selections). Lines above the center position indicate panning to the left, and lines below the center position, panning to the right.



With stereo input/stereo output configurations, the Pan parameter adjusts the stereo balance, not the position of the tap in the stereo field. The Pan parameter takes the form of a *stereo balance dot* on the tap representing its stereo balance. To adjust the balance, click-drag the stereo balance dot up or down the tap.

By default, the stereo spread is set to 100%. To adjust this, click drag on either side of the dot. As you do so, the width of the line (extending outwards from the dot) changes. Keep an eye on the spread parameter in the Tap parameter bar to view the spread percentage numerically.



In surround configurations, the bright line represents the surround angle. See "Working With Delay Designer in Surround" for more information.

Using the Toggle Buttons to Edit Tap Parameters

Each tap has its own toggle button in the Toggle bar. These buttons offer you a quick way to graphically activate and deactivate parameters. The specific parameter being toggled by the toggle buttons depends on the current View button selection:



- Cutoff view: Toggle buttons turn the filter on or off.
- Reso view: Toggle buttons switch filter slope between 6 dB and 12 dB.
- Pitch view: Toggle buttons switch pitch transposition on or off.
- Pan view: Toggle buttons switch between the Flip modes.
- Level view: Toggle buttons mute or unmute the tap.

Option-Command-clicking a toggle button switches the mute state, regardless of the current view. When you release the Option and Command keys, the toggle buttons return to their standard functionality (in the active View mode).

Note: The *first time* you edit a filter or pitch transpose parameter, the respective module will automatically turn on. This saves you the effort of manually turning on the filter or pitch transposition module before editing. Once you manually turn either of these modules off, however, you will need to manually switch it back on.

Editing Tap Parameters Using the Shortcut Menu

Right-clicking or Control-clicking a tap will open a shortcut menu that contains the following commands:

- Copy sound parameters: Copies all parameters—except the delay time—of the selected tap or taps into the Clipboard.
- Paste sound parameters: Pastes the tap parameters stored in the Clipboard into the selected tap or taps. If there are more taps in the Clipboard than are selected in the main Tap display, the extra taps in the Clipboard are ignored.
- Reset sound parameters to default values: Resets all parameters of all selected taps—except the delay time—to the default values.
- Delete all taps: Deletes all taps.

Parameter Editing Suggestions

In general, you'll find editing in the Tap parameter bar fast and precise when you want to edit the parameters of one tap at a time. All parameters of the selected tap are available, with no need to switch display views, or estimate values with vertical lines.

If you want to edit the parameters of one tap relative to other taps, use the Tap display. Also, if you want to edit multiple taps at once, you can use the Tap display to select multiple taps and then edit them together.

Don't forget Command-dragging to draw in different values for multiple taps.

Syncing Delay Taps

Delay Designer can either synchronize to the project tempo, or run independently. When Delay Designer is in synchronized mode (Sync mode), taps snap to a grid of musically relevant positions—based on note durations. You can also set a Swing value, when in Sync mode; this varies the precise timing of the grid, resulting in a more laid back, less robotic feel for each tap. When not in Sync mode, taps don't snap to any grid, nor can you apply the Swing value.

Activating Sync Mode

Sync mode is turned on or off by clicking the Sync button in the Sync section.



An orange ring is shown around the Sync button when Sync mode is on, and a grid that matches the chosen Grid parameter value is shown in the identification bar.

Once Sync mode is activated, all taps will move towards the closest delay time value on the grid. When you subsequently create or move taps, they will always move in increments based on the current grid setting, or will be created at a "snapped" position on the grid.

Setting the Grid Resolution

The Grid menu offers several grid resolutions, which correspond to musical note durations. The grid resolution, along with the project tempo, determines the length of each grid increment.

To set the grid resolution:

• Click the Grid field, and choose the desired grid resolution from the pop-up menu.

As you change grid resolutions, you will notice that the increments shown in the identification bar will change accordingly. This also determines a step limitation for all taps.

As an example: The current project tempo is set to 120 beats per minute, and the Delay Designer Grid parameter is set to 1/16th notes. At this tempo and grid resolution, each grid increment is 125 milliseconds apart. If Tap A is currently set to 380 ms, turning on Sync mode would immediately shift Tap A to 375 ms. If you subsequently moved Tap A forward in time, it would snap to 500 ms, 625 ms, 750 ms, and so on.

At a resolution of 1/8th notes, the steps are 250 milliseconds apart, so Tap A would automatically snap to the nearest division (500 ms), and could be moved to 750 ms, 1000 ms, 1250 ms, and so on.

Setting the Swing Value

The Swing value determines how close to the absolute grid position every second grid increment will be. A Swing setting of 50% means that every grid increment has the same value. Settings below 50% result in every second increment being shorter in time. Settings above 50% result in every second grid increment being longer in time.

To adjust the Swing value:

Click-drag up or down in the Swing field to raise or lower the Swing value.

By subtly varying the grid position of every second increment (values between 45 and 55%), the swing function creates a less rigid rhythmic feel. This can be a very "humanizing" effect, but you are not limited to using the swing function in this way.

Extremely high Swing settings are not subtle at all, as they place every second increment directly beside the subsequent increment. You can use this facility to create interesting and intricate double rhythms with some taps, while retaining the grid to lock other taps into a more rigid synchronization with the project tempo.

Saving Sync Settings

When you save a Delay Designer setting, the Sync mode status, Grid, and Swing values are all saved. When you save a setting with Sync mode on, the grid position of each tap is also stored. This ensures that a setting loaded into a project with a different tempo (to that of the project that the setting was created in), will retain the relative positions, and rhythm, of all taps—at the new tempo.

One point to bear in mind, however, is that Delay Designer offers a maximum delay time of 10 seconds. This means that if you load a setting into a project with a slower tempo than the tempo at which it was created, some taps may fall outside the 10 second limit. In such cases, these taps will not be played, but will still be retained as part of the setting.

The Master Section

The Master section incorporates parameters for two global functions: delay feedback and dry/wet mix.



Using Feedback

In simple delays, the only way for the delay to repeat is to use feedback. As Delay Designer offers 26 taps, you can use these to create repeats, rather than requiring discreet feedback controls for each tap.

Delay Designer's global feedback parameter allows you to send the output of *one* tap back through the effect input, to create a self-sustaining rhythm or pattern.

This tap is called the *feedback tap*.

To toggle feedback on or off:

Click the Feedback button.

When the Feedback button is turned on, it is lit. The orange track around the Feedback Level knob indicates the current feedback level.

Note: If feedback is turned on and you begin creating taps using the Tap pads, feedback is automatically switched off. When you stop creating taps with the Tap pads by clicking the Last Tap button, feedback will automatically be turned back on.

To determine the feedback tap:

Click the Feedback Tap field, and choose the desired tap from the pop-up menu.

You can vary the output level of the feedback tap back into Delay Designer's input between 0% (no feedback) or 100% (the feedback tap is fed back at full volume).

To set the feedback level of the feedback tap, do one of the following:

- Click-drag the Feedback Level knob.
- Click-drag on the Feedback Level field.

The Mix Sliders

Use the Mix sliders to adjust the level of the dry input signal and the (post-processing) wet signal.

Working With Delay Designer in Surround

Delay Designer is optimally designed for use in surround configurations. With 26 taps, you can fly delay taps all over the surround field for some truly amazing rhythmic effects!

When instantiating Delay Designer in any surround configuration, the pan percentage on the Tap parameter bar is replaced with a surround panner, allowing you to determine the surround position of each tap.



The movement of the surround position is made easier with these functions:

- Hold Command to lock diversity.
- Hold Command-Option to lock the angle.
- Option-click the blue dot to reset angle and diversity.

In the Tap display's Pan view, you will only be able to adjust the *angle* of the tap between 0 and 360 degrees, not its diversity.

Delay Designer always processes each input channel independently.

- In a mono/stereo input, surround output configuration, Delay Designer processes the two stereo channels independently, and the surround panner lets you place each delay around the surround field.
- In surround input, surround output configurations, Delay Designer processes each surround channel independently, and the surround panner lets you adjust the surround balance of each tap in the surround field.

Note: The Delay Designer generates separate automation data for stereo pan and surround pan operations. This means that when using the Delay Designer in surround channels, it will not react to existing stereo pan automation data, and vice versa.

Echo

This simple echo effect always synchronizes the delay time to the project tempo, allowing you to quickly create echo effects that run in time with your composition.

Echo Parameters

- *Time*: Sets the grid resolution of the delay time in musical note durations—based on the project tempo. "T" values represent triplets, "." values represent dotted notes.
- Repeat: Determines how often the delay effect is repeated.
- Color: Sets the harmonic content (color) of the delay signal.
- Wet and Dry: These individually control the amount of original and effect signal.

Sample Delay

The Sample Delay is not so much an effect as a tool: You can use it to delay a channel by single sample values. When used in conjunction with the phase inversion capabilities of the Gain effect, the Sample Delay is well-suited to the correction of timing problems that may occur with multi-channel microphones. It can also be used creatively, to emulate stereo microphone channel separation.

The stereo version of the plug-in provides separate controls for each channel, and also offers a Link L & R option that moves both channels by the same number of samples.

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.

Stereo Delay

The Stereo Delay works much like the Tape Delay (see below), but allows you to set the Delay, Feedback, and Mix parameters separately for the left and right channel.

The effect also 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.

You can freely use the Stereo Delay on mono tracks or busses, when you want to create independent delays for the two stereo sides.

Note: If you do use the effect on mono channel strips, the track or bus will have two channels from the point of insertion (all Insert slots after the chosen slot will be stereo).

This section only covers the additional features offered by the Stereo Delay. For more information about the parameters shared with the Tape Delay, see the Tape Delay section below.



- Left Input and Right Input: Use these to choose the input signal for the two stereo sides. Options include Off, Left, Right, L+R, L-R.
- Feedback Phase button: Use to invert the phase of the corresponding channel's feedback signal.
- Crossfeed Left to Right and Crossfeed Right to Left: Use to transfer the feedback signal of the left channel to the right channel, and vice versa.
- Crossfeed Phase buttons: Use to invert the phase of the crossfed feedback signals.

Tape Delay

The Tape Delay simulates the warm sound of vintage tape echo machines, with the convenience of easy delay time synchronization to your project tempo.

The Tape Delay is equipped with a highpass and lowpass filter in the feedback loop, making it easy to create authentic dub echo effects, and also includes an LFO for delay time modulation. The LFO produces a triangular wave, with adjustable speed and modulation intensity. You can use it to produce pleasant or unusual chorus effects, even on long delays.



- Feedback: Determines the amount of delayed and filtered signal that is routed back to the input of the Tape Delay.
- Freeze: Captures the current delay repeats and sustains them until the Freeze parameter is released.
- *Delay:* Sets the current delay time in milliseconds (this parameter is dimmed when you synchronize the delay time to the project tempo).
- *Tempo:* Sets the current delay time in beats per minute (this parameter is dimmed when you synchronize the delay time to the project tempo).
- *Sync button:* Switch this on to synchronize delay repeats to the project tempo (including tempo changes).
- Note buttons: Click to set the grid resolution for the delay time, in note durations.
- *Groove slider:* Determines the proximity of every second delay repeat to the absolute grid position (how close every second delay repeat is, in other words).
- *Distortion Level (Extended Parameter):* Determines the level of the distorted (tape saturation) signal.
- Low Cut and High Cut: Frequencies below the Low Cut value, and above the High Cut value are filtered out of the source signal.
- LFO Speed: Sets the frequency (speed) of the LFO.
- LFO Depth: Sets the amount of LFO modulation. A value of 0 turns delay modulation off.
- Flutter parameters: Simulates the speed irregularities of the tape transports used in analog tape delay units. Flutter Rate adjusts the speed, and Flutter Intensity determines how pronounced the effect is.

- Smooth: Evens out the LFO and flutter effect.
- Dry and Wet: These individually control the amount of original and effect signal.

Setting the Feedback

When you set the Feedback slider 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.

Note: The levels of the original signal and its taps (echo repeats) tend to accumulate, 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 continue to sound good.

Setting the Groove Value

The Groove value determines the proximity (how close) of every second delay repeat to the absolute grid position. A Groove setting of 50% means that every delay will have the same delay time. Settings below 50% result in every second delay being played earlier in time. Settings above 50% result in every second delay being played later in time. 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.

Filtering the Delay Effect

You can shape the sound of the echoes, using the on-board highpass and lowpass filters. The filters are located in the feedback circuit, meaning that the filtering effect increases in intensity with each delay repeat. If you're after 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.

Note: If you're unable to hear the effect, even though you seem to have a suitable configuration, be sure to check out both the Dry/Wet controls *and* the filter settings: Move the High Cut filter slider to the far right, and the Low Cut filter slider to the far left.

Distortion 3

You can use Distortion effects to recreate the sound of analog or digital distortion, and to radically transform your audio.

Distortion effects simulate the distortion created by vacuum tubes, transistors, or digital circuits. Vacuum tubes were used in audio amplifiers before the development of digital audio technology, and are still used in musical instrument amps today. When overdriven, they produce a type of distortion which many people find musically pleasing, and which has become a familiar part of the sound of rock and pop music. Analog tube distortion adds a distinctive warmth and bite to the signal.

There are also distortion effects which intentionally cause clipping and digital distortion of the signal. These can be used to modify vocal, music, and other tracks to produce an intense, unnatural effect, or for creating sound effects.

Distortion effects include parameters for *tone*, which let you shape the way in which the distortion alters the signal (often as a frequency-based filter), and for *gain*, which let you control how much the distortion alters the output level of the signal.

Warning: When set to high output levels, distortion effects can damage your hearing (and speakers). When adjusting effect settings, it is recommended that you lower the output level of the track, and raise the level gradually when you are finished.

The following sections describe the individual effects included with Logic Studio.

- "Bitcrusher" on page 52
- "Clip Distortion" on page 53
- "Distortion" on page 54
- "Distortion II" on page 55
- "Overdrive" on page 56
- "Phase Distortion" on page 57

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Bitcrusher

The Bitcrusher is a low resolution digital distortion effect. You can use it to emulate the sound of early digital audio, create artificial aliasing by dividing the sample rate, or distort signals until they are unrecognizable.

Bitcrusher Parameters



- Drive slider and field: Sets the amount of gain (in decibels) applied to the input signal.
- Resolution slider and field: Sets the bit rate (between 1 and 24 bits).
- Downsampling slider and field: Sets the amount by which the sample rate is reduced. A value of 1x leaves the signal unchanged, a value of 2x halves the sample rate, and a value of 10x reduces the sample rate to one-tenth of the original signal. (For example, if you set Downsampling to 10x, a 44.1 kHz signal is sampled at just 4.41 kHz.)
- *Mode buttons*: Click one of the buttons to set the distortion mode to Folded, Cut, or Displaced (each of which is described in the following section).
- *Clip Level slider and field*: Sets the point below the normal threshold at which the signal starts clipping.
- *Mix slider and field (extended parameter):* Determines the balance of dry and wet signals.

Using the Bitcrusher

Setting the Resolution parameter to a value lower than the bit rate of the original signal degrades the signal, introducing digital distortion. Lowering the value increases the number of sampling errors, generating more distortion. At extremely low bit rates, the amount of distortion can be greater than the level of the usable signal.

The Mode buttons determine whether signal peaks that exceed the clip level are Folded, Cut, or Displaced (as displayed on the button icons and the resulting waveform in the display). The kind of clipping that occurs in 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.

Raising the Drive level tends to increase the amount of clipping at the output of the Bitcrusher as well.

Clip Distortion

Clip Distortion is a nonlinear distortion effect that produces unpredictable spectra. You can use it to simulate warm, overdriven tube sounds, and also to create drastic distortion.

Clip Distortion features an unusual combination of serially connected filters. After being amplified by the Drive value, the signal passes through a highpass filter, and is then subjected to nonlinear distortion, as controlled by the Symmetry parameter. After the distortion, the signal passes through a lowpass filter. The effected signal is mixed with the original signal, after which the mixed signal is sent through another lowpass filter. All three filters have a slope of 6 dB/octave.

This unique combination of filters allows for gaps in the frequency spectra that can sound quite good with this sort of nonlinear distortion. The clip circuit graphic visually depicts every parameter except for the High Shelving filter parameters.

Clip Distortion Parameters



- *Drive slider and field:* Sets the amount of gain applied to the input signal. After being amplified by the *Drive* value, the signal passes through a highpass filter.
- Tone slider and field: Sets the cutoff frequency (in Hertz) of the highpass filter.
- Symmetry slider and field: Sets the amount of nonlinear (asymmetrical) distortion applied to the signal.
- *Clip Filter slider and field:* Sets the cutoff frequency (in Hertz) of the first lowpass filter through which the signal passes after distortion.
- *Mix slider:* Sets the ratio of the effected (wet) signal to the uneffected (dry) signal following the Clip Filter.
- Sum LPF circular slider and field: Sets the cutoff frequency (in Hertz) of the lowpass filter through which the mixed signal passes.
- *High Shelving Frequency knob and field:* Sets the frequency (in Hertz) of the high shelving filter.
- *High Shelving Gain knob and field:* Sets the amount of gain applied to the output signal.

- Input Gain field and slider (extended parameter): Sets the amount of gain applied to the input signal.
- Output Gain field and slider (extended parameter): Sets the amount of gain applied to the output signal.

Using Clip Distortion

If you set the High Shelving Frequency to around 12 kHz, you can use it like the treble control on a mixer channel strip or a stereo hi-fi amplifier. Unlike those types of treble controls, however, you can boost or cut the signal by up to ± 30 dB using the Gain parameter.

Distortion

This Distortion effect simulates the lo-fi, dirty distortion generated by a bipolar transistor. You can use it to simulate playing a musical instrument through a highly overdriven amplifier, or to create unique distorted sounds.

Distortion Parameters



- Drive slider and field: Sets the amount of saturation applied to the signal.
- Tone slider and field: Sets the frequency at which the signal is filtered by a high cut filter. Filtering the harmonically-rich distorted signal produces a somewhat less grating, softer tone.
- Output slider and field: Sets the output volume level. This allows you to compensate for increases in loudness caused by adding distortion.

Distortion II

Distortion II emulates the distortion effect section of a Hammond B3 organ. You can use it on musical instruments to recreate this classic effect, or use it creatively when designing new sounds.

Distortion II Parameters



- PreGain dial: Sets the amount of gain applied to the input signal.
- Drive dial: Sets the amount of saturation applied to the signal.
- *Tone dial:* Sets the frequency at which the signal is filtered. Filtering the harmonically-rich distorted signal produces a somewhat less grating, softer tone.
- *Type pop-up menu:* Choose the type of distortion you want to apply. The choices are: Growl, Bity, and Nasty.
 - *Growl*: Emulates a two-stage tube amplifier, similar to the type found in a Leslie 122 model, often used together with a Hammond B3 organ.
 - Bity: Emulates the sound of a bluesy (overdriven) guitar amp.
 - Nasty: Produces hard distortion, suitable for creating very aggressive sounds.

Overdrive

The Overdrive effect emulates the distortion produced by a field effect transistor (FET), which is commonly used in solid-state musical instrument amplifiers and hardware effects devices. When saturated, FETs generate a warmer sounding distortion than bipolar transistors.

Overdrive Parameters



- Drive slider and field: Sets the amount of saturation of the transistor.
- *Tone slider and field:* Sets the cutoff frequency at which the signal is filtered. Filtering the harmonically-rich distorted signal produces a somewhat less grating, softer tone.
- Output slider and field: Sets the output volume level. Using the Overdrive plug-in tends to increase the level of the original signal, and you can compensate for this by lowering the Output level.

Phase Distortion

The Phase Distortion effect is based on a modulated delay line, similar to a chorus or flanger effect (for more information about these effects, see Chapter 9, "Modulation," on page 135). Unlike these effects, however, in the Phase Distortion effect the delay time is not modulated by a low frequency oscillator (LFO), but rather by a lowpass-filtered version of the input signal itself. This means that the signal modulates its own phase position. The input signal only passes the delay line and is not affected by any other process.

Phase Distortion Parameters



- *Monitor button:* Turn on to hear only the input signal, or turn off to hear the mixed signal.
- Cutoff circular slider and field: Sets the cutoff frequency of the resonant lowpass filter through which the input signal passes.
- Resonance circular slider and field: Sets the resonance of the resonant lowpass filter through which the input signal passes.
- *Mix slider and field:* Adjusts the percentage of the effected signal mixed with the original signal.
- Max Modulation slider and field: Sets the maximum delay time.
- Intensity slider and field: Sets the amount of modulation applied to the signal.
- Phase Reverse pop-up menu (extended parameter): Choose On to have positive input values reduce the delay time on the right channel. Only available for stereo instances of the Phase Distortion effect.

Using the Phase Distortion

The input signal only passes the delay line and is not affected by any other process. The Mix parameter 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, with dedicated Cutoff frequency and Resonance controls. You can listen to the filtered side chain (instead of the Mix signal) by turning on the Monitor button. You set the maximum delay time via the Max Modulation parameter. The amount of modulation itself is controlled with Intensity.

Below the other parameters is the Phase Reverse parameter. Normally, a positive input value results in a longer delay time. By turning on the Phase Reverse parameter, positive input values result in a reduction of the delay time on the right channel only. This is only available for stereo instances of the effect.

Dynamics 4

You can use Dynamics effects to control the perceived loudness of your audio, add focus and punch to tracks and projects, and optimize the sound for playback in different situations.

The *dynamic range* of an audio signal is the range between the softest and loudest parts of the signal (technically, between the lowest and the highest amplitude). Using dynamics effects, you can adjust the dynamic range of individual audio files, tracks, or an overall project to increase the perceived loudness, and highlight the most important sounds while making sure softer sounds are not lost in the mix. Dynamics effects include compressors, limiters, and noise gates.

Compressors

A compressor works like an automatic volume control, lowering the volume whenever it rises above a certain level, called the *threshold*. Why would you want to reduce the dynamic level? By cutting the highest parts of the signal (called *peaks*), the compressor lets you raise the overall level of the signal, increasing the perceived volume. This gives the sound more focus by making the louder foreground parts stand out while keeping the softer background parts from becoming inaudible. Compression also tends to make sounds tighter or punchier because transients are emphasized (depending on attack and release settings) and because the maximum volume is reached more swiftly.

In addition, compression can help make a project sound better when played back in different audio environments. For example, the speakers on a television set or in a car sound system typically have a narrower dynamic range than the sound system in a theater. Compressing the overall mix can help make the sound fuller and clearer in lower-fidelity playback situations.

Compressors are typically used on vocal tracks to make the vocals prominent in an overall mix. They can also be used on music and sound effects tracks, but rarely on ambience tracks.

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Some compressors, called *multiband compressors*, can divide the incoming signal into different frequency bands, and apply different compression settings to each band. This helps achieve the maximum level without introducing compression artifacts, and is typically used on an overall project mix.

Expanders

Expanders are similar to compressors, except that they raise, rather than lower, the signal when it exceeds the threshold. Expanders are used to enliven the audio signal.

Limiters

Limiters (also called *peak limiters*) work in a similar way as compressors, in that they reduce the audio signal when it exceeds a set threshold. The difference is that while a compressor gradually lowers the signal level above the threshold, a limiter quickly reduces any signal louder than the threshold to the threshold level. The main use of a limiter is to prevent clipping while preserving the maximum overall signal level.

Noise Gates

Noise gates alter the signal in the opposite way that compressors or limiters do. While a compressor lowers the level when the signal passes above the threshold, a noise gate lowers the signal wherever it is below the threshold. Louder sounds pass through unchanged, but softer sounds, such as ambient noise or the decay of a sustained instrument, are cut off. Noise gates can be used mainly to eliminate low-level noise or hum from an audio signal.

The following sections describe the effects included with Logic Studio.

- "Adaptive Limiter" on page 61
- "Compressor" on page 62
- "DeEsser" on page 65
- "Ducker" on page 67
- "Enveloper" on page 68
- "Expander" on page 70
- "Limiter" on page 71
- "Multipressor" on page 72
- "Noise Gate" on page 76
- "Silver Compressor" on page 78
- "Silver Gate" on page 79
- "Surround Compressor" on page 79

Adaptive Limiter

The Adaptive Limiter is a versatile tool for controlling the perceived loudness of sounds. It works by rounding and smoothing peaks in the signal, producing an effect similar to an analog amplifier being driven hard. Like an amplifier, it can slightly color the sound of the signal. You can use the Adaptive Limiter to achieve maximum gain without clipping (exceeding 0 dBFS).

The Adaptive Limiter is typically used in the final mix, where it may be placed after a compressor (such as the Multipressor) and before a final gain control in order to produce a mix of maximum loudness. Using the Adaptive Limiter can produce a louder-sounding mix than can be achieved by simply normalizing the signal.

Note: Using the Adaptive Limiter adds latency when the Lookahead parameter is active. In most situations, it should be used for mixing and mastering previously recorded tracks, not when recording.





- *Input Scale knob:* Scales the input level. Scaling is useful with very high or low input signals, to bring the level into the most effective range for the Gain knob to work effectively. In general, it should never exceed 0 dBFS.
- Gain knob: Sets the amount of gain after input scaling.
- Out Ceiling knob: Sets the maximum output level, or ceiling, above which the signal will not rise.

Input meters (to the left of the control dials) show the input levels in real time as the file or project plays. The Output meters show the output levels, allowing you to see the results of the Adaptive Limiter. The two Margin fields show the highest level for input and output respectively (since the start of playback). You can reset the Margin fields by clicking them.

Extended Parameters

- *Mode pop-up menu:* Choose which type of peak smoothing the Adaptive Limiter uses from the menu. The choices are *OptFit*, in which limiting follows a linear curve, which allows signal peaks exceeding 0 dB, and *NoOver*, which avoids distortion artifacts from the output hardware by ensuring that the signal does not exceed 0 dB.
- Lookahead field and slider: Adjusts how far ahead the Adaptive Limiter analyzes the file for peaks.
- Remove DC pop-up menu: Choose On to active a highpass filter that removes DC (direct current) from the signal. DC can be introduced by lower-quality audio hardware.

Compressor

The Compressor is designed to emulate the sound and response of a professional-level analog (hardware) compressor. It tightens up your audio by reducing sounds that exceed a certain threshold level, smoothing out the dynamics and increasing the overall volume—the perceived loudness. Compression helps bring the key parts of a track or a mix into focus while preventing softer parts from being inaudible. It is probably the most versatile and widely used sound-shaping tool used in mixing, next to EQ.

You can use the Compressor with individual tracks, including vocal, instrumental, and effects tracks, as well as on the overall mix. In most cases, you'll want to insert the Compressor directly into a channel.

Compressor Parameters



- *Circuit Type slider and field:* Choose the type of circuit emulated by the Compressor. The choices are Platinum, Classic A_R, Classic A_U, VCA, FET, and Opto (optical).
- Gain Reduction display: Shows the amount of compression applied as the audio plays.
- Attack knob and field: Sets the attack time (the amount of time it takes for the compressor to react when the signal exceeds the threshold).

- Release knob and field: Sets the release time (the amount of time it takes for the compressor to stop reducing the signal once the signal falls below the threshold).
- Auto button: When selected, the release time dynamically adjusts to the audio material.
- Compression curve display: Shows the compression curve created by the Ratio and Knee parameters, with input as the X-axis and output as the Y-axis.
- Ratio slider and field: Sets the compression ratio (the ratio by which the signal is reduced when it exceeds the threshold).
- *Knee slider and field:* Adjusts whether the signal is compressed immediately or more gradually at levels close to the threshold.
- Compression Threshold slider and field: Sets the threshold for the Compressor (the level above which the signal is reduced).
- *Peak/RMS buttons:* Turn on one or the other to set whether the Compressor analyzes the signal using Peak or RMS method when using the Platinum Circuit Type.
- Gain slider and field: Sets the amount of gain applied to the output signal.
- Gain pop-up menu: Choose a value to raise the output level in order to compensate for volume reduction caused by compression. The choices are OFF, 0 dB, and -12 dB.
- Limiter Threshold slider and field: Sets the threshold level for the limiter.
- Limiter button: Turns the integrated limiter on or off.

Extended Parameters

- Output Distortion pop-up menu: Choose whether to apply clipping above 0 dB, and what type of clipping is applied. The values are off, soft, hard, and clip.
- Mix slider and field: Determines the balance of dry and wet signals.

Side Chain Filter

- Activity pop-up menu: Choose whether the Compressor side chain is turned on or off, or in listen (audition) mode. The choices are off, listen, and on.
- *Mode pop-up menu*: Choose the type of filter used for the side chain. The choices are BP (bandpass), LP (lowpass), HP (highpass), ParEQ (parametric), and HS (high shelving).
- Frequency slider and field: Sets the frequency for the side chain filter.
- *Q slider and field:* Sets the bandwidth of the frequency band affected by the side chain filter.
- Gain slider and field: Sets the amount of gain applied to the side chain signal.

Using the Compressor

The following sections provide information on using each of the main Compressor parameters.

Threshold and Ratio

The most important Compressor parameters are Threshold and Ratio. The *Threshold* is the level (in decibels) above which the signal is reduced by the amount set as the Ratio. Because the *Ratio* is a percentage of the overall level, the more the signal exceeds the threshold, the more it is reduced. For example, with the Threshold set at -6 dB and the Ratio set to 4:1, a -2 dB peak in the signal (4 dB louder than the threshold) is reduced by 3 dB so that it is just 1 dB above the threshold, while a +6 dB peak (12 dB above the threshold) is reduced by 9 dB so that it is 3 dB above the threshold. The scale of dynamics is preserved, but the differences between the peaks are evened out.

Attack and Release

After Threshold and Ratio, the most important parameters are Attack and Release. You use the Attack and Release parameters to shape the dynamic response of the Compressor. The Attack parameter sets the amount of time after the audio exceeds the threshold before the Compressor starts reducing the signal. For many sounds, including voices and musical instruments, the initial attack is important for defining the sound, and setting the Attack higher ensures that the original attack is not altered. To maximize the level of an overall mix, setting the Attack lower ensures that the Compressor starts reducing the signal right away.

Similarly, the Release parameters controls how quickly the Compressor stops reducing the signal once it falls below the threshold. Setting the Release higher makes the difference in dynamics smoother, while setting it lower can make the difference more abrupt. Adjusting the Attack and Release parameters properly can help avoid pumping, a common side effect of compression.

Knee

The Knee parameter smooths the effect of the Compressor by controlling whether the signal is slightly compressed as it approaches the threshold. Setting the Knee close to 0 (zero) means that levels just below the threshold are not compressed at all (1:1 ratio), while levels at the threshold are compressed by the full Ratio amount (4:1, 10:1, or more). This is what audio engineers call hard knee compression, which can cause the transition to be abrupt as the signal reaches the threshold. Increasing the value of the Knee parameter applies some compression to the signal as it approaches the threshold, creating a smoother transition. This is called soft knee compression. Setting the Knee parameter controls the shape of compression around the threshold, while the Threshold and Ratio parameters control its intensity.

Other Parameters

Because the Compressor works by reducing levels, the overall volume of its output is typically lower than the input signal. You can adjust the output level using the Gain slider.

You can use the Auto Gain parameter to compensate for the reduction in gain produced by compression, referenced to either -12 dB or 0 dB. Auto Gain sets the level of gain (amplification) to a value of T-(T/R), where T= the Threshold and R= the Ratio.

The Gain Reduction Meter displays the amount of compression occurring as the signal plays. It's useful to watch how much your tracks are being compressed, and to make sure they're not being overly compressed.

When using the Platinum Circuit Type, the Compressor can analyze the signal using one of two methods: Peak or RMS (root mean square). While Peak is more technically accurate, RMS provides a better indication of how people perceive the signal's loudness. When using the Compressor primarily as a limiter, select the Peak button. When compressing individual tracks, especially music tracks, select the RMS button.

If you activate Auto Gain and RMS simultaneously, the signal may be saturated. If you hear any distortion, switch Auto Gain off and adjust the Gain slider until the distortion is gone.

DeEsser

The DeEsser is a frequency-specific compressor, designed to compress only a particular frequency band within a complex audio signal. It is used to eliminate hiss (also called sibilance) from the signal. The advantage of using the DeEsser instead of an EQ effect to cut high frequencies is that it compresses the signal dynamically rather than statically. This prevents the sound from becoming darker when no sibilance is present in the signal. The DeEsser features extremely fast attack and release times.

When using the DeEsser, you can set the frequency range being compressed (the Suppressor frequency) independently of the frequency range being analyzed (the Detector frequency). The two ranges appear separately in the DeEsser window for easy comparison. The DeEsser performs gain reduction on the Suppressor frequency range for as long as the threshold for the Detector frequency is exceeded.

The DeEsser does not use a frequency dividing network (a crossover utilizing low and highpass filters). Rather, it is based on subtracting the isolated frequency band, and so does not alter the phase curve.

DeEsser Parameters



The Detector parameters are on the left side of the DeEsser window, and the Suppressor parameters are on the right. The center section includes the Detector and Suppressor displays and the Smoothing slider.

Detector Section

- Detector Frequency knob: Sets the frequency range the DeEsser analyzes.
- *Detector Sensitivity knob:* Sets the degree of responsiveness to the input signal. At higher ratios, the Detector reacts more responsively.
- Monitor pop-up menu: Choose whether to monitor the filtered Detector signal (Det.), the filtered Suppressor signal (Sup.), or the sound removed from the input signal in response to the Sensitivity parameter (Sens.). Choose Off to hear the DeEsser output.

Suppressor Section

- Suppressor Frequency knob: Sets the frequency band that is reduced when the Detector frequency sensitivity threshold is exceeded.
- Strength knob: Sets the amount of gain reduction around the Suppressor frequency.

Center Section

- Detector and Suppressor frequency displays: The upper display shows the Detector frequency range, and the lower display shows the Suppressor frequency range (in Hz).
- Smoothing slider: Sets the reaction speed of the gain reduction start and end phases.
 Smoothing controls both the attack and release time (as they are used by compressors).

Ducker

Ducking is a common technique used in radio and television broadcasting: when the DJ/announcer speaks while music is playing, the music level is automatically reduced. When the announcement has finished, the music is automatically raised to its original volume level.

The Ducker plug-in provides a simple means of performing this process. It can even reduce the music level before the speaker starts (but this introduces a small amount of latency).

Ducker Parameters



- *Intensity:* Defines the amount of volume reduction (of the music mix track—this, in effect, is the output signal).
- Threshold: Determines the lowest level that a side chain signal must attain before it begins to reduce the (music mix) output level by the amount set with the Intensity slider. If the side chain signal level doesn't reach the threshold, the (music mix) track volume is not affected.
- Attack: Controls how quickly the volume is reduced. If you want the (music mix) signal to be gently faded out, set this slider to a high value. This value also controls whether or not the volume is reduced before the threshold is reached—the earlier this occurs, the more latency is introduced. It should be noted that this only works if the ducking signal is not live (in other words, the ducking signal must be an existing recording): Logic Pro needs to analyze the signal level before it is played back, to anticipate the point where ducking begins.
- Hold: Determines the duration that the (music mix) track volume is reduced for. This control avoids a chattering effect that can be caused by a rapidly changing sidechain level. If the sidechain level hovers around the threshold value, rather than clearly exceeding or falling short of it, set the Hold parameter to a high value to compensate for rapid volume reductions.

• *Release:* Controls how quickly the volume returns to the original level. Set to a high value if you want the music mix to slowly fade up after the announcement.

Using the Ducker

For technical reasons, the Ducker plug-in can only be inserted in output and bus channels.

To use the Ducker plug-in:

- 1 Insert the Ducker plug-in into an audio or bus channel strip.
- 2 Assign all track outputs that are supposed to "duck" (dynamically lower the volume of the mix) to a bus.
- 3 Select the buss that carries the ducking (vocal) signal in the Side Chain menu of the Ducker plug-in.

Note: Unlike all other side chain capable plug-ins, the Ducker side chain is mixed with the output signal after passing through the plug-in. This ensures that the ducking side chain signal (the voice over) is heard at the output.

4 Adjust the Ducker's parameters.

Enveloper

The Enveloper is an unusual effect that lets you shape transients—the attack and release phases of a signal. This gives it a unique capability to shape the signal, and can be used to achieve impressive results different than any other dynamics effect.

Enveloper Parameters



The Gain and Time controls on the left apply to the attack portion of the signal, while the Gain and Time controls on the right apply to the release portion.

- Threshold slider and field: Sets the threshold above which the attack and release levels are altered.
- (Attack) Gain slider and field: Sets the gain on the attack phase of the signal. When set to the center (0) position, the signal is unaffected.

- (Attack) Time knob: Sets the duration from the beginning of the signal considered as the attack.
- Display area: Graphically displays the attack and release curves applied to the signal.
- (Release) Time knob: Sets the duration of the signal considered as the release.
- (Release) Gain slider: Sets the gain on the release phase of the signal. When set to the center (0) position, the signal is unaffected.
- Out Level slider: Sets the level of the output signal.
- Lookahead slider and field: Adjusts how far the Enveloper looks ahead in the signal.

Using the Enveloper

The most important parameters of the Enveloper are the two Gain sliders, one on each side of the central display area, that govern Attack (left) and Release (right). Raising the gain emphasizes the attack or release phase, respectively, while lowering the gain attenuates the corresponding phase.

For example, boosting the attack gives a drum sound more snap, or amplifies the initial pluck (or pick) sound of a stringed instrument. Cutting the attack causes percussive signals to fade in more softly. You can also mute the attack, making it virtually inaudible. Another handy application for this effect is to mask the poor timing of accompanying instruments.

Emphasizing the release also boosts any reverb applied to the affected track. Conversely, toning down the release phase makes tracks originally drenched in reverb sound drier. This is particularly useful when working with drum loops, but it has many other applications as well. Let your imagination be your guide.

When using the Enveloper, set the Threshold to the minimum value and leave it there. Only when you seriously raise the release phase, which boosts the noise level of the original recording, should you raise the Threshold slider a little. This limits the Enveloper to affecting only the useful part of the signal.

Drastic boosting or cutting of either the release or attack phase may change the overall level of the signal. You can compensate for this by lowering the Out Level slider.

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

The Lookahead slider allows you to define how far ahead in the signal the Enveloper looks to anticipate future events. Normally, you won't need to use this feature, except possibly for signals with extremely sensitive transients. If you do raise the Lookahead value, you may need to adjust the attack time accordingly.

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

Expander

The Expander is similar to a compressor except that it increases, rather than reduces, the dynamic range above the Threshold level. You can use the Expander to add liveliness and freshness to your audio, specifically by emphasizing the transients of highly compressed signals.

Expander Parameters



- Threshold slider and field: Sets the level above which the Expander expands the signal.
- Ratio slider and field: Sets the ratio by which the signal is expanded when it exceeds the threshold.
- Attack knob and field: Sets the amount of time it takes for the expander to react when the signal exceeds the threshold.
- Release knob and field: Sets the amount of time it takes for the expander to stop expanding the signal once the signal falls below the threshold.
- *Knee knob and field:* Sets whether the signal is slightly expanded at levels just below the threshold.
- Gain slider and field: Sets the amount of output gain.
- Auto Gain button: When selected, Auto Gain compensates for the increase in gain produced by expansion.
- Expansion display: Shows the expansion curve applied to the signal.
- *Peak/RMS buttons*: Turn on one or the other to set whether the Expander uses the Peak or RMS method to analyze the signal.

Because the Expander is a genuine upward expander (as opposed to a downward expander that increases the dynamic range below the Threshold), the *Ratio* slider features a value range of 1:1 to 0.5:1.

When using the Expander with Auto Gain active, the signal will sound softer even when the peak level remains the same; in other words, the expander decreases loudness. If you dramatically change the dynamics of a signal (by setting higher Threshold and Ratio values), you may find that you need to reduce the output level using the Gain slider to avoid distortion. In most cases, turning on Auto Gain will adjust the signal to the correct level.

Limiter

The Limiter functions similarly to a compressor with one important difference: where a compressor proportionally reduces the signal when it exceeds the threshold, a limiter reduces any peak above the threshold to the threshold level, effectively limiting the signal to this level. The Limiter is used primarily as a mastering effect.

Limiter Parameters



- Gain reduction meter: Shows the amount of limiting while the signal plays.
- Gain slider and field: Sets the amount of gain applied to the input signal.
- Lookahead slider and field: Adjusts how far ahead (in milliseconds) the Limiter analyzes the audio signal.
- Release slider and field: Sets the amount of time after the signal falls below the threshold before the Limiter stops limiting.
- Output Level knob and field: Sets the output level of the signal.
- Softknee button: When selected, the signal is limited only when it reaches the threshold. When switched on, the transition to full limiting is nonlinear, producing a softer, less abrupt effect, and reducing distortion artifacts that can be produced by hard limiting.

The Lookahead parameter allows the Limiter to look forward in the audio so that it can react earlier to peak volumes by adjusting the amount of reduction. Using Lookahead causes latency, but this latency has no perceptible effect when you use the Limiter as a mastering effect, on previously recorded material. Set Lookahead to higher values if you want the limiting effect to take place before the maximum level is reached, creating a smoother transition.

Typically, you apply the Limiter as the very last effect in the mastering signal chain. In this case, you use the Limiter to raise the overall volume of the signal, so that it reaches but does not exceed 0 dB.

The Limiter is designed in such a way that if set to 0 dB Gain and 0 dB Output Level, it has no effect (on a normalized signal). If the signal clips (red gain line), the Limiter—using its basic settings—reduces the level before clipping can occur. (However, the Limiter cannot fix audio that was clipped during recording).

Multipressor

The Multipressor (short for multiband compressor) is an extremely versatile tool used in mastering audio. It splits the incoming signal into different frequency bands (from one to four), and allows you to apply compression to each band independently. After compression is applied, the bands are combined into a single output signal.

The advantage of compressing different frequency bands separately is to apply a higher amount of compression to the bands that need it without producing the pumping effect often associated with high amounts of compression. Using the Multipressor, you can apply higher compression ratios to specific frequency bands, and therefore achieve a higher average volume, without causing audible artifacts.

Raising the overall volume level can result in a dramatic increase of the existing noise floor. Each frequency band features *downward expansion*, which allows you to reduce or suppress this noise. Downward expansion works as a counterpart to compression: while the compressor compresses the dynamic range of the higher volume levels, the downward expander expands the dynamic range of the lower volume levels. With downward expansion, the signal is reduced in level when it falls below the threshold level. This functions in a similar way as a noise gate, but rather than simply cutting off the sound, it smoothly fades the volume using an adjustable ratio.

Multipressor Parameters



The parameters in the Multipressor window are grouped into three main areas: the upper graphic display section, the lower set of controls for each frequency band, and the output parameters on the right.

Graphic Display Section

- *Graphic display:* Each frequency band is represented graphically. The amount of gain change from 0 dB is shown graphically by the blue bars. For active bands, the band number appears in the center of its area. You can adjust each frequency band independently in the following ways:
 - Drag the horizontal bar up or down to adjust the gain makeup for that band.
 - Drag the vertical edges left or right band to set the crossover frequencies for that band (which adjusts the band's frequency range).
- Crossover fields: Sets the crossover frequency between adjacent bands.
- Gain Make-up fields: Sets the amount of the gain makeup for each band.

Frequency Band Section

Below the graphic band display are value fields and other parameters for controlling each frequency band:

- Compr Thrsh fields (short for Compression Threshold): Sets the compression threshold for the selected band. Setting the parameter to 0 dB results in no compression of the band.
- Compr Ratio fields (short for Compression Ratio): Sets the compression ratio for the selected band. Setting the parameter to 1:1 results in no compression of the band.

- Expnd Thrsh fields (short for Expansion Threshold): Sets the expansion threshold for the selected band. Setting the parameter to its minimum value (–50 dB), means that only signals that fall below this level are expanded.
- Expnd Ratio fields (short for Expansion Ratio): Sets the expansion ratio for the selected band.
- Expnd Reduct fields (short for Expansion Reduction): Sets the amount of downward expansion for the selected band.
- *Peak/RMS fields:* Enter a smaller value for shorter peak detection, or a larger value for RMS detection (in milliseconds).
- Attack fields: Sets the amount of time (in milliseconds) that the signal exceeds the threshold before compression starts for the selected band.
- Release fields: Sets the time (in milliseconds) required after the signal falls below the threshold before compression stops on the selected band.
- Band on/off buttons: Each band has a button (numbered from 1 to 4). Click the button to turn on the band (the button becomes light blue, and the band appears in the graphic display area above). Click the button again to turn off the band.
- Bypass buttons: Turn on to bypass the selected frequency band.
- Solo buttons: Turn on to solo the selected frequency band.
- Level meters: The light blue bar on the left shows the input level, and the dark blue bar on the right shows the output. Drag the upper triangular notch to adjust the Compression Threshold (Compr Thrsh), and drag the lower triangular notch to adjust the Expansion Threshold (Expnd Thrsh). Drag between the two notches to move both.

Output Parameters

- Auto Gain pop-up menu: Controls whether the Multipressor references the overall processing of the signal to 0 dB, making the output louder (On), or produces more standard compression, with compressed bands attenuated by the amount the dynamic range is reduced (Off).
- Lookahead value field: Adjusts how far the processor looks forward in the audio, in order to react earlier to peak volumes for smoother transitions.
- Out Gain slider: Sets the overall gain at the output.
- Level meter: Shows the overall output level.

Using the Multipressor

In the graphic display, the blue bars show the gain change, not only the gain reduction as in a standard compressor. The gain change displayed is a composite value of the compression reduction + expander reduction + auto gain compensation + gain makeup.

Compression Parameters

The Compression Threshold and Compression Ratio parameters are the key parameters used to control compression. In most cases, the most useful combinations of these two settings are either a low Compression Threshold with a low Compression Ratio, or a high Compression Threshold with a high Compression Ratio.

Downward Expansion Parameters

The Expansion Threshold, Expansion Ratio, and Expansion Reduction parameters are the key parameters for controlling downward expansion. They determine the strength of expansion applicable to the range that you wish to expand.

Peak/RMS, Attack, and Release Parameters

Adjusting the parameter between Peak (0 ms, minimum value) and RMS (Root Meantime Square –200 ms, maximum value) is dependent on the type of signal you would like to compress. An extremely short Peak detection setting is suitable for compression of short and high peaks of low power, which do not typically occur in music. The RMS detection method measures the power of the audio material over time and thus works much more musically. This is because human hearing is more responsive to the overall power of the signal than to single peaks. As a basic setting for most applications, the centered position is recommended.

Output Parameters

The Out Gain slider sets the overall output level. Set Lookahead to higher values when the Peak/RMS fields are set to higher values (further towards RMS). Setting Auto Gain to On references the overall processing to 0 dB, making the output louder.

Noise Gate

The Noise Gate is commonly used to suppress unwanted noise that is audible when the audio signal is at a low level. You can use it to remove background noise, crosstalk from other signal sources, and low-level hum, among other uses.

The Noise Gate works by allowing signals above the Threshold level to pass unimpeded while reducing signals below the Threshold level. This allows you to remove lower-level parts of the signal, while allowing the intended parts of the audio to pass.

Noise Gate Parameters



Main Parameters

- Threshold slider and field: Sets the level (in decibels) below which the signal is reduced.
- Reduction slider and field: Sets the amount by which the signal is reduced.
- Attack knob and field: Sets the amount of time it takes to fully open the gate after the signal exceeds the threshold.
- Hold knob and field: Sets the amount of time the gate is kept open after the signal falls below the threshold.
- Release knob and field: Sets the amount of time it takes to fully close the gate after the signal falls below the threshold.
- Hysteresis slider and field: Sets the difference (in decibels) between the threshold values that open and close the gate, to prevent it rapidly opening and closing when the input signal is close to the threshold.
- Lookahead slider and field: Sets how far ahead (in milliseconds) the noise gate analyzes the signal.

Sidechain Parameters

- *Monitor button:* Turn on to preview the Sidechain signal, including the effect of the High and Low Cut filters.
- High Cut slider and field: Sets the upper cutoff frequency for the sidechain signal.
- Low Cut slider and field: Sets the lower cutoff frequency for the sidechain signal.

When no external sidechain is selected, the input signal is used as the sidechain.

Using the Noise Gate

In most situations, setting the Reduction slider to the lowest possible value ensures that sounds below the Threshold are completely suppressed. Setting it to a higher value attenuates low-level sounds but still allows them to pass. You can also set Reduction to a value greater than 0 (zero) to boost the signal by up to 20 dB. This is useful for ducking effects.

The three rotary knobs for Attack, Hold, and Release modify 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 a lower value. For other sounds, such as string pads, where the signal fades in more gradually, set Attack to a higher value for a smoother effect. Similarly, when you are working with signals that fade out gradually or which have longer reverb tails, set the Release knob to a higher value so that the signal fades naturally.

The Hold knob determines the minimum amount of time that the gate stays open. This avoids abrupt changes (called chattering) caused when the Noise Gate opens and closes rapidly.

The Hysteresis slider provides another option for avoiding chattering, without needing to define a minimum Hold time. You use it to set the range between the threshold values that open and close the Noise Gate. This is useful when the signal level jitters around the Threshold, fluctuating slightly but rapidly around it. This causes the Noise Gate to switch on and off repeatedly, producing an undesirable chattering effect. Using the Hysteresis slider, you can set the Noise Gate to open at the Threshold level and remain open until the level drops below another, lower, level. As long as the difference between these two values is large enough to contain the fluctuating level of the incoming signal, the Noise Gate can function without creating chatter. This value is always negative. Generally, –6 dB is a good place to start.

In some situations, you may find that the levels of the signal you want to keep and the levels of the noise are close enough to be difficult to separate. For example, if you are recording a drum kit, and using the Noise Gate to isolate the sound of the snare drum, the hi-hat may also open the gate in many cases. To remedy this sort of situation, you can use the Sidechain controls to isolate the desired signal using Hi and Low Cut filters.

To use the Sidechain filters, click the Monitor button to turn on monitoring. This lets you hear how the Hi and Low Cut filters will affect the incoming signal. Now you can drag the High Cut slider to set the frequency above which the signal is filtered out, and drag the Low Cut slider to set the frequency below which the signal is filtered. These filters only allow very high (loud) signal peaks in their range to pass. In our example, you could remove the hi-hat's signal, which is higher in frequency, using the Hi Cut filter, and allow the snare signal to pass. You can turn monitoring off to set a suitable Threshold level more easily.

Silver Compressor

The Silver Compressor is a simplified version of the Compressor. It has fewer parameters and requires less CPU power.

Silver Compressor Parameters



- Gain Reduction display: Shows the amount of compression applied as the audio plays.
- *Threshold slider and field:* Sets the threshold for the Compressor (the level above which the signal is reduced.)
- Attack knob and field: Sets the attack time (the amount of time it takes for the compressor to react when the signal exceeds the threshold).
- Release knob and field: Sets the release time (the amount of time it takes for the compressor to stop reducing the signal once the signal falls below the threshold).
- Ratio slider and field: Sets the compression ratio (the ratio by which the signal is reduced when it exceeds the threshold.)

Using the Silver Compressor

The parameters of the Silver Compressor work in the same way as on the Compressor. For more information, see "Compressor" on page 62.

Silver Gate

The Silver Gate is a simplified version of the Noise Gate. It has fewer parameters and requires less CPU power.

Silver Gate Parameters



- Lookahead slider and field: Adjusts how far ahead (in milliseconds) the noise gate analyzes the signal.
- *Threshold slider and field:* Sets the level (in decibels) below which the signal is reduced.
- Attack knob and field: Sets the amount of time it takes to fully open the gate after the signal exceeds the threshold.
- Hold knob and field: Sets the amount of time the gate is kept open after the signal falls below the threshold.
- Release knob and field: Sets the amount of time it takes to fully close the gate after the signal falls below the threshold.

Using the Silver Gate

The parameters of the Silver Gate work in the same way as on the Noise Gate. For more information, see "Noise Gate" on page 76.

Surround Compressor

The Surround Compressor, based on the Compressor, is especially adapted for compressing complete surround mixes. The Surround Compressor is especially useful when inserted in a surround output or on channels and busses that carry multichannel audio.

You can adjust the compression ration, knee, attack, and release for both the main channels and for the LFE channel. Both the main channels and the LFE channel include an integrated limiter. In addition, you can set the threshold and output level for each channel independently.

You can also link channels by assigning them to one of three groups. When you adjust the threshold or output parameter for any channel assigned to a group, that parameter is adjusted by the same amount for all channels assigned to the group.

Surround Compressor Parameters

The Surround Compressor is divided into three sections: The Link section at the top contains a series of menus where you assign each channel to a group. The Main section includes controls common to all the main channels, and the threshold and output controls for each channel. The LFE section on the lower right includes separate controls for the LFE channel.



- *Circuit Type slider and field:* Choose the type of circuit emulated by the Compressor. The choices are Platinum, Classic A. R. Classic A. U., VCA, FET, and Opto (optical).
- Detection pop-up menu: Choose whether the Surround Compressor uses the maximum of all detection signals (Max) or the sum of all detection signals (Sum) to determine if it is above or below the threshold.

Link Section

- Grp. (Group) pop-up menus: For each channel, choose whether the channel is in group A, B, or C, or in no group (–). Moving the Threshold or Output Level slider for any channel assigned to a group will move the sliders for all channels assigned to that group.
- Byp (Bypass) buttons: For each channel, click to bypass that channel.

Main Section

- Ratio slider and field: Sets the ratio by which the signal is reduced when it exceeds the threshold.
- *Knee knob:* Adjusts whether the signal is compressed immediately or more gradually at levels close to the threshold.
- Attack knob: Sets the amount of time it takes to reach full compression after the signal exceeds the threshold.
- *Release knob:* Sets the amount of time it takes to return to zero compression after the signal falls below the threshold.
- Auto button: When selected, the release time dynamically adjusts to the audio material.
- Limiter button: Turns limiting for the main channels on or off.
- Threshold knob: Sets the threshold for the limiter on the main channels.
- *Main Compressor Threshold sliders*: Sets the threshold (the level above which the signal is reduced) for each channel.
- Main Output Levels controls: Sets the output level for each channel.

LFE Section

- Ratio slider and field: Sets the compression ratio for the LFE channel.
- Knee knob: Sets the knee for the LFE channel.
- Attack knob: Sets the attack time for the LFE channel.
- Release knob: Sets the release time for the LFE channel.
- Auto button: When selected, the release time automatically adjusts to the audio signal.
- Limiter button: Turns on limiting for the LFE channel on or off.
- Threshold knob: Sets the threshold for the limiter on the LFE channel.

Using the Surround Compressor

Using the Link controls, you can assign each channel independently to one of three groups (Group A, B, or C). When you adjust the Threshold or Output Level slider for any channel in a group, the sliders for all channels in the same group are adjusted by the same amount. Also, clicking the Bypass button for any grouped channel bypasses all channels in the group.

You can temporarily unlink a channel by pressing and holding Command and Option while you move the Threshold or Output Level slider for that channel. When you move either slider without pressing Command-Option, the channels move in step, keeping their relative positions. This allows you to make independent threshold settings while maintaining the sidechain detection link necessary for a stable surround image.

EQ 5

EQ (short for Equalization) lets you shape the sound of your audio by changing the level of specific frequency bands.

EQ is one of the most commonly used audio effects, both for music projects and in post-production work for video. You can use EQ to shape the sound of an audio file, track, or project by adjusting specific frequencies or frequency ranges. Using EQ, you can create both subtle and extreme changes to the sound of your projects.

EQ effects include a variety of single-band filters and multiband EQs. All EQ effects use filters which allow certain frequencies to pass through unchanged, while raising or lowering the level of other frequencies (also referred to as *boosting* or *cutting* frequencies). EQs can be used as "broad brush" effects to boost or cut a large range of frequencies, and some EQs (particularly parametric and multiband EQs) can be used for more precision work.

Single Band EQs

The simplest types of EQ effects are single band EQs, which include low and high cut, low and highpass, shelving, and parametric EQ.

- Low cut EQ attenuates only frequencies below a specific frequency, called the *cutoff* frequency, by a fixed number of decibels per octave, called the *slope*. High cut EQ attenuates only frequencies above its cutoff frequency, by a fixed slope.
- Lowpass EQ attenuates frequencies above the cutoff frequency, while highpass EQ lowers frequencies below the cutoff. In addition, you can control the slope of the filter (how gradually frequencies beyond the cutoff are attenuated) using the Order parameter.
- High and low shelving EQ lets you set the cutoff frequency and also control the *gain* (the amount of boost or cut), allowing you to change it by a fixed amount rather than a slope.
- Parametric EQ boosts or cuts all frequencies close to the *center frequency* (both above and below the center frequency). You can set the center frequency, and also set the *bandwidth* or *Q*, which determines how wide a range of frequencies around the center frequency are altered.

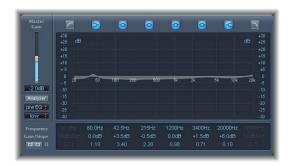
Multiband EOs

Multiband EQs give you control over a set of filters which together cover a large part of the frequency spectrum. On multiband EQs, you can set the frequency, bandwidth, and Q of each band independently. Using a multiband EQ (such as the Channel EQ, Fat EQ, or Linear Phase EQ), you can perform extensive tone-shaping on any audio source. Multiband EQs are equally useful for shaping the sound of an individual track or an overall project mix.

The following sections describe the individual effect included with Logic Studio.

- "Channel EQ" on page 84
- "DJ EQ" on page 88
- "Fat EQ" on page 88
- "Linear Phase EQ" on page 90
- "Match EQ" on page 91
- "Single Band EQs" on page 96
 - "High Cut and Low Cut Filter" on page 96
 - "High Pass and Low Pass Filter" on page 96
 - "High Shelving and Low Shelving EQ" on page 96
 - "Parametric EQ" on page 96
- "Silver EQ" on page 97

Channel EQ



The Channel EQ is a highly versatile multiband EQ. It gives you eight frequency bands, including low and highpass filters, low and high shelving filters, and four flexible parametric bands. It also features an integrated Fast Fourier Transform (FFT) Analyzer that you can use to view the frequency curve of the audio you want to modify, allowing you to see which parts of the frequency spectrum need to be boosted or cut.

You can use the Channel EQ in many ways: to shape the sound of individual tracks or audio files, or for tone-shaping on an overall project mix. With its Analyzer and graphic controls, it is very easy to observe the audio signal and make adjustments in real time.

Channel EO Parameters

On the left side of the Channel EQ window is the Gain control and parameters for the Analyzer, while the central area of the window includes the graphic display and parameters for shaping each EQ band.

- Master Gain slider and field: Sets the output level of the signal. After boosting or cutting individual frequency bands, you can use the Master Gain fader to adjust the output level.
- Analyzer button: Turns the Analyzer on or off.
- *Pre/Post EQ button:* When Analyzer mode is active, sets whether the Analyzer shows the frequency curve before or after EQ is applied.
- Resolution pop-up menu: Choose the sample resolution for the Analyzer from the menu. The choices are: low (1024 points), medium (2048 points), and high (4096 points).

Graphic Display Section

- Band On/Off buttons: Located above the graphic display. Click a button turn the corresponding band on or off. Each button has a icon showing the type of EQ it uses:
 - Band 1 is a highpass filter.
 - Band 2 is a low shelving filter.
 - Bands 3 through 6 are parametric bell filters.
 - Band 7 is a high shelving filter.
 - Band 8 is a lowpass filter.
- *Graphic display:* Shows the current curve of each EQ band. You can adjust the frequency of each band by dragging left or right in the section of the display for that band, and adjust the gain of each band (except bands 1 and 8) by dragging up or down in the band's section. The display reflects your changes immediately.

Parameter Section

Below the graphic display area are controls that you can use to show the settings for each band and adjust each band's settings.

- Frequency fields: Adjust the frequency of each band.
- *Gain/Slope fields*: Adjust the amount of gain for each band. For bands 1 and 8, this changes the slope of the filter.
- *Q fields:* Adjust the Q or resonance for each band (the range of frequencies around the center frequency that are affected).

The Q parameter of band 1 and band 8 has no effect when the slope is set to 6 dB/Oct. When the Q parameter of bands 3 through 6 is set to an extremely high value (such as 100), these filters only affect a very narrow frequency band, and can be used as notch filters.

• Link button: Activates Gain-Q coupling, which automatically adjusts the Q (bandwidth) when you raise or lower the gain on any EQ band, to preserve the perceived bandwidth of the bell curve.

Setting Gain-Q Couple to *strong* preserves the perceived bandwidth almost entirely, while *light* and *medium* settings allow some change as you raise or lower the gain. The asymmetric settings feature a stronger coupling for negative gain values than for positive values, so the perceived bandwidth is more closely preserved when you cut than when you boost gain.

Note: If you play back automation of the Q parameter with a different Gain-Q Couple setting, the actual Q values will be different than when the automation was recorded.

- Analyzer Mode pop-up menu (extended parameter): Choose Peak or RMS.
- Analyzer Decay slider and field (extended parameter): Adjust the decay rate (in dB per second) of the Analyzer curve (peak decay in Peak mode or an averaged decay in RMS mode).
- Gain-Q Couple Strength pop-up menu (extended parameter): Choose the amount of Gain-Q coupling.

Using the Channel EQ

How you use the Channel EQ depends on your audio and what you intend to do, but a useful workflow for many situations is as follows: with the Channel EQ set to a flat response (no frequencies boosted or cut), turn on the Analyzer and play the audio, observing the graphic display to see which parts of the frequency spectrum have frequent peaks and which stay at a low level. Notice particularly any places where the signal distorts or clips. Then, using the graphic display or the Parameter controls, adjust the frequency bands as desired to obtain the sound you want.

You can attenuate those frequencies that experience clipping to reduce or eliminate the distortion, and raise the quiet areas to make them sound more pronounced. You can adjust the center frequency for bands 2 through 7 to affect a specific frequency (either one you want to emphasize, such as the root note of the music, or one you want to eliminate, such as hum or other noise), and narrow the Q so that only a narrow range of frequencies are affected, or widen it to alter a broad area.

In the graphic display, each EQ band appears as a different color. You can graphically adjust the frequency of a band by dragging horizontally in the area of the band. Drag vertically to adjust the amount of gain for the band (For bands 1 and 8, the slope values can only be changed in the parameter area below the graphic display). Each band has a pivot point, which appears as a small circle on the curve at the location of the band's frequency; you can adjust the Q or width of the band by dragging the pivot point vertically.

You can also adjust the decibel scale of the graphic display by vertically dragging either the left or right edge of the display (where the dB numbers appear) when the Analyzer is not active. When the Analyzer is active, dragging the left edge adjusts the linear dB scale, and dragging the right edge adjusts the Analyzer dB scale.

To increase the resolution of the EQ curve display in the most interesting area around the zero line, drag the dB scale on the left side of the graphic display upward. Drag downward to decrease the resolution. The overall range is always ± 30 , but small values are easier to recognize.

When you work with the Channel EQ, you can turn off any bands you are not using to shape the sound. Inactive bands do not use any computer resources.

Using the Analyzer

When you turn on the Analyzer, the Channel EQ shows a realtime curve of all frequency components of the signal as the audio plays, superimposed over the EQ curves you set, using a Fast Fourier Transformation (FFT). The Analyzer curve uses the same scale as the EQ curves, allowing you to easily recognize important frequencies in the audio and use the EQ curves to raise or lower them.

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

When choosing a resolution from the menu, keep in mind that the higher the resolution, the more CPU power is required. High resolution is necessary whenever you need reliable results in very low bass frequencies, for example. The bands derived from FFT analysis are divided in accordance with the frequency linear principle, meaning that there are more bands in higher octaves than in lower ones.

Note: The FFT Analyzer needs additional CPU resources. In fact, CPU usage increases significantly at higher resolutions! It is recommend that you disable the Analyzer or close the Channel EQ window when you play or record the project, after setting the desired EQ parameters. This will free up CPU resources for other tasks.

DJ EQ

The DJ EQ combines high and low shelving filters, each with a fixed frequency, and one parametric EQ for which you can adjust Frequency, Gain, and Q-Factor. A special feature of the DJ EQ is that it allows the gain of the filters to be reduced up to -30 dB.

DJ EO Parameters

- High Shelf field and slider: Sets the amount of gain for the high shelving filter.
- Frequency field and slider: Sets the center frequency of the parametric EQ.
- Q-Factor field and slider: Sets the range (bandwidth) of the parametric EQ.
- Gain field and slider: Sets the amount of gain for the parametric EQ.
- Low Shelf field and slider: Sets the amount of gain for the low shelving filter.

Fat EO



The Fat EQ effect is a versatile multiband EQ with up to five individual frequency bands. You can use Fat EQ for individual tracks or for overall mixes. The Fat EQ includes a graphic display of the EQ curves and a set of parameters for each band.

Fat EQ Parameters

The main area of the Fat EQ window includes a graphic display area and a set of strips with parameters for each frequency band. To the right of the parameter section are the Master Gain slider and field.

Graphic Display Section

- Band Type buttons: Located above the graphic display. For bands1-2 and 4-5, click one of the pair of buttons to select the type of EQ for the corresponding band.
 - For Band 1, click the highpass or the low shelving button.
 - For Band 2, click the low shelving or the parametric button.
 - Band 3 always acts as a parametric EQ band. (Click the button to turn it on or off.)
 - For Band 4, click the parametric or the high shelving filter.
 - For Band 5, click the high shelving or the lowpass button.

• *Graphic display:* Shows the EQ curve of each frequency band. When you adjust each band's settings using the controls in the Parameter section, the display reflects your changes immediately.

Parameter Section

Below the graphic display area are controls which both show the settings for each band, and which you can use to adjust each band's settings.

- Frequency fields: Sets the frequency for each band.
- Gain knobs: Sets the amount of gain for each band.
- *Q/Order fields:* Sets the Q or bandwidth for each band (the range of frequencies around the center frequency that are altered). For bands 1 and 5, this changes the slope of the filter.
- Band On/Off buttons: Click the numbered button to turn each band on or off. Inactive bands do not use your computer's resources.

Master Gain Section

• Master Gain slider and field: Located to the right of the Parameter section. Sets the output level of the signal. After boosting or cutting frequency bands, you can use the Master Gain fader to adjust the output level.

Using the Fat EQ

The icons above the graphic display let you switch the type of EQ for each band, except for Band 3, which always operates as a fully parametric bell filter. You can use the controls in the Parameter section to set the frequency, gain, and Q for each band, as well as turn individual bands on or off.

At low *Q* values, the EQ covers a wider frequency range, while at high *Q* values, the effect of the EQ band is limited to a very narrow frequency range. Keep in mind that the Q value can significantly influence how audible your changes are: if you're working with a narrow frequency band, you'll generally need to cut or boost it more drastically to notice the difference.

Linear Phase EQ



The high-quality Linear Phase EQ effect is similar in appearance to the Channel EQ, sharing the same parameters and eight-band layout. However, the Linear Phase EQ uses a different underlying technology, which preserves the phase of the audio signal 100%—even when you apply the wildest EQ curves to the sharpest signal transients!

The Linear Phase EQ uses more CPU resources than the Channel EQ, and introduces greater amounts of latency. For this reason, it is strongly recommended that you use it for mastering previously recorded audio—and don't use it when playing software instruments live, for example. The latency compensation feature in Logic Pro can be helpful when using this effect as well.

Linear Phase EO Parameters

The parameters of the Linear EQ are identical to the Channel EQ. For information on the Channel EQ parameters, see "Channel EQ Parameters" on page 85.

Using the Linear Phase EQ

In operation, the Linear Phase EQ is similar to the Channel EQ. For more information, refer to "Channel EQ" on page 84. Because the parameters of the Channel EQ and Linear Phase EQ are almost identical, you may freely copy settings between them. In Logic Pro, if you replace a Channel EQ with a Linear Phase EQ on the same slot (or vice versa), the settings are transferred to the new EQ.

One difference, however, is that the Linear Phase EQ uses a set amount of your computer's CPU resources, regardless of how many bands are active.

Match EQ

The Match EQ effect allows you to store the average frequency spectrum of an audio file as a template and apply the template to your project, so that it matches the spectrum of the original file. Using Match EQ you can acoustically match the sound of different songs you plan to include on an album, or impart the particular sound of any source recording to your own projects. Beyond matching the project's frequency spectrum to the original file's EQ, you can also manually modify the filter curve before you apply it to your project.

Note: Match EQ acoustically matches the frequency curve of two audio signals. However, it does not match any dynamic differences in the two signals.

Match EQ Parameters



Left Side

• Analyzer button: Turns the Analyzer function on or off.

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

- *Position button:* Sets whether the Analyzer looks at the signal before the filter curve is applied (Pre), or after (Post).
- *View pop-up menu:* Choose what information appears on the graphic display. The choices are:
 - Automatic: Automatically displays information for the current function, as determined by which of the buttons below the graphic display is selected.
 - *Template*: Displays the frequency curve for the source file learned as the template (shown in red).
 - Current Material: Displays the frequency curve for the audio learned as the current material (the track on which the Match EQ is applied, or a loaded plug-in settings file or template—shown in green).
 - *Filter:* Displays the filter curve created by matching the template and the current material (shown in yellow).

- Format button: Sets whether the Analyzer displays audio channels via separate curves (L&R for stereo, All Cha for surround) or the summed maximum level (LR Max for stereo, Cha Max for surround).
- Select buttons: Click one of the buttons to control whether any changes you make to the filter curve created by matching the template with the current material are applied only to the left channel (L), the right channel (R), or both channels (L+R).
- Select menu (not pictured): When using a surround instance of the Match EQ, the Select buttons are replaced by a menu, from which you can choose an individual channel for changes to the filter curve, or choose All.
- Channel Link slider: Refines the settings made using the Select buttons or Select menu. When set to 1.0, all channels (L and R for stereo, or all channels for surround) are represented by a common EQ curve. When set to 0.0 (the minimum value), a separate filter curve is displayed for each channel, which can be selected using the Select buttons or Select menu. Settings between 0.0 and 1.0 allow you to blend these values with your changes to the filter curves being transferred to each channel, as determined by the setting.

Note: The View, Select, and Channel Link parameters are disabled when using the effect on a mono channel.

• LFE Handling (extended parameter): In surround instances, the LFE Handling radio buttons allow you to process or bypass the LFE channel.

Center Section

- *Graphic display*: Displays the filter curve created by matching the template to the current material. You can edit the filter curve (see "Editing the Filter Curve").
- *Template Learn button:* Click to start the process of learning the frequency spectrum of the source file. Click again to stop the learning process.
- Current Material Learn button: Click to start the process of learning the frequency spectrum of the project you want to match the source file. Click again to stop the learning process.
- *Current Material Match button:* Matches the frequency spectrum of the current material to that of the template (source) file.

Right Side

- *Phase button:* Sets whether processing alters (Minimal) or is prevented from altering the signal phase (Linear). The Linear setting increases latency, while the Minimal setting results in lower latency.
- Apply slider and field: Modifies the effect of the filter curve on the signal. Values between 101% and 200% magnify the effect, values between 1% to 99% reduce it, and values from –1% to –100% invert the peaks and troughs in the filter curve. A value of 100% produces no modification of the filter curve.

• Smoothing slider and field: Sets the amount of smoothing for the filter curve. A value of 0.0 leaves the filter curve unchanged. At all other settings, the filter curve is smoothed at a constant bandwidth, determined by the set value in semitones. For example, a value of 1.0 means that the smoothing uses a bandwidth of one semitone. A value of 4.0 produces a smoothing bandwidth of four semitones (a major third), a value of 12.0 produces a bandwidth of one octave, and so on.

Note: Smoothing does not affect any manual changes you make to the filter curve.

• Fade Extremes pop-up menu (extended parameters): Choose On to smooth the filter curve at the high and low extremes of the frequency spectrum.

Using the Match EQ

Match EQ is a learning equalizer that analyzes or learns the frequency spectrum of an audio signal, such as, an audio file, a track input signal, or a template. You can also load a previously saved plug-in settings file, or import the settings of another Match EQ instance by copying and pasting.

Match EQ analyzes the average frequency spectrum of the source file (the template) and of your project (the current material), then matches the two spectra, creating a filter curve. This filter curve adapts the frequency response of the current material to match that of the template. Before applying the filter curve, you can modify it by boosting or cutting any number of frequencies, or by inverting the curve. The Analyzer allows you to visually compare the frequency spectrum of the source file and the resulting curve, making it easier to make manual corrections at specific points within the spectrum.

You can use the Match EQ in different ways, depending on your intent and the audio you're working with. In general, you will want to make your mix sound similar to an existing recording—either your own or that of another artist. Following is a common usage example that you can adapt to your own workflow. In this example, you match the frequency spectrum of a mix to the spectrum of a source audio file.

To match the EQ of a project mix to the EQ of a source audio file:

- 1 In the project you want to match to the source audio file, instantiate a Match EQ (typically on Output 1-2).
- 2 Drag the source audio file onto the Template Learn button.
- 3 Return to the start of your mix, click Current Material Learn, then play your mix (the current material) from start to finish.
- 4 When done, click Current Material Match (this automatically disengages the Current Material Learn button).

To use the matched EO on a track:

- 1 Set the track you want to match as a sidechain input to the Match EQ.
- 2 Click the "Template Learn" button, play the entire source audio track from start to finish, then click the "Template Learn" button again.
- 3 Return to the start of your mix, click Current Material Learn, then play your mix (the current material) from start to finish.
- 4 When done, click Current Material Match (this automatically disengages the Current Material Learn button).

Match EQ creates a filter curve based on the differences between the spectrum of the template and the current material. This curve automatically compensates for differences in gain between the template and the current material, with the resulting EQ curve referenced to 0 dB. A yellow filter response curve appears in the graphic display, showing the average spectrum of your mix approximating (mirroring) the average spectrum of your source audio file.

You can also drag an audio file onto the Template Learn or Current Material Learn buttons to use as either the template or the current material. A progress bar appears while the Match EQ is analyzing the file.

Control-clicking (or right-clicking, with an appropriate mouse) either of the Learn buttons opens a shortcut menu containing a variety of options to apply to the spectrum of the template or the current material, including: clearing, copying to the Match EQ clipboard (which you can access from any Match EQ instance in the current project), pasting from the Match EQ clipboard to the current instance, loading from a stored settings file, or generating a frequency spectrum for an audio file you choose in a File dialog.

When you click either of the Learn buttons, the View parameter is set to Automatic, and the graphic display shows the frequency curve for the function matching the selected button. You can review any of the frequency curves when no file is being processed by choosing one of the other View options.

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

Only one of the Learn buttons can be active at a time. For example, if the Learn button in the Template section is active and you press the Learn button in the Current Material section, the analysis of the template file stops, the current status is used as the spectral template, and analysis of the track (Current Material) begins.

Note: Each time you match two audio signals—either by loading/learning a new spectrum while Match is activated or by activating Match after a new spectrum has been loaded—any existing changes to the filter curve are discarded, and Apply is set to 100%.

By default, the Apply slider is set to 100% when you learn the frequency curve of an audio signal. In many cases, you may want to lower it slightly to avoid extreme spectral changes to your mix. It is also recommended that you use the Smoothing slider in order to adjust the spectral detail of the generated EQ curve.

Editing the Filter Curve

You can graphically edit the matched filter curve in the graphic display, by clicking any point on the filter curve. Drag horizontally to shift the peak frequency for this band (over the entire spectrum). Drag vertically to adjust the gain of this band (between –24 to +24 dB). To adjust the Q-Factor, hold down the Shift key and drag vertically. Hold down the Option key while dragging to reset the gain to 0 dB. As you drag, the current values appear in a small box inside the graphic display, allowing you to make precise adjustments graphically.

Note: If you manually modify the filter curve, you can restore it to the original (or flat) curve by Option-clicking on the background of the Analyzer display. Option-clicking the background again restores the most recently modified curve.

The Q-factor of the filter is set by the vertical distance between the point where you click and the curve. By clicking on the curve, the maximum Q-value of 10 (for notch-like filters) is used. Clicking above or below the curve decreases the Q-value. The further you click from the curve, the smaller the value (down to the minimum of 0.3).

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

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

Single Band EQs

Following are descriptions of each of the effects found in the Single Band submenu.

High Cut and Low Cut Filter

As their names suggest, the Low Cut Filter attenuates the frequency range below the selected frequency, while the High Cut Filter attenuates the frequency range above the selected frequency. Each has a single parameter letting you set the cutoff frequency.

High Pass and 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 eliminate the bass below a selectable frequency. In contrast, the Low Pass Filter affects the frequency range above the selected frequency. Both filter plug-ins offer the following parameters:

- Frequency field and slider: Sets the cutoff frequency.
- Order field and slider: Sets the filter order.
- Smoothing field and slider: Adjusts the amount of smoothing (in milliseconds).

High Shelving and Low Shelving EQ

The Low Shelving EQ affects only the frequency range below the selected frequency, while the High Shelving EQ affects only the frequency range above the selected frequency. Each has parameters for Gain, which you use to boost or cut the level of the selected frequency band, and Frequency, which you use to set the cutoff frequency.

Parametric EQ

The Parametric EQ is a simple filter with a variable center frequency. It can be used to boost or cut any frequency band in the audio spectrum, either with a wide frequency range, or as a notch filter with a very narrow range. A symmetrical frequency range on either side of the center frequency is boosted or cut. The Parametric EQ offers the following parameters:

- Gain field and slider: Sets the amount of gain.
- Frequency field and slider: Sets the cutoff frequency.
- *Q-Factor field and slider:* Adjusts the Q (bandwidth).

Silver EQ

The Silver EQ, a Legacy effect, includes three bands: a high shelving EQ, parametric EQ, and low shelving EQ. You can adjust the cutoff frequencies for the high and low shelving EQs, and adjust the center frequency, gain, and Q for the parametric EQ.

Silver EQ Parameters

- High Frequency field and slider: Sets the cutoff frequency for the high shelving EQ.
- Frequency field and slider: Sets the center frequency of the parametric EQ.
- Q-Factor field and slider: Adjusts the range (bandwidth) of the parametric EQ.
- Gain field and slider: Sets the amount of gain for the parametric EQ.
- Low Frequency field and slider: Sets the cutoff frequency for the low shelving EQ.

Frequency Ranges Used With EQ

All sounds can be thought of as falling into one of three basic frequency ranges: bass, midrange, or high (or treble). These can each be further divided to include low bass, low and high midrange, and low and high highs. The following table describes some of the sounds that fall into each range:

Name	Frequency range	Description	
High High	8–20 kHz	Includes cymbal sounds and highest harmonics of instruments. Boosting frequencies in this range slightly can add sparkle and presence.	
High	5–8 kHz	This range corresponds roughly to the treble tone control on a stereo. Boosting frequencies in this range can add brightness and shine.	
Low High	2.5–5 kHz	Includes the higher harmonics of voices and musical instruments. This range is important for adding presence. Excessive boosting in this range can sound shrill or harsh.	
High Midrange	1.2–2.5 kHz	Includes the consonants of voices and the high harmonics of musical instruments, especially brass instruments. Excessive boosting in this range can create a pinched, nasal sound.	
Midrange	750 Hz-1.2 kHz	Includes the vowels of voices and the harmonics of musical instruments that create tone color.	
Low Midrange	250-750 Hz	Includes the fundamentals and lower harmonics of voices and musical instruments; careful EQing of each can keep them from competing. Excessive boosting in this range can result in muddy and unclear audio; excessive cutting can produce thin-sounding audio.	
Bass	50–250 Hz	Corresponds roughly to the bass tone control on a stereo. Includes the fundamental frequencies of voices and of musical instruments. Excessive boosting in this range can sound boomy and thick.	
Low Bass	50 Hz and below	Also called <i>sub bass</i> . Very little of the sound of voices or musical instruments falls in this range. Many sound effects used in movies, such as explosions and earthquakes, fall in this range.	

Note: The frequencies shown for each range are approximate. Any division of sound into frequency ranges is somewhat arbitrary, and is meant only to give a general indication of each range.

Filter 6

In addition to the filters in EQ effects, you can use filters to change the character of your audio in both familiar and unusual ways.

The Filter submenu contains a variety of filter-based effects that you can use to creatively modify your audio, including autofilters, filter banks, vocoders, wah-wah effects, and a gate that uses frequency rather than the amplitude (volume) as the criteria for which part of the signal is allowed to pass through.

The following sections describe the individual plug-ins included with Logic Studio.

- "AutoFilter" on page 100
- "EVOC 20 Filterbank" on page 103
- "EVOC 20 TrackOscillator" on page 107
- "Fuzz-Wah" on page 116
- "Spectral Gate" on page 118
- "Soundtrack Pro Autofilter" on page 120

AutoFilter



The AutoFilter is a versatile filter effect with several unique features. You can use it to create classic, analog-style synthesizer effects, or as a tool for creative sound design. The filter cutoff can be dynamically modulated using both a synthesizer-style ADSR envelope and an LFO (low frequency oscillator). In addition, you can choose between different filters types and slopes, control the amount of resonance, add distortion for more aggressive sounds, and mix the original, dry signal with the processed signal.

AutoFilter Parameters

The main areas of the AutoFilter window include the Envelope, LFO, Filter, and Distortion sections. The overall Threshold control is in the upper-left corner, and the Output controls are on the right side of the window.

Threshold Slider

The Threshold slider sets the cutoff frequency which applies to both the envelope and LFO. When the input signal level exceeds the Threshold level, the envelope and LFO are retriggered. The Threshold parameter always applies to the envelope. It applies to the LFO only if the Retrigger button is selected.

Envelope Section

- Attack knob and field: Sets the attack time for the envelope.
- Decay knob and field: Sets the decay time for the envelope.
- Sustain knob and field: Sets the sustain time for the envelope.
- Release knob and field: Sets the release time for the envelope.
- *Dynamic knob and field:* Sets the amount by which the input signal modulates the peak value of the envelope.
- Cutoff Mod. slider and field: Sets the intensity of the control signal's effect on the cutoff frequency.

LFO Section

- Coarse and Fine Rate knobs and field: Use together to set the frequency of the LFO. Drag the Coarse slider to set the LFO frequency in Hertz, then drag the Fine slider to fine tune the frequency in 1/000s of a Hertz.
- Beat Sync button: When selected, the LFO is synchronized to the sequencer's tempo.
- *Phase knob:* Lets you shift the phase relationship between the LFO and the sequencer when Beat Sync is active.
- Decay/Delay knob and field: Sets the amount of time the LFO takes to go from 0 to its maximum value.
- Rate Mod. knob and field: Sets the rate of modulation for the LFO frequency, independent of the input signal level. When the input signal exceeds the Threshold, the modulation width of the LFO increases from 0 to the Rate Mod. value.
- Stereo Phase knob and field: For stereo instances of the AutoFilter, sets the phase relationships of the LFO modulations on the two stereo channels.
- *Cutoff Mod. slider and field:* Sets the intensity of the control signal's effect on the cutoff frequency.
- Retrigger button: When selected, the waveform starts at 0° as soon as the Threshold is exceeded.
- Waveform buttons: Click one of the buttons to set the shape of the LFO waveform.
- Pulse Width slider and field: Lets you shape the curve of the selected waveform.

Filter Section

- Cutoff Freq. knob: Sets the cutoff frequency for the lowpass filter.
- *Resonance knob*: Sets the width of the frequency band around the cutoff frequency that is emphasized.
- Fatness slider and field: Adjusts the amount of fatness (low-frequency boost). When you set Fatness to its maximum value, adjusting Resonance has no effect on frequencies below the cutoff frequency.
- State Variable Filter buttons: Click one of the buttons to set whether the filter is a highpass (HP), bandpass (BP), or lowpass (LP) filter.
- 4-Pole Lowpass Filter buttons: Click one of the buttons to set the slope of the lowpass filter to 6, 12, 18, or 24 dB per octave.

Distortion Section

- *Input knob*: Sets the amount of distortion applied before the filter section.
- *Output knob:* Sets the amount of distortion applied after the filter section.

Output Section

- Dry Signal slider and field: Sets the amount of the original (dry) signal added to the filtered signal.
- *Main Out slider and field*: Sets the final output volume of the AutoFilter.

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Using the AutoFilter

The following section provides additional information on using the parameters in the AutoFilter window.

Filter Parameters

The most important parameters are located on the right side of the AutoFilter window. The Filter Cutoff knob determines the point where the filter kicks in. Higher frequencies are attenuated, while lower frequencies are allowed to pass through.

The Resonance knob controls how much frequencies around the cutoff frequency are emphasized. When you turn Resonance up sufficiently, the filter itself begins oscillating at the cutoff frequency. Self-oscillation begins before you max out the Resonance parameter, just like the filters on a Minimoog synthesizer. Increasing Resonance causes the lowpass filter to cut the bottom end, making the signal sound thinner. You can compensate for this thinness using the Fatness slider.

Both the envelope and LFO parameters are used to dynamically modulate the cutoff frequency. The Threshold parameter at the upper-left corner of the AutoFilter window applies to both sections, and analyzes the level of the input signal. If the input signal level exceeds the Threshold level, the envelope and LFO are retriggered.

Envelope Parameters

When the input signal exceeds the Threshold level, the control signal is triggered at the minimum value. Over the period of time determined by the Attack parameter, the signal reaches its maximum level. It then decreases for the period of time defined by the Decay value, and then stays at a constant level for the duration of the Sustain value. Once the signal level drops below the Threshold value, it decreases to its minimum value over the time period 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. You can modulate the peak value of the Envelope section using the level of the input signal by adjusting the Dynamic parameter. The Cutoff Mod. slider determines the intensity of the control signal's effect on the cutoff frequency.

LFO Parameters

You set the waveform of the LFO by clicking one of 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 select a waveform, you can shape the curve with the Pulsewidth slider. Use the Coarse and Fine Frequency knobs to set the LFO frequency. The Rate Mod. (Rate Modulation) knob controls modulation of the LFO frequency independent of the input signal level. If the input signal exceeds the Threshold level, the modulation width of the LFO increases from 0 to the Rate Mod. value. You can also define the amount of time this process takes, by entering the desired value with the Decay/Delay knob. If the Retrigger button is turned on, the waveform starts at 0° whenever the Threshold is exceeded. For stereo instances of the AutoFilter, you can control the phase relationships of the LFO modulations on the two stereo sides with the Stereo Phase knob.

Turning on Beat Sync synchronizes the LFO to the sequencer's tempo. The speed values include bar values, triplet values, and more. These are determined by the Rate knob next to the Beat Sync button. Use Sync Phase to shift the phase relationship between the LFO and the sequencer.

Distortion Parameters

The Distortion Input and Output parameters let you individually control pre-input and post-output distortion. Although the two distortion modules work in an identical way, their respective positions in the signal chain—before and after the filter, respectively—result in remarkably different sounds.

Output Parameters

The Dry Signal parameter sets the level ratio of the non-effected (dry) signal mixed with the processed signal. The Main Out parameter can lower the output volume by as much as 50 dB, allowing you to compensate for higher levels caused by adding distortion or other processing.

EVOC 20 Filterbank

The EVOC 20 Filterbank consists of two formant filter banks, which are also used in the EVOC 20 PolySynth vocoder plug-in for Logic Pro and MainStage.

The input signal passes through the two filter banks in parallel. Each bank features volume faders for ten frequency bands, allowing you to adjust the volume of each band independently. Setting a fader to its minimum value completely suppresses the formants in that band. You can control the position and width of the filter bands using the Formant Stretch and Formant Shift parameters. In addition, you can also crossfade between the two filter banks.

For more information on filter banks, refer to "How Does a Filter Bank Work?" on page 207.

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EVOC 20 Filterbank Parameters



The EVOC 20 Filterbank window is divided into three main sections: the Formant Filter section in the center of the window, the Modulation section at the bottom center, and Output section along the right side.

Formant Filter Section



The parameters in this section control the frequency bands in the two filter banks: Filter Bank A and Filter Bank B.

- Frequency band faders: Set the volume of each frequency band in Filter Bank A using the upper (blue) faders, and set the volume of each frequency band in Filter Bank B using the lower (green) faders.
 - You can easily create complex bar curves by dragging horizontally across either row of faders. This method makes editing multiple frequency band levels quick and convenient.
- Frequency bar and fields: The blue bar above the upper row of faders controls the overall frequency range for both filter banks. Drag the bar to move the entire frequency range, drag the left end to move only the lower frequency (between 75 and 750 Hz), or drag the right end to move only the upper frequency (between 800 and 8000 Hz). You can also edit the numerical values above the bar directly (between 80 and 8000 Hz).

- Formant Shift knob: Moves the position of all bands in both filter banks up or down the frequency range. You can jump directly to the values -0.5, -1, 0, +0.5 or +1.0 by clicking one of these numbers on the edge of the knob.
- Bands value field: Sets the number of frequency bands in each filter bank. The range is from 5 to 20 bands.

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

- Lowest button: Sets whether the lowest band of each filter bank acts as a lowpass or bandpass filter.
- Resonance knob: Controls the basic sonic character of both filter banks. Increasing the Resonance emphasizes the middle frequency of each band. Low settings give a softer character, while high settings give a sharper character.
- Boost A knob: Sets the amount of boost (or cut) applied to the frequency bands in Filter Bank A. The range is ±20 dB. This allows you to compensate for the reduction in volume caused by lowering the level of one or more bands.
 - Boost is also quite handy to adjust the levels of both filter banks to each other, so that using Fade A/B (see below) leads only to a sound color change, but not to a level change.
- *Highest button:* Sets whether the highest band of each filter bank acts as a highpass or bandpass filter.
- Slope pop-up menu: Sets the amount of filter slope applied to all filters of both filter banks. Choices are 1 (filter attenuation of 6 dB/Oct.) and 2 (filter attenuation of 12 dB/Oct.). 1 sounds softer, 2 sounds tighter.
- Boost B knob: Sets the amount of boost (or cut) applied to the frequency bands in Filter Bank B. The range is ± 20 dB. This allows you to compensate for the reduction in volume caused by lowering the level of one or more bands.
- Fade AB slider: Crossfades between Filter Bank A and Filter Bank B. At the top position (0%), only Bank A is audible, while at the bottom position (100%), only Bank B is audible. In the middle position (50%), the banks are evenly mixed.

Modulation Section



The parameters in this section control the LFOs that modulate the Formant Shift and Fade A/B parameters in the Formant Filter section, respectively. The LFO Shift parameters on the left modulate the Formant Shift parameter of the filter bands, and the LFO Fade parameters on the right modulate the Fade AB parameter.

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- LFO Shift Intensity slider: Sets the amount by which the LFO modulates the Formant Shift parameter.
- LFO Shift Rate knob: Sets the speed of the modulation for Formant Shift. Values to the left of center are synchronized to the tempo in bars and other musical values, while values to the right of center are free values in Hertz.
- Waveform buttons: Select the waveforms used by the LFO Shift and LFO Fade LFOs
 respectively. From top to bottom, the available waveforms are: triangle, falling
 sawtooth, rising sawtooth, square up and down around zero (bipolar), square up
 from zero (unipolar), sample and hold (a random stepped waveform), and smoothed
 sample and hold.
- LFO Fade Intensity slider: Sets the amount by which the LFO modulates the Fade AB parameter.
- LFO Fade Rate knob: Sets the speed of the modulation for Fade AB. Values to the left of center are synchronized to the tempo in bars and other musical values, while values to the right of center are free values in Hertz.
- Tip: The Formant Shift and Fade LFO modulations are the keys to the most extraordinary sounds of the EVOC 20 Filterbank: Make sure to set up either completely different or complementary filter curves in both filter banks. You can use rhythmic material, such as a drum loop, as an input signal, and set up tempo-synchronized modulations with different Rates for each LFO. You can then try a tempo-synchronized Tape Delay after the EVOC 20 Filterbank, to produce unique rhythms.

Output Section



The parameters in this section control the overall output of the EVOC 20 Filterbank.

- Overdrive button: Turns the overdrive circuit on or off.
 - **Note:** To hear the Overdrive effect, you may need to boost the level of one or both filter banks.
- Level slider: Sets the level of the output signal.
 Stereo Mode pop-up menu: Sets the input/output mode of the EVOC 20 Filterbank.
 The choices are m/s (mono input to stereo output), and s/s (stereo input to stereo output).
 - Set Stereo Mode to m/s if the input signal is mono, and to s/s if the input signal is stereo. In s/s mode, the left and right stereo channels are processed by separate filter banks. When using m/s mode on a stereo input signal, the signal is first summed to mono before it is passed to the filter banks.
- Stereo Width knob: Controls how the output signals of the filter bands are distributed in the stereo field.
 - At the left position, the output of all bands are centered.
 - At the centered position, the output of all bands ascends from left to right.
 - At the right position, the bands are output evenly on the left and the right channel. The stereo/stereo mode (s/s) uses one A/B filter bank per channel. The positioning of the frequency bands correspond to that described above, but the bands of each filter bank ascend in opposing directions, from left to right.

EVOC 20 TrackOscillator

The EVOC 20 TrackOscillator is a vocoder with a monophonic pitch *tracking oscillator*. The tracking oscillator allows the EVOC 20 TrackOscillator to track (follow) the pitch of a mono input signal. For example, if the input signal is a vocal melody, the individual pitches of the sung notes will be tracked and mirrored by the *synthesis* engine.

The EVOC 20 TrackOscillator features two formant filter banks, an analysis and a synthesis filter bank. Each has several (configurable) input parameters. You can use the track into which the EVOC is inserted as the analysis signal source, or use another audio track as the input source via a Side Chain. The synthesis source can be the track into which the EVOC is inserted, another audio track, or the tracking oscillator. Each analysis frequency band has an envelope follower that tracks the level of that band, so that it can be remodeled more precisely.

The signal path of the EVOC 20 TrackOscillator is shown in the block diagram on page 224.

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Tip: For good pitch tracking, it is essential to use a mono signal (with no overlap of pitches) that is as unprocessed as possible. Avoid using a signal with background noises. Using a signal processed with even a slight amount of reverb, for example, will produce strange (and likely undesirable) results. Even stranger results will result when a signal with no audible pitch (such as drum loop) is used. In some situations, however, the resulting artifacts might be desirable.

The EVOC 20 TrackOscillator is not limited to pitch tracking effects. It can vocode a signal by itself, making it very useful for unusual filter effects. Try this with different Resonance, Formant Shift, and Formant Stretch settings. As both analysis and synthesis input signals are freely selectable, you can even vocode an orchestra with train noises, for example.

More information about vocoders can be found in "Vocoder Basics" on page 206.





The EVOC 20 TrackOscillator window is divided into the following sections, from left to right: Analysis In, Synthesis In, Tracking Oscillator, Formant Filter, LFO, U/V Detection and Output.

Analysis In Section



The parameters in section control various aspects of the analysis signal.

Attack knob: Controls how quickly the envelope follower coupled to each analysis
filter band reacts to rising signals. Longer Attack times result in a slower tracking
response to transients of the analysis input signal.

Note: A long attack time on percussive input signals (such as spoken word or hi-hat parts) results in a less articulated vocoder effect. Set Attack as low as possible to achieve precise articulation.

Release knob: Controls how quickly the envelope follower coupled to each analysis
filter band reacts to falling signals. Longer release times make transients of the
analysis input signal sound for a longer period of time at the Vocoder's output.

Note: A long release time on percussive input signals results in a less articulated vocoder effect. Release times that are too short result in rough, grainy vocoder sounds. Release values of around 8 to 10 ms are a useful starting point.

Freeze button: When selected, the current analysis sound spectrum is held
indefinitely. This can capture a particular characteristic of the source signal, which is
then imposed as a complex sustained filter shape on the Synthesis section. While
Freeze is selected, the analysis filter bank ignores the input source, and the Attack
and Release parameters have no effect.

Using a spoken word pattern as a source, for example, the Freeze parameter can capture the attack or tail phase of an individual word within the pattern; for example a yowel sound.

Another use of the Freeze parameter is to compensate for people's inability to sustain sung notes for a long period without taking a breath. If you wish the synthesis signal to be sustained when the analysis source signal is not, Freeze can be used to lock the current formant levels (of a sung note), even during gaps in the vocal part, such as between words in a vocal phrase.

- Analysis In pop-up menu: Sets the analysis signal source. The choices are:
 - *Track:* Sets the audio track into which the EVOC 20 TrackOscillator is inserted as the analysis signal.
 - Side Chain: Sets the Side Chain (another audio track) as the analysis signal. You
 choose the Side Chain source track from the Side Chain pop-up menu at the top of
 the EVOC 20 TrackOscillator window.

Note: If Side Chain is selected, and no Side Chain track is assigned, the track's signal is used as the analysis source.

Synthesis In Section



The parameters in section control various aspects of the synthesis signal.

• Synthesis In pop-up menu: Sets the synthesis signal source. The choices are:

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- Oscillator (Osc.): Sets the tracking oscillator as the synthesis source. The oscillator tracks the pitch of the analysis input signal. Choosing Osc. activates the other parameters in the Synthesis section. If Osc is not chosen, the FM Ratio, FM Int, and other parameters in this section have no effect.
- *Track*: Sets the audio track into which the EVOC 20 TrackOscillator is inserted as the synthesis source signal.
- Side Chain (SideCh): Sets the Side Chain (another audio track) as the synthesis source signal. You choose the Side Chain source track from the Side Chain pop-up menu at the top of the EVOC 20 TrackOscillator window.

Note: If Side Chain is selected, and no Side Chain track is assigned, the track's signal is used as the synthesis source.

• Bands value field: Sets the number of frequency bands used by the EVOC 20 TrackOscillator filter banks. The range is from 5 to 20. The greater the number of bands, the more precisely the sound can be reshaped. Increasing the number of bands also increases the CPU overhead.

Tracking Oscillator

The parameters in the Tracking Oscillator section control various aspects of the tracking oscillator.

The FM tone generator for the tracking oscillator consists of two oscillators, each of which generates a sine wave. The frequency of Oscillator 1 (the carrier) is linearly modulated by Oscillator 2 (the modulator), which deforms the sine wave of Oscillator 1 to a waveform with a rich harmonic structure.

The FM Int control determines whether the tracking oscillators delivers a sawtooth wave or the signal of an FM tone generator.



- FM Ratio knob: Sets the ratio between Oscillators 1 and 2, which defines the basic character of the sound. Even-numbered values (or their multiples) produce harmonic sounds are produced, while odd-numbered values (or their multiples) produce inharmonic, metallic sounds.
 - An FM Ratio of 1.000 produces results resembling a sawtooth waveform.
 - An FM Ratio of 2.000 produces results resembling a square wave with a pulse width of 50%.

- An FM Ratio of 3.000 produces results resembling a square wave with a pulse width of 33%.
- FM Int knob: Selects the basic waveform and controls the intensity of FM modulation.
 - At a value of 0, the FM tone generator is disabled, and a sawtooth wave is generated instead.
 - For values higher than 0, the FM tone generator is activated. Higher values result in a more complex and brighter sound.
- Coarse Tune value field: Sets the pitch offset of the oscillator in semitones, up to ±2 octaves.
- Fine Tune value field: Sets the pitch offset in cents. The default value is concert pitch A = 440 Hz. The range is from 425.00 to 455.00 Hz.

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



- *Pitch Quantize Strength slider:* Determines how pronounced the automatic pitch correction is.
- *Pitch Quantize Glide slider:* Sets the amount of time the pitch correction takes, allowing sliding transitions to the quantized pitches.
- Root/Scale Keyboard and pop-up menu: Use together to define the pitch or pitches to which the tracking oscillator is quantized, as described below:
 - Click the value below the word *Scale* to display the Root/Scale pop-up menu.
 - Choose the scale or chord to use as the basis for pitch correction from the pop-up menu.
 - Set the root key of the respective scale or chord by vertically dragging the *Root* parameter, or double-click and enter a root between C and B. The *Root* parameter is not available when the Root/Scale value is *chromatic* or *user*.

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- You can add notes to the chosen scale or chord by clicking keys on the small keyboard, and remove notes by clicking notes already selected. Selected notes appear bright green. Selecting any notes sets the Root/Scale value to user.
- Your last edit is remembered. If you choose a new scale or chord, but do not make any changes, you can jump back to the previously set *user* scale.

You can automate the Root and Scale parameters and the keys of the onscreen keyboard in Logic Pro.

Max Track value field: Sets the frequency above which the high frequencies of the
analysis signal are cut, making the pitch detection more robust. Should the pitch
detection produce unstable results, reduce the Max Track parameter value to the
lowest possible setting.

Formant Filter Section



The parameters in section control the two formant filter banks of the EVOC 20 TrackOscillator.

- Formant Filter graphic display: Shows the frequency bands for the analysis and synthesis parameters.
 - The Formant Filter display is divided into two sections by a horizontal line. The upper half applies to the Analysis section and the lower half to the Synthesis section. Changes made to the High/Low frequency parameters, the Bands parameter or the Formant Stretch and Formant Shift parameters will result in visual changes to the Formant Filter display. This provides you with invaluable feedback on what is happening to the signal as it is routed through the two formant filter banks.
- Frequency bar and fields: The blue bar above the upper row of faders controls the upper and lower frequencies for both filter banks. Drag the bar to move both the upper and lower frequencies, drag the left end to move only the lower frequency (the value range is 75 to 750 Hz), or drag the right end to move only the upper frequency (the value range is 800 to 8000 Hz). You can also edit the numerical values above the bar directly (between 80 and 8000 Hz).
- Lowest button: Sets whether the lowest band of each filter bank acts as a lowpass or bandpass filter.

- *Highest button:* Sets whether the lowest band of each filter bank acts as a highpass or bandpass filter.
- Formant Stretch knob: Alters the width and distribution of the bands in the synthesis filter bank, extending or narrowing the frequency range defined by the blue bar (Low/High frequency parameters) for the synthesis filter bank.
 - If Formant Stretch is set to 0, the width and distribution of the bands in the synthesis filter bank is equal to the width of the bands in the analysis filter bank. Low values narrow the width of each band, while high values widen it. The control range is from 0.5 to 2 (as a ratio of the overall bandwidth).

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

- Formant Shift knob: Shifts the position of the bands in the synthesis filter bank up or down. When set to 0, the position of the bands in the synthesis filter bank is equal to the position of the bands in the analysis filter bank. Positive values will move the bands up in frequency, while negative values will move them down in respect to the analysis filter bank. You can jump directly to the values -0.5, -1, 0, +0.5 and +1 by clicking on their numbers.
- **Tip:** When combined, Formant Stretch and Formant Shift alter the formant structure of the resulting vocoder sound, and can result in some interesting timbre changes. As an example, using speech signals and tuning Formant Shift up results in Mickey Mouse effects. Formant Stretch and Formant Shift are especially useful if the frequency spectrum of the synthesis signal does not complement the frequency spectrum of the analysis signal. You could create a synthesis signal in the high frequency range from an analysis signal which mainly modulates the sound in a lower frequency range, for example.
 - Resonance knob: Controls the basic sonic character of both filter banks. Increasing the Resonance emphasizes the middle frequency of each band. Low settings give a softer character, while high settings give a sharper character.

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

LFO Section



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The parameters in this section control the LFO that can be used to modulate either the frequency (Pitch) of the tracking oscillator (vibrato), or the Formant Shift (Shift) parameter of the synthesis filter bank.

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

- Wave buttons: Select the waveform used by the LFO. A selection of Triangle, falling and rising Sawtooth, Square up and down around zero (bipolar, good for trills), Square up from zero (unipolar, good for changing between two definable pitches), a random stepped waveform (S&H), and a smoothed random waveform is available. Simply click on the appropriate button to select a waveform type.
- LFO Formant Shift Intensity slider: Controls the amount of format shift by the LFO.
- LFO Pitch Intensity: Controls the amount of modulation (vibrato) by the LFO.
- LFO Rate knob and field: Sets the speed of the LFO modulation. Values to the left of
 the center positions are synchronized with the sequencer's tempo and include bar
 values, triplet values and more. Values to the right of the center positions are nonsynchronous and displayed in Hertz (cycles per second).

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

U/V Detection Section

The U/V Detection section detects the unvoiced portions of the sound in the analysis signal, improving speech intelligibility. Please refer to "Unvoiced/Voiced (U/V) Detection" on page 221, for a full explanation of the U/V Detection principle. More information about improving speech intelligibility can be found in "Tips for Better Speech Intelligibility" on page 224.



- Sensitivity knob: Sets the degree of responsiveness of U/V detection. By turning this knob to the right, more of the individual *unvoiced* portions of the input signal are recognized.
 - When high settings are used, the increased sensitivity to unvoiced signals can lead to the U/V source—determined by the Mode parameter—being used on the majority of the input signal, including *voiced* signals. Sonically, this results in a sound that resembles a radio signal which is breaking up and contains a lot of static or noise.
- Mode pop-up menu: Choose the sound source(s) which can be used to replace the unvoiced content of the input signal. The choices are: Off, Noise, Noise + Synth, or Blend.
 - Noise: Uses noise alone for the unvoiced portions of the sound.
 - Noise + Synth: Uses noise and the synthesizer for the unvoiced portions of the sound,
 - Blend: Uses the analysis signal after it has passed through a highpass filter, for the unvoiced portions of the sound. This filtered analysis signal is then mixed with the EVOC 20 TrackOscillator output signal. The Sensitivity parameter has no effect on this setting.
- Level slider: Controls the amount of the signal (Noise, Noise + Synth, or Blend) used to replace the *unvoiced* content of the input signal.

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

Output Section



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- Signal pop-up menu: Choose the signal to send to the EVOC 20 TrackOscillator's main outputs. The choices are: Voc(oder), Syn(thesis), and Ana(lysis). To hear the vocoder effect, choose Voc. The other two settings are useful for monitoring purposes.
- Level slider: Sets the level of the output signal.
- Stereo Mode pop-up menu: Sets the input/output mode of the EVOC 20 Filterbank. The choices are m/s (mono input to stereo output), and s/s (stereo input to stereo output).

Set Stereo Mode to m/s if the input signal is mono, and to s/s if the input signal is stereo. In s/s mode, the left and right stereo channels are processed by separate filter banks. When using m/s mode on a stereo input signal, the signal is first summed to mono before it is passed to the filter banks.

- Stereo Width knob: Controls how the output signals of the filter bands are distributed in the stereo field.
 - At the left position, the output of all bands are centered.
 - At the centered position, the output of all bands ascends from left to right.
 - At the right position, the bands are output evenly on the left and the right channel.

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

Fuzz-Wah

The Fuzz-Wah emulates classic wah effects often used with a Clavinet, and adds compression and fuzz distortion effects as well.

Parameters of the Fuzz-Wah



• Effect Order buttons: Select whether the wah effect precedes the fuzz effect in the signal chain (Wah-Fuzz), or vice versa (Fuzz-Wah).

The integrated compressor always precedes the fuzz effect. When Wah-Fuzz is selected, the compressor comes between the wah and the fuzz effect; conversely, when Fuzz-Wah is selected, the compressor comes first in the signal chain.

Wah Section

- Wah Mode pop-up menu: Choose one of the six modes, which emulate various classic wah effects and filter types, or choose off.
- Auto Gain button: The wah effect can cause the output level to vary widely. Turning Auto Gain On compensates for this tendency, and keeps the output signal within a more stable range.
- Wah Level knob: Sets the amount of the wah-filtered signal.
- Relative Q slider: Adjusts the sharpness of the wah sweep by raising or lowering the filter peak. A setting of 0 (zero) retains the original peak level for each mode.
- *Pedal Range slider:* Sets the sweep range of the wah filter controlled using a MIDI foot pedal, letting you compensate for the difference in mechanical range between a MIDI foot pedal and a classic wah pedal.

You can set the upper and lower limits of the range independently by dragging the left and right edges of the Pedal Range slider, or move the entire range at once by dragging the center section of the slider.

AutoWah Section

- Depth knob: Sets the depth of the autowah effect.
- Attack knob: Sets the time it takes for the wah filter to fully open.
- Release knob: Sets the time it takes for the wah filter to close.

Fuzz Section

- Comp (Compression) Ratio knob: Sets the compression ratio of the integrated compressor.
- Fuzz Gain knob: Sets the level of distortion for the fuzz effect between 0 dB and 20 dB.
- Fuzz Tone knob: Adjusts the tone of the fuzz effect between 2 kHz and 20 kHz.

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

Using the Fuzz-Wah

The following sections cover various aspects of the Fuzz-Wah parameters.

Setting the Wah Level With Auto Gain

The wah effect can cause the output level to vary widely. Turning Auto Gain on compensates for this tendency, and keeps the output signal within a more stable range.

To hear the difference Auto Gain can make:

- 1 Switch Auto Gain to on.
- 2 Raise the effect level to a value just below the mixer's clipping limit.

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- 3 Make a sweep with a high relative Q setting.
- 4 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.

AutoWah Depth

In addition to using MIDI foot pedals (see above), the wah effect can be controlled using the Auto Wah facility. The sensitivity of the Auto Wah can be set with the *Depth* parameter. Range: 0.00 to 100. (See also "Envelope (Depth)" on page 374.)

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)

Setting the 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 (represents the current position of the Wah pedal). 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.

Spectral Gate

The Spectral Gate separates the signal above and below the Threshold level into two independent frequency ranges that you can modulate separately. It can produce some unusual and rich filtering effects.

Spectral Gate Parameters



- Threshold slider and field: Sets the threshold level at which the frequency band defined by the Center Freq. and Bandwidth parameters is divided into upper and lower frequency ranges.
- Speed slider and field: Sets the modulation frequency for the defined frequency band.
- *CF (Center Frequency) Modulation slider and field:* Sets the intensity of center frequency modulation.
- BW (Band Width) Modulation slider and field: Sets the amount of bandwidth modulation.
- *Graphic display:* Shows the frequency band defined by the Center Freq. and Bandwidth parameters.
- Center Freq. (Frequency) knob and field: Sets the center frequency of the frequency band to be processed by the Spectral Gate.
- Bandwidth knob and field: Sets the bandwidth of the frequency band to be processed by the Spectral Gate.
- Low Level slider and field: Blends the frequencies of the original signal below the selected frequency band with the processed signal.
- Super Energy knob and field: Controls the level of the frequency range above the threshold.
- Sub Energy and field: Controls the level of the frequency range below the threshold.
- *High Level slider and field:* Blends the frequencies of the original signal above the selected frequency band with the processed signal.
- Gain slider and field: Adjusts the amount of gain for the final output signal.

Using the Spectral Gate

Using the Center Freq. and Bandwidth parameters, set the frequency band you want to process using the Spectral Gate. The graphic display visually indicates the band defined by these two parameters.

Once the frequency band is defined, use the Threshold parameter to set the level above and below which the frequency band is divided into upper and lower ranges. Use the Super Energy knob to control the level of the frequencies above the Threshold, and use the Sub Energy knob to control the level of the frequencies below the Threshold.

You can also mix the frequencies from the original signal outside the frequency band defined by the Center Freq. and Bandwidth with the processed signal. Use the Low Level slider to blend the bass frequencies below the defined frequency band with the processed signal, and use the High Level slider to blend in frequencies above the defined frequency band.

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You can modulate the defined frequency band using the Speed, CF Modulation, and BW Modulation parameters. Speed determines the modulation frequency, CF (Center Frequency) Modulation defines the intensity of the center frequency modulation, and BW (Band Width) Modulation controls the bandwidth modulation.

After making your adjustments, you can use the Gain slider to adjust the final output level of the processed signal.

One way to get better acquainted with the operation of the Spectral Gate would be to start with a drum loop. Set the Center Freq. to its minimum (20 Hz) and the Bandwidth to its maximum (20000 Hz) value (so that the entire frequency range is processed). Turn up the Super Energy and Sub Energy knobs, one at a time, then try different Threshold settings. This should give you a good sense of how different Threshold levels affect the sound of Super Energy and Sub Energy. When you come across a sound that you like or consider useful, narrow the Bandwidth drastically, gradually increase the Center Freq., and then use the Low Level and High Level sliders to mix in some treble and bass from the original signal. At lower Speed settings, turn up the CF Mod. or BW Mod. knobs.

Soundtrack Pro Autofilter

The Soundtrack Pro Autofilter is a simple lowpass resonance filter that offers the following parameters:



- Cutoff Frequency slider and field: Sets the cutoff frequency for the lowpass filter.
- Resonance slider and field: Adjusts the amount of emphasis in the frequency band around the cutoff frequency.

Imaging 7

You can use the Logic Studio Imaging plug-ins to extend the stereo base of a recording, and to alter perceived signal positions.

These effects enable you to make certain sounds, or the overall mix, seem wider and more spacious. You can also alter the phase of individual sounds within a mix, to enhance or suppress particular transients.

The following sections describe the Imaging plug-ins included with Logic Studio:

- "Binaural Post-Processing Plug-in" on page 121.
- "Direction Mixer" on page 123.
- "Stereo Spread" on page 125.

Binaural Post-Processing Plug-in

Logic Pro allows you to route channel outputs through the Binaural Panner, a psychoacoustic processor that is able to simulate arbitrary sound source positions (including up and down information) by means of a standard stereo signal. The signal that results from Binaural Panner use is best suited for headphone playback. You can, however, use the integrated conditioning of the Binaural Panner to ensure a neutral sound for speaker (or headphone) playback.

When using multiple Binaural Panners (on several channels), you should turn the integrated conditioning off, and route the output of all binaurally panned signals to an aux channel—in which you should insert the Binaural Post-Processing plug-in. The Binaural Post Processing plug-in allows you to apply diffuse-field compensation to all Binaural Panner outputs at once, saving CPU power.

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The Binaural Post-Processing plug-in is available in aux and output channels.



The Binaural Post-Processing plug-in offers the following Compensation choices:

- Headphone FF optimized for front direction: Setting for headphone playback, utilizing free-field compensation. In this compensation mode, sound sources placed in front of the listening position will have neutral sound characteristics.
- *Headphone HB optimized for horizontal directions:* Setting for headphone playback. Optimized to deliver the most neutral sound for sources placed on, or close to, the horizontal plane.
- Headphone DF averaged over all directions: Setting for headphone playback, utilizing diffuse-field compensation. In this compensation mode, the sound will (on average) be most neutral for arbitrarily placed, or moved, sources.
- Speaker CTC Cross Talk Cancellation: Setting for speaker playback, which allows you to play back binaurally panned signals via a stereo loudspeaker setup. Good spatial reproduction is restricted to a limited range of listening positions (on the symmetrical plane) between the speakers. If you choose the Speaker CTC setting there is an additional parameter: CTC Speaker Angle. To achieve the best binaural effect, enter the angle that your stereo speakers are turned towards the center (listening position).

For full details on using the Binaural Panner with the Binaural Post-Processing plug-in, see the *Logic Pro 8 User Manual*.

Direction Mixer

You can use the Direction Mixer plug-in to decode middle and side (MS) audio recordings (see "What Is MS?" on page 124), or to spread the stereo base of a (left/right) recording, and determine its pan position.



- Input buttons: Use the LR or MS buttons to determine whether the input signal is a standard left/right signal, or if you're dealing with an MS encoded (middle and side) signal.
- Spread slider and field: Determines the spread of the stereo base.
- *Direction knob and field:* Determines the direction from which the middle of the recorded stereo signal will emanate from within the mix, or in less complicated terms, its pan position.

Using the Direction Mixer

The Direction Mixer is a simple plug-in to use, as it only offers two parameters: Spread and Direction. Each alters the incoming signal differently when either the LR or MS Input buttons are active.

Using the Spread Parameter on LR Input Signals

At a neutral value of 1, the left side of the signal is positioned precisely on the left, and the right side precisely on the right. As you decrease the Spread value, the two sides move towards the center of the stereo image. A value of 0 produces a mono signal (both sides of the input signal are routed to the two outputs at the same level—a true middle signal). At values greater than 1, the stereo base is extended out to an imaginary point beyond the spatial limits of the speakers.

Note: If simply using the Direction Mixer to spread the stereo base, monaural compatibility decreases with Spread values above 1. Once a stereo signal has been processed at an extreme Spread setting of 2, the signal will be canceled out completely if played back in mono—after all, L–R plus R–L doesn't leave you with much.

Using the Spread Parameter on MS Input Signals

When you alter MS levels with the Spread parameter (above a value of 1), the level of the side signal becomes higher than that of the middle signal. At a value of 2, you will only hear the side signal (on the left, you'll hear L–R and on the right, R–L).

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Setting the Direction Parameter

When Direction is set to a value of 0, the middle of the stereo recording will be dead center within the mix. If you use positive values, the midpoint of the stereo recording is moved towards the left. Negative values move the midpoint to the right. Here's how this works:

- At 90°, the midpoint of the stereo recording is panned hard left.
- At –90°, the midpoint of the stereo recording is panned hard right.
- Higher values move the midpoint back towards the center of the stereo mix, but this
 also has the effect of swapping the stereo sides of the recording. To explain: At
 values of 180° or –180°, the midpoint of the recording is dead center in the mix, but
 the left and right sides of the recording are swapped.

What Is MS?

Relegated to obscurity for a good long while, MS stereo (middle-side as opposed to left-right) has recently enjoyed a renaissance of sorts.

Making a Middle Side Recording

Two microphones are positioned as closely together as possible (usually on a stand or hung from the studio ceiling). One is a cardioid (or omnidirectional) microphone which directly faces the sound source that you want to record—in a straight alignment. The other is a bidirectional microphone, with its axes pointing to the left and right (of the sound source) at 90° angles.

- The cardioid microphone records the middle signal to the left side of a stereo track.
- The bidirectional microphone records the side signal to the right side of a stereo track.

MS recordings made in this way can be decoded by the Direction Mixer.

Why Make MS Recordings?

The advantage that MS recordings have over XY recordings (two cardioid microphones that are directed to a point halfway to the left and right of the sound source) is that the stereo middle is actually located on the on-axis (main recording direction) of the cardioid microphone. This means that slight fluctuations in frequency response that occur off the on-axis—as is the case with every microphone—are less troublesome.

In principle, MS and LR signals are equivalent, and can be converted at any time. When "-" signifies a phase inversion, then the following applies:

$$M = L + R$$
$$S = I - R$$

In addition, L can also be derived from the sum of—and R, from the difference between—M and S.

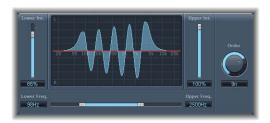
Here's some interesting trivia for you: Radio (FM) broadcasts feature M and S stereo. The MS signal is actually converted to a signal suitable for the left and right speakers by the receiver.

Stereo Spread

The Stereo Spread effect is typically used for mastering. There are several ways to extend the stereo base (or perception of space), including use of reverbs and other effects, and altering the signal's phase. They can all sound great, but can also weaken the overall sound of your mix by ruining transient responses, for example.

The Stereo Spread plug-in extends the stereo base by distributing a (selectable) number of frequency bands from the middle frequency range to the left and right channels. This is done alternately—middle frequencies to the left channel, middle frequencies to the right channel, and so on. This greatly increases the perception of stereo width without making the sound totally unnatural, especially when used on mono recordings.

Stereo Spread Parameters



- Lower Int. slider and field: Sets the amount of stereo base extension for the lower frequency bands.
- *Upper Int. slider and field:* Sets the amount of stereo base extension for the upper frequency bands.
 - A point to note when you are setting the Lower and Upper Int. sliders is that the stereo effect is most apparent in the middle and higher frequencies, and that distributing low frequencies between the left and right speakers significantly reduces the energy from both speakers. For this reason, you should use a lower Lower Int. setting, and avoid setting the Lower Freq. below 300 Hz.
- *Graphic display:* Shows the number of bands the signal is divided into, and the intensity of the Stereo Spread effect in the upper and lower frequency bands. The upper section represents the left channel, and the lower section represents the right channel. The frequency scale displays frequencies in ascending order, from left to right.

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- *Upper and Lower Freq. slider and fields:* Use these to determine the upper and lower limits of the highest frequency, and lowest band, to be distributed in the stereo image.
- Order knob: Sets the number of frequency bands that the signal is divided into. A value of 8n is usually sufficient for most tasks, but you can use up to 12 bands.

Metering 8

You can use the Metering plug-ins of Logic Studio to analyze audio in a variety of ways.

Each Metering plug-in allows you to view different characteristics of an audio signal. As examples: The BPM Counter displays the tempo of an audio file, the Correlation Meter displays the phase relationship, and the Level Meter displays the level of an audio recording.

This chapter describes the Metering plug-ins included with Logic Studio:

- "BPM Counter" on page 128
- "Correlation Meter" on page 128
- "Level Meter" on page 129
- "MultiMeter" on page 129
- "Surround MultiMeter" on page 133
- "Tuner" on page 134

BPM Counter



You can use the BPM Counter to analyze the tempo of an audio track. Insert the plug-in into a track, to analyze the dynamic events of the audio signal.

The detection circuit looks for any transients in the input signal. Transients are very fast, non-periodic sound events in the attack portion of the signal. The more obvious this impulse is, the easier it is for the BPM Counter to detect the tempo. As a result, percussive drum and instrumental rhythm tracks (basslines, for example) are very well suited for tempo analysis. Pad sounds are a poor choice.

The LED shows the current analysis status. If the LED is flashing, a tempo measurement is taking place. When lit, analysis is complete, and the tempo is displayed. The measurement ranges from 80 to 160 beats per minute. The measured value is displayed with an accuracy of one decimal place.

The BPM Counter also detects any tempo variation in the signal, and tries to analyze it/them accurately. If the LED starts flashing during playback, this indicates that the BPM Counter has detected a tempo that has deviated from the last received (or set) tempo. As soon as a new, constant tempo is recognized, the LED will be solidly lit.

Click the LED to reset the BPM Counter.

Correlation Meter



The Correlation Meter displays the phase relationship of a stereo signal.

- A correlation of +1 (plus one, the far right position) means that the left and right channels correlate 100% (they are completely in-phase).
- A correlation of 0 (zero, the center position) indicates the widest permissible left/right divergence, often audible as an extremely wide stereo effect.
- Correlation values lower than zero indicate that out-of-phase material is present, which can lead to phase cancelations if the stereo signal is combined into a monaural signal.

Level Meter



The Level Meter displays the current signal level on a decibel scale. The signal level for each channel is represented by a blue bar. When the level exceeds 0 dB, the portion of the bar above the 0 dB point becomes red. Stereo instances of the Level Meter show independent left and right bars, while mono instances display only a single bar. Surround instances display a bar for each channel, in a vertical rather than horizontal orientation.

The current peak values are displayed numerically, superimposed over the graphic display. You can reset these values by clicking in the display.

The Level Meter can be set to display levels using Peak, RMS, or Peak & RMS characteristics. Simply choose the desired setting in the pop-up menu (below the graphic display). RMS levels appear as dark blue bars. Peak levels appear as light blue bars. You can also choose to view both Peak and RMS levels simultaneously.

MultiMeter

The MultiMeter provides a collection of professional gauge and analysis tools in a single window. It includes:

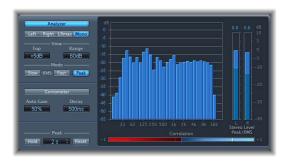
- An Analyzer to view the level of each 1/3-octave frequency band.
- A Goniometer for judging the phase coherency in the stereo sound field.
- a Correlation Meter to spot mono phase compatibility.
- An integrated Level Meter to view the signal level for each channel.

There is also a surround version of the MultiMeter, with parameters for each channel and a slightly different layout. For more information on the Surround MultiMeter, see "Surround MultiMeter" on page 133.

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MultiMeter Parameters

You can view either the Analyzer or Goniometer in the main display area. You switch the view and set other MultiMeter parameters using the controls on the left side of the window. The Phase Correlation meter is always visible at the bottom of the window, and the Level Meters are visible on the right.



Analyzer Section

- Analyzer button: When selected, displays the Analyzer in the center of the window.
- Left, Right, LRmax, and Mono buttons: Sets which channels are displayed in the
 Analyzer. Selecting Left or Right display only those channels, respectively, selecting
 LRMax displays the maximum level of the stereo inputs, and selecting Mono displays
 the spectrum of the mono sum of both (stereo) inputs.
- *Mode buttons:* Click one of the buttons to set whether levels are displayed using Peak, Slow RMS, or Fast RMS characteristics.

Goniometer Section

- Goniometer button: When selected, displays the Goniometer in the center of the window.
- Auto Gain field (display only): Sets the amount by which the display compensates for low input levels. You can set Auto Gain levels in 10% increments, or set it to off.
- *Decay field*: Sets the amount of time it takes for the Goniometer trace to fade to black.

Peak Section

- *Hold button:* When selected, activates peak hold for the all of the metering tools in the MultiMeter. Peak hold has the following effects:
 - In the Analyzer, a small yellow segment above each 1/3 octave level bar labels the most recent peak level.
 - In the Goniometer, all illuminated pixels of the display are held in place during peak hold.

- In the Correlation Meter, a growing horizontal area around the white correlation indicator shows any phase correlation deviations—in both directions. A vertical red line to the left of the correlation indicator permanently shows the maximum negative phase deviation value. You can reset this line by clicking on it during playback.
- In the Level Meter, a small yellow segment above each stereo level bar labels the most recent peak level.
- Hold Time pop-up menu: When peak hold is active, sets the hold time for all metering tools to either 2 s, 4 s, 5 s, 6 s or infinite.
- Reset button: Click to reset the peak hold segments of all metering tools.

Graphic Display

Displays either the Analyzer or Goniometer in the center of the window. The Correlation Meter always appears in the bottom of the window, and the Level Meters always appear on the right side of the window.

Using the MultiMeter

The following section provides information on using the different meters in the MultiMeter.

Analyzer

The Analyzer displays the frequency spectrum of the input signal in 31 independent frequency bands. Each frequency band represents one-third of an octave.

You can display only part of the input signal using one of the channel buttons. Select either the Left or Right button to view these channels independently, select LRmax to see the maximum band levels of either channel, or select Mono to view the levels of the stereo signal as a summed mono entry.

You can alter the scale of values displayed in the Analyzer in several ways. Use the View parameters, which let you set the maximum level displayed and the overall dynamic range, by vertically dragging the dB scale on the left edge of the Analyzer. Adjusting the scale is useful when analyzing highly compressed material, so that you can identify smaller level differences more easily by moving and/or reducing the display range. You can also change the range (the minimum and maximum values displayed) by vertically dragging in the Analyzer display area, and adjust the maximum value shown by dragging vertically in the display area.

The two RMS modes with Slow and Fast response settings show the effective signal average, and provide a representative overview of the perceived volume levels. The Peak mode shows level peaks accurately.

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Goniometer

A Goniometer helps you to judge the coherence of the stereo image and determine phase differences between the left and right channels. Phase problems are easily spotted as trace cancelations along the center line (M—mid/mono).

The idea of the Goniometer was born with the advent of early two-channel oscilloscopes. To use such devices as Goniometers, users would connect the left and the right stereo channels to the X and Y inputs, while rotating the display by 45 degrees to produce a useful visualization of the signal's stereo phase. The signal trace slowly fades to black, imitating the retro glow of the tubes found in older Goniometers, while also enhancing the readability of the display.

In the MultiMeter's Goniometer, the Auto Gain parameter lets you obtain a higher readout on low-level passages, by allowing the display to automatically compensate for low input levels. You can set the amount of compensation in 10% increments. Remember that, for the MultiMeter, Auto Gain is a display parameter only, and increases display levels in order to enhance readability. It does not change the actual audio levels.

Correlation Meter

The Correlation Meter provides an additional phase measurement tool that gauges the phase relationship of a stereo signal. The Correlation Meter's scale ranges from -1 to +1, and different values provide the following indications:

- A +1 correlation value indicates that the left and right channels correlate 100%. In other words, the left and right signals are in phase and are the same shape.
- Correlation values in the blue zone (between +1 and the middle position) indicate that the stereo signal is mono compatible.
- The middle position indicates the highest permissible amount of left/right divergence, which is often audible as an extremely wide stereo effect.
- Once the correlation meter moves into the red area to the left of the center position, out-of-phase material is present. This will lead to phase cancelations if the stereo signal is combined into a mono signal.

Level Meter (Peak/RMS Meter)

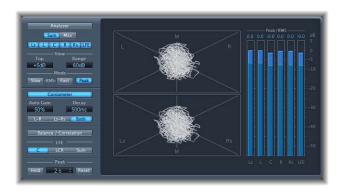
The Level Meter displays the current signal level on a logarithmic decibel scale. The signal level for each channel is represented by a blue bar. RMS and Peak levels are shown simultaneously, with RMS levels appearing as dark blue bars, and Peak levels appearing as light blue bars. When the level exceeds 0 dB, the portion of the bar above the 0 dB point becomes red.

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

Surround MultiMeter

The surround version of the MultiMeter includes some additional parameters for use in analyzing multichannel surround files.

Surround MultiMeter Parameters



Analyzer Section

These parameters are similar to those in the stereo MultiMeter, but include buttons for each surround channel. You can select a single channel or a combination of channels. When analyzing a combination of channels, you can set the Analyzer to display either the maximum or sum of the selected channels using the Sum and Max buttons.

Goniometer Section

These parameters are similar to those in the stereo MultiMeter, but include buttons for each channel pair (L–R, Lm–Rm, Ls–Rs), or for all channel pairs (both). When using the Surround MultiMeter in configurations with exactly two channel pairs (quad, 5.1, and 6.1 configurations), the Goniometer can optionally display both pairs by selecting Both. One pair (for L–R) appears in the upper center of the graphic display, and one (for Ls–Rs) appears in the lower center.

Balance/Correlation Section

In the Surround MultiMeter, the Correlation Meter is not visible when the Analyzer or Goniometer are active, but is displayed separately when you select the Balance/Correlation button. The highlighted area indicates the overall balance of the surround signal.

The Balance/Correlation Meter combines two meters into one compact, easy-to-read display. It is based on the balance meter, showing the sound placement, while also incorporating a correlation display which shows strongly correlated signals as a sharp marker, less strongly correlated signals as a blurred area, and negative correlations as red.

Peak (Level Meter) Section

These parameters are the same as those in the stereo MultiMeter.

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Tuner

You can tune both acoustic and electric music instruments connected to your system using the Tuner. Tuning your instruments ensures that your recordings will be in tune with any software instruments, existing samples, or existing recordings in your projects.

Tuner Parameters



- Graphic tuning display: As you play, the pitch of the note appears in the semicircular area, centered around the Keynote. If the highlight bar moves to the left of center, the note is flat; if the highlight bar moves to the right of center, the note is sharp. The numbers around the edge of the display show the variance, in cents, from the target pitch.
- Keynote/Octave display: The upper Keynote area shows the target pitch of the note you play (the closest pitch in tune). The lower Octave area indicates which octave the note belongs to. This matches the MIDI octave scale, with the C above middle C displayed as C4, and middle C displayed as C3.
- Tuning Adjustment knob and field: Sets the pitch of the note used as the basis for tuning. By default, the Tuner is set to concert pitch A = 440 Hz. Drag the knob left to lower the pitch corresponding to A, or drag the knob right to raise the pitch corresponding to A. The current value is displayed in the field.

Using the Tuner

Using the Tuner is simple. With your instrument (or microphone capturing the sound of an acoustic instrument) connected to the channel with the Tuner, play a single note and watch the display. If the note is flat of the Keynote, the segments left of center light, showing how far (in cents) the note is off pitch. If the note is sharp, the midpoint segments right of center light. Adjust the tuning of your instrument until the center segment lights (red).

On the tuning display, the range is marked in single semitone steps ± 6 cents close to the center, and then in larger increments to a maximum of ± 50 cents.

Modulation

Modulation effects are used to add motion and depth to your sound.

Modulation effects include chorus, flanging, and phasing among others, which make sounds richer or more animated. This is often achieved through the use of an LFO, which is controlled with parameters such as speed or frequency, and depth (also called width, amount, or intensity). You can also control the ratio of the affected (wet) signal and the original (dry) signal. Some modulation effects include feedback parameters, which add part of the effect's output back into the effect input.

Logic Studio includes the following modulation effects:

- "Chorus" on page 136
- "Ensemble" on page 136
- "Flanger" on page 137
- "Microphaser" on page 138
- "Modulation Delay" on page 138
- "Phaser" on page 140
- "Ringshifter" on page 142
- "Rotor Cabinet" on page 147
- "Scanner Vibrato" on page 148
- "Spreader" on page 149
- "Tremolo" on page 150

Chorus

The Chorus effect delays the original signal. The delay time is modulated with an LFO. The delayed, modulated signal is mixed with the original, dry signal.

You can use the Chorus effect to enrich the sound and create the impression that it's being played by multiple instruments or voices, in unison. The slight delay time variations generated by the LFO simulate the subtle pitch and timing differences heard when several people perform together. Using chorus also adds fullness or richness to the signal, and can add movement to low or sustained sounds.



- Intensity slider and field: Defines the modulation amount.
- Rate knob and field: Defines the frequency, and therefore the speed, of the LFO.
- Mix slider and field: Determines the balance of dry and wet signals.

Ensemble

The Ensemble combines up to eight chorus effects. Two standard LFOs and one random LFO (which generates random modulations) enable you to create complex modulations. The Ensemble's graphic visually represents the processed signals.



- Voices slider and field: Determines how many individual chorus instances are used, and therefore how many voices (or signals) are generated, in addition to the original signal.
- Rate knobs and fields: Use the respective knob to control the frequency of each LFO.
- Intensity sliders and fields: Use these to set the amount of modulation for each LFO.

- Phase knob and field: Controls the phase relationship between the individual voice
 modulations. The value that you choose here is dependent on the number of voices,
 which is why it is shown as a percentage value rather than degrees. The value 100 (or
 –100) is equal to the greatest possible distance between the modulation phases of all
 voices.
- Spread slider and field: Used to distribute the voices across the stereo or surround field. When you set a value of 200%, the stereo or surround base is expanded artificially. Please note that monaural compatibility may suffer if you choose to do this.
- Mix slider and field: Determines the balance between dry and wet signals.
- Effect Volume knob and field: Use this to independently determine the level of the effects signal. This is a useful tool which compensates for changes in volume caused by changes to the Voices parameter.

Using the Ensemble Effect in Surround

The Ensemble effect converts the input signal to mono before processing it. Therefore, the Ensemble effect is inserted as multi channel effect in surround to surround channels. To access the Surround parameters of the Ensemble effect, you need to insert the Ensemble as a Mono to Surround or Stereo to Surround effect instance.

Flanger

The Flanger effect works in much the same way as the Chorus effect, but uses a significantly shorter delay time. In addition, the effect signal can be fed back into the input of the delay line.

Flanging is typically used to create changes sometimes described as adding a spacey or underwater effect.



- Rate knob and field: Defines the frequency, and therefore the speed, of the LFO.
- Intensity slider and field: Determines the modulation amount.
- Feedback slider and field: Determines the amount of the effect signal that is routed back into the input. Negative values invert the phase of the routed signal.
- Mix slider and field: Determines the balance between dry and wet signals.

Microphaser

The Microphaser is a simple phaser effect that allows you to quickly create swooshing, phasing effects with just three parameters:

- LFO Rate slider and field: Defines the frequency, and therefore the speed, of the LFO.
- Feedback slider and field: Determines the amount of the effect signal that is routed back into the input.
- Intensity slider and field: Determines the amount of modulation.

Modulation Delay

The Modulation Delay is based on the same principles as the Flanger and Chorus effects, but you can set the delay time, allowing both chorus and flanging effects to be generated. It can also be used—without modulation—to create resonator or doubling effects. The modulation section consists of two LFOs with variable frequencies.



- Feedback slider and field: Determines the amount of the effect signal that is routed back 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.
- Flanger-Chorus knob and field: Sets the basic delay time. Set to the far left position to create flanger effects, to the center for chorus effects, and to the far right to hear clearly discernible delays.
- Intensity slider and field: Sets the modulation amount.
- *De-Warble button:* Switch on to ensure that the pitch of the modulated signal remains constant.
- Constant Mod. (Constant Modulation) button: Switch on to ensure that the modulation width sounds constant, regardless of the modulation rate. Note that when switched on, higher modulation frequencies will reduce the modulation width.
- LFO Mix slider and fields: Determines the balance between the two LFOs.

- LFO 1 and LFO 2 Rate knobs and fields: Use the left knob to set the modulation rate for the left stereo channel, and the right knob to set the modulation rate of the right stereo channel. The right LFO Rate knob is only available in stereo and surround instances, and can only be set separately if the Left Right Link button is not enabled. In Surround instances, the center channel is assigned the middle value of the LFO left and LFO right Rate knobs. The other channels are assigned values between the left and right LFO rates.
- LFO Left Right Link button (only available in stereo and surround instances): Switch on to tie the modulation rates of the left and right stereo channels to each other.
- LFO Phase knob and field (only available in stereo and surround instances): Controls the phase relationship between the individual channel modulations. At 0°, the extreme values of the modulation are achieved simultaneously for all channels. 180° or –180° is equal to the greatest possible distance between the modulation phases of the channels. The LFO Phase parameter is only available, if the LFO Left Right Link button is active.
- Distribution menu (only available in surround instances): Defines how the phase offsets between the individual channels are distributed in the surround field. You can choose from Circular, Random, Front <> Rear, and Left <> Right distribution. When you load a setting that uses the Random option, the saved phase offset value is recalled. If you want to randomize the phase setting again, choose "new random" in the Distribution menu.
- *Vol.Mod.* (*Volume Modulation*) *slider and field:* Use this to determine how much of an impact the LFO modulation has on the amplitude of the effect signal.
- Output Mix slider and field: Determines the balance between dry and wet signals.
- True Analog menu (extended parameter): Use to introduce an additional all-pass filter into the signal path. An all-pass filter shifts the phase angle of a signal, influencing its stereo image.
- Analog Left and Analog Right sliders and fields (extended parameter): This controls the
 frequency at which the phase shift crosses 90° (the half-way point of the total 180°)
 for each of the stereo channels. In Surround instances, the other channels are
 assigned values between the two settings.

Phaser

The Phaser effect combines the original signal with a copy that is slightly out of phase with the original. This means that the amplitude of the two signals reach their highest and lowest points at slightly different times. The timing differences between the two signals is modulated by two independent LFOs.

In addition, the Phaser includes a filter circuit and a built-in envelope follower, which tracks any volume changes in the input signal, generating a dynamic control signal.



- Filter button: Click to activate the filter section, which processes the feedback signal of the Pitch Shifter.
- LP and HP knobs and fields: Use these to set the cutoff frequency of the filter section's highpass and lowpass filters.
- Feedback slider and field: Determines the amount of the effect signal that is routed back into the input of the effect.
- Ceiling and Floor slider and fields: Use the individual slider handles to determine the frequency range that will be affected by the LFO modulations.
- Order slider and field: Allows you to choose between different phaser algorithms. The more orders a phaser has, the heavier the effect.
- Env Follow slider and field (Sweep section): Determines how much the frequency range (as set with the Ceiling and Floor controls) is modulated by the level of the input signal.
- LFO 1 and LFO 2 Rate knobs and fields: Use to set the speed for each LFO independently.
- LFO Mix slider and fields: Determines the balance between the two LFOs.
- Env Follow slider and field (LFO section): Use this to set how much the speed of LFO 1 is modulated by the level of the input signal.
- Phase knob and field (only available in stereo and surround instances): Controls the phase relationship between the individual channel modulations. At 0°, the extreme values of the modulation are achieved simultaneously for all channels. 180° or –180° is equal to the greatest possible distance between the modulation phases of the channels.

- Distribution menu (only available in surround instances): Defines how the phase offsets between the individual channels are distributed in the surround field. You can choose from Circular, Random, Front <> Rear, and Left <> Right distribution. When you load a setting that uses the Random option, the saved phase offset value is recalled. If you want to randomize the phase setting again, choose "new random" in the Distribution menu.
- Output Mix slider and field: Determines the balance of dry and wet signals. Negative values result in a phase inverted mix of the effect and direct (dry) signal.
- *Warmth button:* Click to switch on an additional distortion circuit, which allows the creation of warm overdrive effects.

Setting the Phaser Orders

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. 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.

Ringshifter

The Ringshifter effect combines a ring modulator with a frequency shifter effect. Both effects were popular during the 1970s, and are currently experiencing something of a renaissance.

- The ring modulator modulates the amplitude of the input signal using either the
 internal oscillator or a side chain signal. The frequency spectrum of the resulting
 effect signal equals the sum and difference of the frequency content in the two
 original signals. Its sound is often described as metallic or clangorous. The ring
 modulator was used extensively on jazz rock and fusion records in the early 70s.
- The frequency shifter moves the frequency content of the input signal by a fixed amount and, in doing so, alters the frequency relationship of the original harmonics.
 The resulting sounds range between sweet and spacious phasing effects to strange robotic timbres. Frequency shifting should not be confused with pitch shifting. Pitch shifting transposes the original signal, leaving its harmonic frequency relationship intact.



The Ringshifter consists of the following parameter groups:

- *Mode buttons*: Determine whether the Ringshifter operates as frequency shifter or ring modulator.
- Oscillator parameters: Use these to configure the internal sine wave oscillator, which
 modulates the amplitude of the input signal—in both frequency shifter modes and
 the ring modulator OSC mode.
- Envelope follower and LFO parameters: The oscillator frequency and output signal can be modulated with an envelope follower and LFO.
- Delay parameters: Use these to delay the effect signal.
- Output parameters: The output section of the Ringshifter includes a feedback loop and controls to set the stereo width and amount of the dry and wet signals.

Modes

The four mode buttons determine whether the Ringshifter operates as a frequency shifter or as a ring modulator.



- Single (Frequency Shifter) button: The frequency shifter generates a single, shifted effect signal. The oscillator Frequency control determines whether the signal is shifted up (positive value) or down (negative value).
- Dual (Frequency Shifter) button: The frequency shifting process produces one shifted effect signal for each stereo channel—one is shifted up, the other is shifted down. The oscillator Frequency control determines the shift direction in the left versus the right channel.
- OSC (Ring Modulator) button: The ring modulator uses the internal sine wave oscillator to modulate the input signal.
- Side Chain (Ring Modulator) button: The ring modulator modulates the amplitude of the input signal with the audio signal assigned via the side chain input. The sine wave oscillator is switched off, and the Frequency controls are not accessible when Side Chain mode is active.

The Oscillator

In both frequency shifter modes, and the ring modulator OSC mode, the internal sine wave oscillator is used to modulate the amplitude of the input signal. The Frequency control sets the frequency of the sine wave oscillator.

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



- Frequency control: Sets the frequency of the sine oscillator.
- Lin(ear) and Exp(onential) buttons: Use these buttons to switch the scaling of the Frequency control:
 - The exponential scaling offers extremely small increments around the 0 point, which is useful for programming slow moving phasing and tremolo effects.
 - In the Lin(ear) mode, the resolution of the scale is even across the entire control range.
- Env Follower slider and field: Use to determine how much the oscillator is modulated by the level of the input signal.
- LFO slider and field: Use to determine the amount of oscillator modulation by the LFO.

Delay

The effect signal is routed through a delay, following the oscillator.



- Time knob and field: Sets the delay time.
- *Sync button:* Turn this on to synchronize the delay to your project tempo, in musical note values.
- Level knob and field: Sets the level of the delay added to the ring modulated or frequency shifted signal. A Level value of 0 passes the effect signal directly to the output (bypass).

Output



- Feedback knob and field: Sets the amount of the signal that is routed back to the effect input.
- Stereo Width knob and field: Determines the breadth of the effect signal in the stereo field. Stereo Width only affects the effect signal of the Ringshifter, not the dry input signal.
- *Dry/Wet knob and field:* Set the mix ratio of the dry input signal and the wet effect signal.
- Env Follower slider and field: Use to determine how much the Dry/Wet parameter is modulated by the level of the input signal.
- LFO slider and field: Sets the modulation depth of the Dry/Wet parameter by the LFO.

Setting the Feedback

Feedback gives the Ringshifter sound an additional edge, and is useful for a variety of special effects. It produces a rich phasing sound when used in combination with a slow oscillator sweep. Comb filtering effects are created by using high Feedback settings with a short delay time (< 10 ms). Using longer delay times, in conjunction with Feedback, creates spiralling, continuously rising and falling frequency shift effects.

Modulation Sources

The oscillator Frequency and Dry/Wet parameters can be modulated via the internal envelope follower and LFO. The oscillator frequency even allows modulation through the 0 Hz point, thus changing the oscillation direction.

Envelope Follower

The envelope follower analyzes the amplitude (volume) of the input signal and uses this to create a continuously changing control signal—a dynamic volume envelope of the input signal. This control signal can be used for modulation purposes.



- Power button: Turns the envelope follower on or off.
- Sens(itivity) slider and field: Determines how responsive the envelope follower is to the input signal. At lower settings, the envelope follower will only react to the most dominant signal peaks. At higher settings, the envelope follower will track the signal more closely, but may react less dynamically.
- Attack slider and field: Sets the response time of the envelope follower.
- *Decay slider and field:* Controls the time it takes the envelope follower to return from a higher to a lower value.

LFO

The LFO is the second modulation source. The LFO produces continuous, cycled control signals.



- Power button: Turns the LFO on or off.
- Symmetry and Smooth sliders and fields: These controls shape the LFO waveform. The LFO waveform display provides visual feedback.
- Rate knob and field: Sets the cycle speed of the LFO.
- *Sync button:* Turn this on to synchronize the LFO cycles (LFO rate) with the project tempo, using musical note values.

Rotor Cabinet

The Rotor Cabinet effect emulates the rotating loudspeaker cabinet of a Hammond organ's Leslie effect. It simulates both the rotating speaker cabinet, with and without deflectors, and the microphones which pick up the sound.



- *Rotor speed buttons:* These switch the rotor speed. Chorale switches to slow movement, Tremolo to fast movement, and Brake stops the rotor.
- Cabinet Type menu: Choose between various cabinet sizes, shapes, and material.
- Horn Deflector slider: Switches the deflector of the rotor cabinet's double horn on or off. Switching the deflector off increases amplitude modulation, and decreases frequency modulation.
- *Motor Control menu:* Allows you to choose between various modes that define the speed of the bass and treble controls separately.
- Rotor Fast Rate slider: Use this to set the maximum possible rotor speed for the Tremolo mode. While moving the slider, the Tremolo (rotation) speed is displayed in Hertz.
- Acc/Dec Scale slider: Determines the speed at which the Leslie motor accelerates the
 rotors (time it takes to get the rotors up to a determined speed), and the length of
 time it takes for them to slow down.
- *Mic Angle slider:* Changes the angle of the simulated microphones. An angle of 0° results in a mono sound, while an angle of 180° causes phase cancellations.
- *Mic Distance slider:* Sets the distance of the simulated microphones from the sound source. Higher values result in a darker, and less defined sound.

Choosing the Cabinet Type

You can choose between the following cabinet types in the Cabinet Type menu:

- Off: Choose to switch off the rotor. There's an alternative to switching the rotor effect off: in the Brake mode, the speakers don't rotate, but are still picked up by the simulated microphones, in a random position (see "Rotor Speed" paragraph below).
- *Wood:* Mimics a Leslie with a wooden enclosure, and sounds like the Leslie 122 or 147 models.
- Proline: Mimics a Leslie with a more open enclosure, similar to a Leslie 760 model.
- Single: Simulates the sound of a Leslie with a single, full-range, rotor. The sound resembles the Leslie 825 model.
- *Split*: The bass rotor's signal is routed more to the left side, and the treble rotor's signal is routed more to the right side.

- Wood & Horn IR: This setting uses an impulse response (a recording) of a Leslie with a
 wooden enclosure.
- *Proline & Horn IR*: This setting uses an impulse response of a Leslie with a more open enclosure.
- Split & Horn IR: This setting uses an impulse response of a Leslie with the bass rotor signal routed more to the left side, and the treble rotor signal routed more to the right side.

Choosing a Motor Control Mode

In the Motor Control menu, you can set different speeds for the bass and treble rotors:

- Normal: Both rotors use the speed set by the rotor speed buttons.
- *Inv (inverse mode):* In Tremolo mode, the bass compartment rotates at a fast speed, while the horn compartment rotates slowly. This is reversed in Chorale mode. In Brake mode, both rotors will stop.
- 910: The 910, or Memphis mode, stops the bass drum rotation at slow speed, while the speed of the horn compartment can be switched. This may be desirable, if you're after a solid bass sound, but still want treble movement.
- *Sync:* The acceleration and deceleration of the horn and bass drum are roughly the same. This sounds as if the two are synchronized, but is only clearly audible during acceleration or deceleration.

Note: If you choose Single Cabinet in the Cabinet Type menu, the Motor Ctrl setting is irrelevant, because there are no separate bass and treble rotors in a Single Cabinet.

Scanner Vibrato

The Scanner Vibrato effect simulates the scanner vibrato section of a Hammond organ. You can choose between three different vibrato and chorus types. The stereo version of the effect features two additional parameters: Stereo Phase and Rate Right. These allow you to set the modulation speed independently for the left and right channels.





The stereo parameters of the mono version of the Scanner Vibrato are hidden behind a transparent cover.

- *Vibrato knob:* Sets the desired vibrato or chorus types. The C0 setting disables both the vibrato and chorus effect.
- Chorus Int knob: Sets the intensity of a chosen chorus effect type. If a vibrato effect type is chosen, this parameter has no effect.

- Stereo Phase knob: If set to a value between 0 and 360 degrees, Stereo Phase determines the phase relationship between left and right channel modulations, thus enabling synchronized stereo effects. If "free" is chosen, you can set the modulation speed of the left and right channel independently.
- Rate Left knob: Sets the modulation speed of the left channel when Stereo Phase is set to free. If Stereo Phase is set to a value between 0 and 360 degrees, Rate Left sets the modulation speed for both the left and right channels. Rate Right has no function when in this mode.
- Rate Right knob: Sets the modulation speed of the right channel when Stereo Phase is set to free.

Spreader

You can use the Spreader effect to widen the stereo spectrum of a signal. The Spreader effect periodically shifts the frequency range of the original signal in a nonlinear way, changing the perceived width of the signal.



- Intensity slider and field: Determines the modulation amount.
- Speed knob and field: Defines the frequency, and therefore the speed, of the LFO.
- Channel Delay slider and field: Determines the delay time in samples.
- Mix slider and field: Sets the balance between the dry and wet signals.

Tremolo

The Tremolo effect modulates the amplitude of a signal, resulting in periodic volume changes. You'll recognize this effect from vintage guitar combo amps (where it is sometimes incorrectly referred to as vibrato). The graphic display shows all parameters, except Rate.



- Depth slider and field: Determines the modulation amount.
- Rate knob and field: Defines the frequency, and therefore the speed, of the LFO.
- Symmetry and Smoothing knobs and fields: Use these to set the shape of the modulation.
- Phase knobs and fields (only available in stereo and surround instances): Controls the phase relationship between the individual channel modulations. At 0°, the extreme values of the modulation are achieved simultaneously for all channels. 180° or –180° is equal to the greatest possible distance between the modulation phases of the channels.
- Distribution menu (only available in surround instances): Defines how the phase offsets between the individual channels are distributed in the surround field. You can choose from Circular, Random, Front <> Rear, and Left <> Right distribution. When you load a setting that uses the Random option, the saved phase offset value is recalled. If you want to randomize the phase setting again, choose "new random" in the Distribution menu.
- Offset slider and field (extended parameter): Sets the amount that the modulation (cycle) is shifted to the left or right, resulting in subtle or noticeable tremolo variations.

Symmetry and Smoothing

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.

Pitch 10

You can use the Pitch effects of Logic Studio to transpose or correct the pitch of audio tracks.

These effects can also be used for creating unison or slightly thickened parts, or even the creation of harmony voices.

Logic Studio includes the following Pitch effects:

- "Pitch Correction" on page 151
- "Pitch Shifter II" on page 155
- "Vocal Transformer" on page 156

Pitch Correction

You can use the Pitch Correction plug-in to correct the pitch of audio tracks. Improper intonation is a common problem with vocal tracks, for example. The sonic artifacts that can be introduced by the process are minimal and can barely be heard, as long as your corrections are moderate. The correction works by accelerating and slowing down the playback speed, ensuring that the input signal (sung vocal) always matches the correct note pitch. If you use the algorithm to correct larger intervals, you can create special effects. The natural articulations of the performance, such as breath, are preserved. Any scale can be defined as a pitch quantization grid. Improperly intonated notes will be corrected in accordance with this grid.

Note: Polyphonic recordings (choirs on one track) and highly percussive signals, with prominent noisy portions, can't be corrected to a specific pitch. Despite this, feel free to try the plug-in on drum tracks!

Pitch Correction Parameters



- *Normal and Low buttons:* These determine the pitch range that is scanned (for notes that need correction).
- *Use Global Tuning button:* Enable to use the project's Tuning settings for the pitch correction process. If this button is switched off, you can use the Ref. Pitch field to freely set the desired reference tuning, in cents.
- Scale field: Click to choose different pitch quantization grids from the Scale menu.
- Root field: Click to choose the root note of the scale.
- Keyboard: Click a key to exclude the corresponding note from pitch quantization grids. This effectively removes this key from the scale, resulting in note corrections that are forced to the nearest available pitch (key).
- Bypass buttons: Click to exclude the corresponding note from pitch correction. In
 other words, all notes that match this pitch will not be corrected. This applies to both
 user and preset scale quantization grids. The Bypass All button provides a quick way
 to compare the corrected and original signals, or for automation changes.
- Show Input and Show Output buttons: Click to display the pitch of the input or output signal, respectively, on the notes of the keyboard.
- Correction Amount display: Indicates the amount of pitch change. The red marker indicates the average correction amount over a longer time period.
- Response slider and field: Determines how quickly the voice reaches the corrected destination pitch.
- Detune Slider and field: Detunes the incoming signal by the set value.

Using the Pitch Correction Plug-in

You can use the Normal and Low buttons to determine the pitch range that you want to scan for notes that need correction. Normal is the default range, and works for most audio material. Low should only be used for audio material that contains extremely low frequencies (below 100 Hz), which may result in inaccurate pitch detection. These parameter have no affect on the sound, they are simply optimized tracking options for the chosen target pitch range.

Defining the Pitch Quantization Grid

The Scale menu allows you to choose different pitch quantization grids. The scale that is set manually (with the keyboard) is called the User Scale. The default setting is the chromatic scale. The other scale names are self-explanatory. If you're unsure of the intervals used in any given scale, simply choose it in the Scale menu and check out the values shown on the keyboard. You can alter any note in the scale by clicking on the keyboard keys. Any such adjustments will overwrite the existing User Scale settings.

There is only one User Scale per project. You can, however, create multiple User Scales, and save them as Pitch Correction plug-in settings files.

Tip: The Drone scale uses a fifth as a quantization grid, the Single scale defines a single note. Both of these scales aren't meant to result in realistic singing voices, so if you're after interesting effects, you should give both of them a try.

Click the Root field to choose the root note of the scale from a pop-up menu. there are two scales that are excluded: the User and Chromatic scales—where there is no root note (none). You may freely transpose the major and minor scales, and scales named after chords.

You can use the keyboard to exclude notes from the pitch quantization grid. When you first open the plug-in, all notes of the chromatic scale are selected. This means that every incoming note will be altered to fit the next semitone step of the chromatic scale. If the intonation of the singer is poor, this might lead to notes being incorrectly identified, and corrected. As an example, the singer might have intended to sing an E, but the note is actually closer to a D#. If you don't want the D# in the song, the D# key can be disabled on the keyboard. Given that the original pitch was sung closer to an E than D, it will be corrected to an E.

Note: The settings are valid for all octave ranges. Individual settings for different octaves aren't provided.

Excluding Notes From Correction

Use of the small bypass buttons (byp) above the green (black) and below the blue (white) keys excludes notes from correction. This is useful for blue notes. Blue notes are notes that slide between pitches, making the major and minor status of the keys difficult to identify. As you may know, one of the major differences between C minor and C major is the Eb (E flat) and Bb (B flat), instead of the E and B. Blues singers glide between these notes, creating an uncertainty or tension between the scales. Use of the bypass switches allows you to exclude particular keys from changes, leaving them as they were.

If you enable the Bypass All button, the input signal is passed through unprocessed and uncorrected. This is useful for spot corrections to pitch through use of automation. Bypass All is optimized for seamless bypass switching in all situations.

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Tip: You'll often find that it's best to only correct notes with the most harmonic gravity. As an example, choose the "sus 4" Scale, and set the Root note to match the project key. This will limit correction to the root note, the fourth and the fifth of the key scale. Switch all other notes to Bypass, and only the most important and sensitive notes will be corrected, while all other singing remains untouched.

Setting the Reference Tuning

The File > Project Settings > Tuning settings determine the tuning reference for all software instruments. If you engage Use Global Tuning in the Pitch Correction window, the project's Tuning settings will be used for the pitch correction process. If this parameter is switched off, you can use the Ref. Pitch field to freely set the desired reference tuning.

As an example: The intonation of a vocal line is often slightly sharp or flat throughout an entire song. Use the Reference Pitch parameter to address this issue at the input of the pitch detection process. Set the Reference Pitch to reflect the constant pitch deviation in cent values. This allows the pitch correction to perform more accurately.

Note: Tunings that differ from software instrument tuning can be interesting, when you want to individually correct the notes of singers in a choir. If all voices were individually and perfectly corrected to the same pitch, the choir effect would be partially lost. You can avoid this by (de)tuning the pitch corrections individually.

Setting the Response of Pitch Changes

Use the Response parameter to determine how quickly the voice reaches the destination (corrected) pitch. Singers use portamenti and other gliding techniques. If you choose a Response value that's too high, seamless portamenti turn into semitone-stepped glissandi, but the intonation will be perfect. If the Response value is too slow, the pitch of the output signal won't change quickly enough. The response of pitch changes is indicated in milliseconds. The optimum setting for this parameter depends on the singing style, tempo, vibrato, and quality of the original performance.

Understanding the Correction Amount Display

The amount of pitch change is indicated in the horizontal bar displayed below the keyboard. The red marker indicates the average correction amount over a longer time period.



If you keep a close eye on this display, you can use it for two important tasks: To better understand the inner workings of the algorithm, and adjust the Response accordingly. You can also use the display when discussing (and optimizing) the vocal intonation with a singer during a recording session.

Automating the Pitch Correction Plug-in

Pitch Correction can be fully automated. This means that you can automate the Scale and Root parameters to follow the harmonies of the song. Depending on the quality of the original intonation, setting the key Scale might suffice. Weaker intonations might need more significant changes to the Scale and Root parameters.

Pitch Shifter II

The Pitch Shifter II provides a simple way to combine a pitch-shifted version of the signal with the original signal.

Pitch Shifter II Parameters



- Semi Tones slider and field: Sets the pitch shift value in semitones.
- Cents slider and field: Controls detuning of the pitch shift value in cents (100ths of a semitone).
- *Drums, Speech, and Vocals buttons:* Select one of the three presets to optimize Pitch Shifter II operation for common types of audio material:
 - Drums leaves the groove of the original track intact.
 - Vocals retains the intonation of the original with no change. Hence Vocals is wellsuited for any signals that are inherently harmonic or melodious, such as string pads.
 - Speech provides a compromise between the two by attempting to retain both the rhythmic and harmonic aspects of the signal. This is suitable for complex signals such as spoken-word recordings, rap music, and other hybrid signals such as rhythm guitar.
- Mix slider and field: Sets the amount of the processed signal mixed with the original signal.

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• Timing pop-up menu (extended parameters): Sets whether the timing follows the selected preset (Preset), creates a new preset by analyzing the incoming signal (auto), or uses the settings of the Delay, Crossfade, and Stereo Link parameters, described below (manual).

The following three parameters are active only when manual is chosen in the Timing pop-up menu.

- Delay slider and field: Sets the amount of delay applied to the input signal. The deeper the frequencies of the input signal, the higher (longer) a delay time you should set—in order to effectively pitch shift the signal.
- *Crossfade slider and field:* Sets the amount of crossfading between the two pointers—used by the Pitch Shifter to analyze the input signal.
- Stereo Link pop-up menu: Choose Inv. to invert the stereo channel's signals, with the processing for the right channel occurring on the left, and vice versa. Choosing Normal produces no change to the signal.

Using Pitch Shifter II

Set the amount of transposition (pitch shift) with the Semi Tones parameter, then set the amount of detuning with the Cents parameter. Use one of the three presets (Drums, Vocals, or Speech) depending on the material you are working with. For other types of material, you can try each of the presets (starting with Speech), compare the results, and use the one that best suits your material. While you are auditioning and comparing different settings, it's often a good idea to temporarily set the Mix parameter to 100% to hear the maximum effect of the processing. Keep in mind that Pitch Shifter II artifacts are much harder to hear with Mix set to a smaller percentage.

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

Vocal Transformer

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

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

Vocal Transformer Parameters



- Pitch knob and field: Determines the amount of transposition applied to the input signal.
- Glide (extended parameter): Determines the amount of time the vocal transformation takes, allowing sliding transitions to the set Pitch value.
- Formant knob and field: Shifts the formants of the input signal.
- Formants menu (extended parameters): Determines whether the Vocal Transformer processes all formants ("Process always" setting), or only the voiced ones ("Keep unvoiced formants" setting). The "Keep unvoiced format" option leaves the typical sibilant noise of a vocal performance untouched. For some applications, this setting will produce a more natural sounding transformation effect.
- *Robotize button:* Click to switch the Vocal Transformer to Robotize mode. Robotize mode is used for augmenting, diminishing, or mirroring the melody.
- *Tracking slider and buttons (only available in Robotize mode):* Control how the melody is changed in Robotize mode.
- *Pitch Base slider and field (only available in Robotize mode):* Use to transpose the note that the Tracking parameter is following.
- Mix slider and field: Defines the level ratio between the original (dry) and effect signals.
- Grain Size field and slider (extended parameter): The Vocal Transformer's algorithm is based on granular synthesis. The Grain Size parameter allows you to set the size of the grains, and thus affect the precision of the process. Experiment to find the best setting. Try Auto first.

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Setting the Pitch and Formant Parameters

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

Transpositions of a fifth upward (Pitch = +7), a fourth downward (Pitch = -5), or by an octave (Pitch = ± 12) are the most useful, harmonically.

As you alter the Pitch parameter, you might notice that the formants don't change.

Formants are characteristic emphases of certain frequency ranges. They are static and do not change with pitch. Formants are responsible for the specific timbre of a given human voice.

The Pitch parameter is expressly used to change the pitch of a voice, not its character. If you set negative Pitch values for a female soprano voice, you can turn it into an alto voice, without changing the specific character of the singer's voice.

The Formant parameter shifts the formants, while maintaining—or independently altering—the pitch. If you set this parameter to positive values, the singer sounds like Mickey Mouse. By altering the parameter downwards, you can achieve vocals reminiscent of Darth Vader.

Tip: If you set Pitch to 0 semitones, Mix to 50%, and Formant to +1 (with Robotize switched off), you can effectively place a singer (with a smaller head) next to the original singer. Both will sing with the same voice—in a choir of two. This choir effect is quite effective, and is easily controlled with the Mix parameter.

Using Robotize Mode

If you switch Robotize on, the Vocal Transformer can augment or diminish the melody. You can control the intensity of this distortion with the Tracking parameter.

The four -1, 0, 1, and 2 buttons set the Tracking slider to values of -100%, 0%, 100%, and 200%, respectively. These buttons are convenience controls that make it easier to set the Tracking parameter to the most useful settings.

- At a value of 100% (switch 1), the range of the melody is maintained. Higher values augment, and lower values diminish, the melody.
- At a setting of 200% (switch 2), the intervals are doubled.
- A setting of 0% (switch 0) delivers interesting results, with every syllable of the vocal track being sung at the same pitch. Low values turn sung lines into spoken language.
- At a setting of -100% (switch -1), all intervals are mirrored.

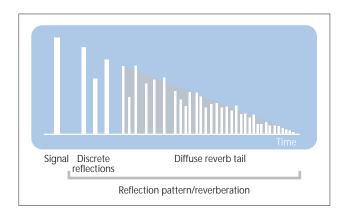
The Pitch Base parameter is used to transpose the note that the Tracking parameter is following. As an example: the note which is spoken, if Tracking is set to 0%.

Reverb 11

You can use Reverb effects to simulate the sound of acoustic environments such as rooms, concert halls, caverns, or the sound of infinite space.

Sounds bounce off the surfaces of any space, or off objects within a space, repeatedly, gradually dying out until they are inaudible. The bouncing soundwaves result in a reflection pattern, more commonly known as a reverberation (or reverb).

The early part of a reverb consists of a number of discrete reflections that you can clearly discern before the diffuse reverb tail builds up. These early reflections are essential to how you perceive the space of a room. All information about the size and shape of a room that the human ear can discern is contained in these early reflections.



Plates, Digital Reverb Effects, and Convolution Reverb

The first form of reverb used in music production was actually a special room with hard surfaces (called an echo chamber). It was used to add echoes to the signal. Mechanical devices, including plates and springs, were used to add reverberation to the output of musical instruments and microphones.

Digital recording introduced digital reverb effects, which consist of thousands of delays of varying lengths and intensities. The time between the original signal and the arrival of the early reflections can be adjusted by a parameter commonly known as predelay. The average number of reflections in a given period of time is determined by the density parameter. The regularity or irregularity of the density is controlled with the diffusion parameter.

Ever-increasing computing power has made it possible to sample the reverb characteristics of real spaces, using convolution reverbs. These room characteristic sample recordings are known as impulse responses.

Convolution reverbs work by convolving (combining) an audio signal with the impulse response recording of a room's reverb character.

This chapter describes the reverb effects included in Logic Studio:

- "AVerb" on page 160
- "EnVerb" on page 161
- "GoldVerb" on page 162
- "PlatinumVerb" on page 164
- "SilverVerb" on page 167
- Space Designer: Space Designer is a convolution reverb and is described separately in Chapter 12, "Convolution Reverb: Space Designer," on page 169.
- "Soundtrack Pro Reverb" on page 168

AVerb

The AVerb is a simple reverb effect that uses one parameter to control both the effect's early reflections and diffuse reverb tail: Density/Time. Small value settings tend to generate clearly discernible early reflection clusters, generating something similar to an echo. High values result in a reverb-like effect.

The AVerb is a quick and easy tool for creating a range of interesting sonic space and echo effects. The AVerb may not be the best choice for the simulation of real acoustic environments, however.



• *Predelay:* Determines the time between the original signal and the early reflections of the reverb signal.

- Reflectivity: Defines how reflective the imaginary walls, ceiling, and floor are. How hard the walls are, and what they're made of, in other words; glass, stone, timber, carpet, and other materials have a dramatic impact on the tone of the reverb.
- Room Size: Defines the dimensions of simulated rooms.
- Density/Time: Determines both the density and duration of the reverb.
- Mix: Determines the balance between the effected (wet) and direct (dry) signals.

FnVerb

The EnVerb is a versatile reverb effect with a unique feature: It allows you to freely adjust the envelope of the diffuse reverb tail.



The interface can be broken down into three areas:

- *Time parameters:* These determine the delay time of the original signal and reverb tail, and change the reverb tail over time. The graphic display visually represents the envelope of the reverb.
- Sound parameters: This area allows you to shape the sound of the reverb signal. You can also use the Crossover parameter to split the incoming signal into two bands, and set the level of the low frequency band separately.
- *Mix parameter:* Determines the balance between the effected (wet) and direct (dry) signals.

Time Parameters

- *Dry Signal Delay:* Determines the delay of the original signal. You can only hear the dry signal when the Mix parameter is set to a value other than 100%.
- *Predelay*: Sets the time between the original signal, and the starting point of the reverb attack phase (the very beginning of the first reflection).
- Attack: Defines the time it takes for the reverb to climb to its peak level.
- Decay: Defines the time it takes for the level of the reverb to drop from its peak to the sustain level.
- Sustain: Sets the level of the reverb that remains constant throughout the sustain phase. It is expressed as a percentage of the full scale volume of the reverb signal.

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- Hold: Sets the duration (time) of the sustain phase.
- *Release:* Sets the time that the reverb takes to fade out completely, after it has completed the sustain phase.

Sound Parameters

- Density: Sets the reverb density.
- *Spread:* Controls the stereo image of the reverb. At 0%, the effect generates a monaural reverb. At 200%, the stereo base is artificially expanded.
- *High Cut*: Frequencies above the set value are filtered out of the reverb tail.
- *Crossover:* Defines the frequency at which the input signal is split into two frequency bands, for separate processing.
- Low Freq Level: Determines the relative reverb level of frequencies below the crossover frequency. In most cases, you'll get better-sounding results when you set negative values for this parameter.

GoldVerb

The GoldVerb allows you to edit both the early reflections and diffuse reverb tail separately, making it easier to precisely emulate real rooms.



The interface can be broken down into four parameter groups:

- Early Reflections parameters: This section emulates the original signal's first reflections as they bounce off the walls, ceiling, and floor of a natural room.
- Reverb parameters: This section controls the diffuse reverberations.
- Balance ER/Reverb parameter: Controls the balance between the Early Reflections and Reverb section. When you set the slider to either of its extreme positions, the unused section is deactivated.
- *Mix parameter:* Determines the balance between the effected (wet) and direct (dry) signals.

Early Reflection Parameters

- *Predelay:* Determines the amount of time between the start of the original signal, and the arrival of the early reflections.
- Room Shape: Defines the geometric form of the room. The numeric value (3 to 7) represents the number of corners in the room. The graphic display visually represents this setting.
- *Room Size:* Determines the dimensions of the room. The numeric value indicates the length of its walls—the distance between two corners.
- Stereo Base: Defines the distance between the two virtual microphones that you are using in the simulated room. Spacing the microphones slightly further apart than the distance between two human ears generally delivers the best results. More realistic results can be obtained if you choose to use the distance between two ears located on opposite sides of the same head. This parameter is only available in stereo instances of the effect.

Reverb Parameters

- Initial Delay: Sets the time between the original signal and the diffuse reverb tail.
- Density: Controls the density of the diffuse reverb tail.
- Diffusion (Extended Parameters): Sets the diffusion of the reverb tail.
- Reverb Time: Time it takes for the reverb level to drop by 60 dB.
- *High Cut*: Frequencies above the set value are filtered from the reverb signal.
- *Spread:* Controls the stereo image of the reverb. At 0%, the effect generates a monaural reverb. At 200%, the stereo base is artificially expanded.

Setting Predelay and Initial Delay

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 optimum Predelay setting depends on the type (or envelope) of 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 you start to hear undesirable side effects, such as an audible echo.

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

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Setting Density and Diffusion

Ordinarily, you want the signal to be as dense as possible. However, use of a lower Density value means the effect eats up less computing power. Beyond this, in rare instances, a high Density value can color the sound, which you can fix by simply reducing the Density knob value. Conversely, if you select a Density value that is too low, the reverb tail will sound grainy.

High Diffusion values represent a regular density, with few alterations in level, times, and panorama position. Low Diffusion values result in the reflection density becoming irregular and grainy. The stereo spectrum changes, too.

Setting the Reverb Time

Reverb Time 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.

Setting the 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.

PlatinumVerb

The PlatinumVerb allows you to edit both the early reflections and diffuse reverb tail separately, making it easier to precisely emulate real rooms. Its dual-band Reverb section splits the incoming signal into two bands, each of which is processed (and can be edited) separately.



The interface can be broken down into four parameter groups:

- Early Reflections parameters: Emulates the original signal's first reflections as they bounce off the walls, ceiling, and floor of a natural room.
- Reverb parameters: Controls the diffuse reverberations.
- Balance ER/Reverb parameter: Controls the balance between the Early Reflections and Reverb section. When you set the slider to either of its extreme positions, the unused section is deactivated.
- *Output section:* Determines the balance between the effected (wet) and direct (dry) signals.

Early Reflection Parameters

- *Predelay:* Determines the amount of time between the start of the original signal and the arrival of the early reflections.
- Room Shape: Defines the geometric form of the room. The numeric value (3 to 7) represents the number of corners in the room. The graphic display visually represents this setting.
- Room Size: Determines the dimensions of the room. The numeric value indicates the length of its walls—the distance between two corners.
- Stereo Base (only available in stereo instances): Defines the distance between the two virtual microphones that you are using in the simulated room. Spacing the microphones slightly further apart than the distance between two human ears generally delivers the best results. More realistic results can be obtained if you choose to use the distance between two ears located on opposite sides of the same head.
- ER Scale (Extended Parameter): Scales the early reflections along the time axis, influencing the Room Shape, Room Size, and Stereo Base parameters simultaneously.

Reverb Parameters

- Initial Delay: Sets the time between the original signal and the diffuse reverb tail.
- *Spread:* Controls the stereo image of the reverb. At 0%, the effect generates a monaural reverb. At 200%, the stereo base is artificially expanded.
- *Crossover:* Defines the frequency at which the input signal is split into two frequency bands, for separate processing.
- Low Ratio: Determines the reverb time of the bass band in relation to the reverb time of the high band. It is expressed as a percentage, ranging from 0 to 200%.
- Low Freq Level: Sets the level of the bass reverb. At 0 dB, the volume of the two bands is equal.
- High Cut: Frequencies above the set value are filtered from the reverb signal.
- Density: Controls the density of the diffuse reverb tail.
- Diffusion: Sets the diffusion of the reverb tail.
- Reverb Time: Determines the reverb time of the high band.

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Output Parameters

- Dry: Controls the amount of the original signal.
- Wet: Controls the amount of the effect signal.

Setting Predelay and Initial Delay

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 optimum Predelay setting depends on the type (or envelope) of 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 you start to hear undesirable side effects, such as an audible echo.

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

Setting Density and Diffusion

Ordinarily, you want the signal to be as dense as possible. However, use of a lower Density value means the effect eats up less computing power. Beyond this, in rare instances, a high Density value can color the sound, which you can fix by simply reducing the Density knob value. Conversely, if you select a Density value that is too low, the reverb tail will sound grainy.

High Diffusion values represent a regular density, with few alterations in level, times, and panorama position. Low Diffusion values result in the reflection density becoming irregular and grainy. The stereo spectrum changes, too.

Setting the Reverb Time

Reverb Time 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.

Setting the 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.

Setting the Reverb Time and Level of the Low Frequency Band

You can use the Low Ratio control to offset the reverb time of the low frequency 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 on a kick and snare drum loop, a short reverb time for the kick drum allows you to set a substantially higher wet signal.

The Low Freq Level slider allows you to boost or attenuate the level of the low frequency band. 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.

SilverVerb

The SilverVerb is similar to the AVerb, but provides an additional LFO that you can use to modulate the reverberated signal. It also includes a high and a low cut filter, allowing you to filter frequencies from the reverb signal. High frequencies usually sound somewhat unpleasant, hamper speech intelligibility, or mask the overtones of the original signals. Long reverb tails with a lot of bottom end generally makes for a flabby mix.



- Predelay: Determines the time between the original signal and the reverb signal.
- Reflectivity: Defines how reflective the imaginary walls, ceiling, and floor are.
- Room Size: Defines the dimensions of simulated rooms.
- Density/Time: Determines both the density and duration of the reverb.
- *High Cut:* Frequencies above the set value are filtered out of the reverb signal. Note that this only affects the tone of the reverb, not the source signal.
- Low Cut: Frequencies below the set value are filtered out of the reverb tail. As above, it only affects the reverb signal.
- Mod Rate: Sets the frequency (speed) of the LFO.

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- Modulation Phase: Defines the phase of the modulation between the left and right channels of the reverb signal. At 0°, the extreme values (minimum or maximum) of the modulation are achieved simultaneously on both the left and right channels. At a value of 180°, the extreme values opposite each other (left channel minimum, right channel maximum, or vice-versa) are reached simultaneously.
- Mod. Intensity: Sets the modulation amount. A value of 0 turns the delay modulation
 off.
- Mix: Sets the balance between the effected and dry signal.

Soundtrack Pro Reverb

The Soundtrack Pro Reverb provides a simple reverb effect that requires only modest CPU resources. You can use it on both musical and non-musical audio material.



- Mix (dry/wet mix %) slider and field: Sets the ratio of the original (dry) signal to the effected (wet) signal at the output.
- Decay (decay %) slider and field: Sets the percentage of the processed signal fed back into the effect.

Convolution Reverb: Space Designer

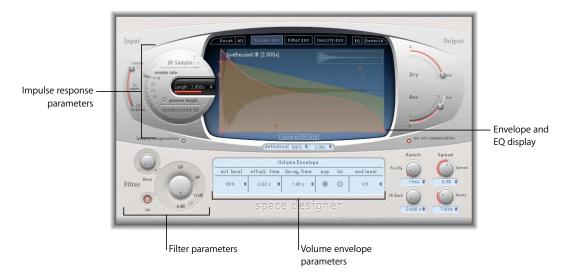
Space Designer is a *convolution* reverb effect. You can use it to create exceptionally realistic reverberations.

Space Designer generates reverb by convolving, or combining, an audio signal with an impulse response (IR) reverb sample. For example, imagine that you apply the Space Designer to a vocal track. If you load an IR recorded from an actual opera house into Space Designer, it will convolve the IR of the opera house with your vocal track and put the singer inside the opera house.

An impulse response is a recording of a room's reverb character; to be more precise: a recording of all the reflections in a given room, following an initial signal spike. The actual impulse response file is a standard audio file—it is not the file type that is unique, it is how the file is used.

Space Designer can operate as mono, stereo, true stereo (meaning each channel is processed discretely), or surround effect. Space Designer not only loads existing impulse responses, but offers sound sculpting features such as envelopes, filters, EQ, and stereo/surround balance controls that provide you with unprecedented control over dynamics, timbre, and length via a comprehensive set of parameters. In addition, Space Designer includes an on-board impulse response synthesis facility.

You can use Space Designer to create both a highly realistic reverb if you use an IR recorded from a real space, or a completely unique effect if you use a synthesized IR that doesn't represent any real space. Convolution can be used to put your audio signal inside any space, including a speaker cabinet, plastic toy, and so on, if you make an IR from it. And with Space Designer's extensive audio processing features, you can exactly tailor its space to your material.



Space Designer consists of the following parameter groups:

- Impulse response parameters: Use these parameters to load, save, or manipulate impulse response files. The IR file you choose determines what Space Designer will use to convolve with your audio signal. These parameters will be the initial parameters you use to load your IR file, as well as the last if you wish to save your synthesized IR. See "Impulse Response Parameters" on page 171.
- Global parameters: Once your IR is loaded, you will use these general parameters to adjust how Space Designer operates on the signal and IR. Global parameters include input and output parameters, delay and volume compensation, predelay, and so on. These parameters affect the overall processing of Space Designer, as opposed to the more particular parameter groups that affect one specific aspect of Space Designer's processing. See "Global Parameters" on page 174.
- Envelope and EQ display: Use the button bar at the top to switch the display between envelopes and the EQ. You can edit the selected parameters both graphically and numerically in the display itself.
- *Volume envelope parameters:* Use the volume envelope to dynamically animate the volume of your reverb over the duration of the impulse response. See "Volume Envelope Parameters" on page 183.

- Filter parameters: You can further modify the timbre of the Space Designer reverb using these resonant filter parameters. You can choose from a number of filter modes, adjust the resonance of the filter, as well as adjust the filter envelope dynamically over time, as you can with the volume envelope. See "Filter Parameters" on page 184.
- Synthesized impulse response parameters: If you do enough processing of the original IR, you may want to synthesize a new IR from your edited parameters. Use these parameters to adjust the density envelope and other synthesized IR parameters. See "Synthesizer Impulse Response Parameters" on page 186.
- EQ: For final sound sculpting, Space Designer includes a built-in four-band EQ: two shelving filters, and two parametric filters. Use these parameters to fine tune the sound of your reverb to your taste. See "EQ Parameters" on page 188.

Important: In order to convolve audio in real time, Space Designer must first calculate any parameter adjustments to the impulse response. This requires a moment or two, following parameter edits, and is indicated by a blue progress bar. During this time you can continue to adjust the parameter. Once calculation starts, the blue bar is replaced by a red bar, advising you that calculation is taking place.





Impulse Response Parameters

Space Designer can use either impulse response files or its own synthesized impulse responses. The circular area to the left of the Envelope and EQ display contains the impulse response parameters. Here you can determine the Impulse Response mode (IR Sample mode or Synthesized IR mode), load or create impulse responses, and set the sample rate.



• *IR Sample button:* Click to switch Space Designer to the IR Sample mode. The loaded impulse response sample is used to generate reverberation.

- IR Sample Arrow button: Click to load an impulse response.
- Sample Rate parameter: Determines the sample rate of the loaded impulse response.
- *Preserve Length option:* Activate to preserve the length of the impulse response when changing the sample rate.
- Length parameter: Adjusts the length of the impulse response.
- Synthesized IR button: Click to switch Space Designer to Synthesized IR mode. In this mode, Space Designer generates a new synthesized impulse response from the values of the Length, envelope, filter, EQ, and Spread parameters. You may freely switch between a loaded impulse response sample and a synthesized impulse response without losing the settings of the other. See "Synthesizer Impulse Response Parameters" for more information on working in Synthesized IR mode.

Working in IR Sample Mode

When you first click the IR Sample button, a file selector box opens, allowing you to select the desired impulse response file from a folder on your hard disk or CD. If you have already loaded an impulse response file, this button switches back from Synthesized IR to IR Sample mode.

Loading IRs

To change the currently loaded impulse response, click the downward pointing arrow to the right of the button. This accesses the following menu functions:

- Load IR: Loads an impulse response sample without changing the envelopes.
- Load IR & Init: Loads an impulse response sample and initializes the envelopes.
- Show in Finder: Opens a Finder window that displays the file location.

The name of the loaded impulse response file and its length are displayed in the main display's Envelope view.

All impulse responses that ship with Logic Studio are installed in the /Library/Audio/ Impulse Responses/Apple folder. The default name for deconvolution files consists of the source file name, appended with an ".sdir" file extension.

Impulse Response Formats

Any mono, stereo, AIFF, SDII, or WAV file can be used. In addition, since Space Designer supports surround formats up to 7.1 surround, discreet audio files and B-format audio files that comprise a single surround IR can also be used.

Setting the Sample Rate

The Sample Rate slider is used to determine the sample rate of an impulse response. You can choose between the following settings:

- Orig: Space Designer uses the current project sample rate. When loading an impulse response, Space Designer automatically converts the sample rate of the impulse response to match the current project sample rate—should it be necessary. As an example, this allows you to load a 44.1 kHz impulse response into a project running at 96 kHz, and vice versa.
- /2, /4, /8: These settings are half-divisions of the preceding value—one-half, one-quarter, one-eighth. As examples:
 - If the project sample rate is 96 kHz, the options will be 48 kHz, 24 kHz, and 12 kHz.
 - If 44.1 kHz is the project sample rate, the options will be 22.05 kHz, 11.025 kHz, and 5512 Hz.

Changing the sample rate increases (up) or reduces (down) the frequency response (and length) of the impulse response, and to a degree the overall sound quality of the reverb. That said, don't worry too much if the maximum bandwidth of the reverb tail is reduced to 11.025 kHz when you select a sample rate of 22.05 kHz (half of 44.1 kHz). Natural room surfaces (concrete and tiles excluded) barely reflect such high frequencies.

By selecting half the sample rate, the impulse response becomes twice as long. The highest frequency that can be reverberated will be halved. This results in a behavior that is much like doubling every dimension of a virtual room (multiplying a room's volume by eight).

Another benefit of reducing the sample rate is that processing requires significantly less CPU power, making half sample rate settings the ideal solution for large, open spaces.

Activating the Preserve Length button preserves the length of the impulse response when the sample rate is changed. Manipulating these two parameters as you see fit can lead to interesting results.

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

If running Space Designer in a project that uses a higher sample rate than the impulse response, you may also want to reduce the impulse response sample rate. Make sure the Preserve Length function is enabled. This cuts CPU power consumption, without compromising reverb quality. There is no loss in reverb quality, because the impulse response will not benefit from the higher project sample rate.

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

Setting the Length of the Impulse Response

You can use the Length parameter to set the length of the impulse response (sampled or synthesized).

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

When using an impulse response file, the Length parameter value cannot exceed the length of the actual impulse response *sample*. Longer impulse responses (sampled or synthesized) place a higher strain on the CPU.

Global Parameters

Space Designer's Envelope and EQ display contains most of Space Designer's interface elements that change to reflect the current parameter group you are adjusting. The global parameters, spread throughout the interface around and below the Envelope and EQ display, remain constant. These parameters can be divided by the relief graphics of the interface.



The upper (raised) section of Space Designer contains the following global parameters:

• Input slider: Determines how Space Designer processes a stereo or surround input signal. See "Input Slider" on page 176 for more information.

- Latency Compensation button: Switches Space Designer's internal latency compensation feature on or off. See "Latency Compensation" on page 177 for more information.
- Definition area: Lets you configure Space Designer to switch to a less defined IR set in order to emulate reverb diffusion and save CPU. See "Definition" on page 177 for more information.
- Rev Vol Compensation: Engages Space Designer's internal IR volume matching (see "Rev Vol Compensation" on page 177 for more information).
- Output sliders: Adjust output levels (see "Output Parameters" for more information).



The lower (flat) section of Space Designer contains the following global parameters:

- *Filter parameters:* Activate, adjust the resonance, and select the mode of Space Designer's resonant filter. See "Filter Parameters" on page 184 for more information.
- *Pre-Dly knob:* Sets the reverb's predelay time, or time between the original signal and the first reflections from the reverb. See "Pre-Dly (Predelay)" on page 179 for more information.
- IR Start knob: Shifts the point at which the impulse response will play back. See "IR Start" on page 180 for more information.
- Spread knob: For synthesized impulse responses, this adjusts the perceived stereo width (for stereo instances of Space Designer) or surround width (for surround instances) to widen the perceived stereo or surround field and enhance the stereo or surround effect. See "Spread Parameters" on page 187 for more information.
- *Xover knob:* Sets the crossover frequency below which synthesized IRs will be processed by the Spread knob. See "Spread Parameters" on page 187 for more information.

Input Slider

The Input slider functions as either a stereo processing slider for stereo instances of Space Designer, or as an LFE to Reverb slider in surround mode. The Input slider does not appear in mono or mono to stereo instances of Space Designer.

Stereo Mode

For stereo instances of Space Designer, the Input slider determines how a stereo signal is processed:



- Stereo setting (top of slider): The signal is processed on both channels, retaining the stereo balance of the original signal.
- Mono setting (middle of slider): The signal is processed in mono.
- XStereo setting (bottom of slider): The signal is inverted, with processing for the right channel occurring on the left, and vice versa.
- In-between positions: A mixture of stereo to mono crossfeed signals is produced.
- Tip: The three base positions of the Input slider are clickable key parameter positions—if you click them the slider will immediately jump to the clicked position.

Surround Mode

For surround instances of Space Designer, the Input slider determines how much of the LFE signal is mixed with the surround channels feeding the reverb.



At its lowest setting, the slider acts as an LFE bypass, with all the LFE signal passed through the reverb unprocessed.

Latency Compensation

The complex calculations made by Space Designer take time. This time results in a processing *latency*, or delay, between the direct (input) signal, and the processed (output) signal. The Latency Compensation button determines how Space Designer delays the direct signal in relation to the processed signal.



Space Designer's processing latency is 128 samples at 44.1 kHz, and doubles at each lower sample rate division. For example, if you set Space Designer's Sample Rate slider to "/2" the processing latency increases to 256 samples. The processing latency does not increase in surround mode or with higher sample rates than 44.1 kHz.

When activated, this parameter delays the direct signal (in the Output section) to match the processing delay of the effect signal. This is not related to latency compensation in the host application—this compensation occurs *within* Space Designer and applies only to Space Designer.

Rev Vol Compensation

Rev Vol Compensation (Reverb Volume Compensation) attempts to match the perceived (not actual) volume differences of impulse response files.



It is switched on by default and should generally be left in this mode, although you may find that it isn't successful with all types of impulse responses. In such situations, switch it off and adjust the input and output levels accordingly.

Definition

The Definition parameter appears in the definition area at the bottom of the Envelope and EQ display.



Calculating every precise detail of an impulse response derived reverb uses significant CPU power. The Definition parameter emulates the diffusion of natural reverb patterns while at the same time reducing Space Designer's CPU consumption.

Natural reverbs contain most of their spacial information in the first few milliseconds. Towards the end of the reverb, its reflection pattern diffuses more and more, containing less spacial information. In order to emulate this phenomenon—as well as conserve CPU power—you can configure Space Designer to only use the full IR resolution at the onset of the reverb, and to use a reduced IR resolution towards the end of the reverb.

The Definition parameter acts as the crossover point at which this switch to the reduced IR resolution occurs. The parameter is displayed in both milliseconds (indicating when the crossover occurs) and a percentage (100% is equal to the length of the full resolution IR).

Note: The Definition slider appears only when you have loaded CPU intensive impulse response formats, such as true stereo.

Output Parameters

The output parameters let you adjust the mix between the direct (dry) and processed signals. Which parameters are available depends on Space Designer's input configuration.

Mono and Stereo Configurations

If you insert Space Designer as mono, mono to stereo, or stereo effect, Space Designer offers two output sliders: one for the direct signal, and one for the reverb signal.



- Dry slider: Sets the level of the non-effected (dry) signal. Set this to a value of 0 (mute) if Space Designer is inserted in a bus channel, or when using modeling impulse responses such as speaker simulations.
- Rev(erb) slider: Adjusts the output level of the effected (wet) signal.

Surround Configuration

In surround configurations, Space Designer offers four output sliders that together comprise a small surround output mixer.



These sliders have the following functions:

- C(enter): Adjusts the center reverb level
- Bal(ance): Sets the balance between the L-C-R front and the Ls-Rs rear speakers. In
 7.1 ITU surround, the balance pivots around the Lm-Rm speakers, taking the
 surround angles into account. With 7.1 SDDS surround, the Lc-Rc speakers are
 considered front speakers.
- Rev(erb): Adjusts the output level of the effected (wet) signal.
- *Dry:* Sets the level of the non-effected signal. Set this to a value of 0 (mute) when using Space Designer on an aux channel.

Pre-Dly (Predelay)

Predelay is the amount of time that elapses between the original signal, and the initial early reflections from the reverb generated by Space Designer.

For a room of any given size and shape, predelay determines the distance between the listener and the walls, ceiling, and floor. Of course, Space Designer allows you to adjust this parameter 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, leaving an audible gap between the signals. The ideal predelay setting to create a realistic space depends on the properties of, 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 rule of thumb is to use the longest predelay possible before undesirable side effects, such as an audible echo, begin materializing.

Obviously, these guidelines are designed to help you design realistic sounding spaces. If you want to create unnatural soundscapes using Space Designer, experiment with the Pre-Dly parameter to create otherworldly reverbs and echoes.

IR Start

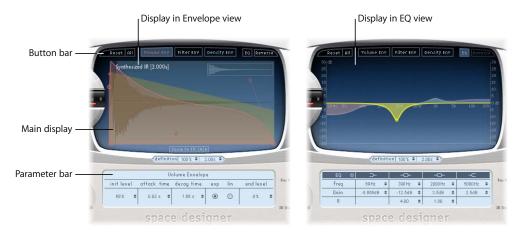
The IR Start parameter enables you to shift the playback point into the impulse response, which will effectively cut off the beginning of the impulse response.

The IR Start parameter can for example be used to eliminate any peaks at the beginning of the impulse response sample. It also offers a number of creative options, such as its use when combined with the Reverse function (see "Button Bar" on page 181).

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

Envelope and EQ Display

Space Designer's Envelope and EQ display features two components: the button bar at the top and the main display (including its parameter bar). The display itself shows either the envelope being edited or the EQ curve, depending on which button you engage.



Button Bar

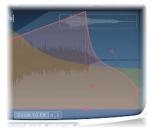
The Envelope and EQ display's button bar includes buttons to switch the main display between envelopes and the EQ, as well as some function buttons.



- Reset button: Click to reset the currently displayed envelope or EQ to its default values.
- All button: Click to reset all envelopes and the EQ to default values.
- Volume Env button: Click this button to bring the volume envelope to the front of the main display. The other envelope curves are shown as transparencies in the background. See "Volume Envelope Parameters" on page 183 for information on the volume envelope.
- Filter Env button: Click this button to bring the filter envelope to the front of the main display. The other envelope curves are shown as transparencies in the background. See "Filter Parameters" on page 184 for information on the filter envelope.
- Density Env button: Click this button to bring the density envelope to the front of the main display. The other envelope curves are shown as transparencies in the background. See "Synthesizer Impulse Response Parameters" on page 186 for information on the density envelope.
- *EQ button:* Click this button to switch the main display to Space Designer's four-band parametric EQ. See "EQ Parameters" on page 188 for more information on Space Designer's EQ.
- Reverse button: Click to reverse the impulse response together with its envelopes. When you reverse the impulse response, you are effectively using the tail rather than the front end of the sample. As such, you may need to use lower or even negative Pre-Dly values when reversing.

Additional View Items in Envelope View

When displaying envelopes, the main display offers a few other buttons and an overview that is not part of the EQ view:



• *Impulse response overview:* Indicates which portion of the impulse response file is currently visible, helping you to orientate yourself when zoom is active.

- Zoom to Fit button: Switch on to display the entire impulse response waveform. The display will automatically follow any envelope length changes.
- A and D buttons: Click to limit the Zoom to Fit function to the attack and decay portions of the (currently selected) envelope. The A and D buttons are available only to the volume and filter envelopes.

Setting Envelope Parameters

Space Designer allows you to edit the volume and filter envelopes of all IRs, and the density envelope of synthesized IRs. All three envelopes can be adjusted both graphically and numerically (in the parameter bar).

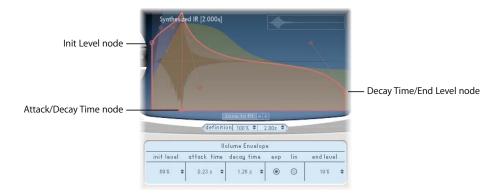
While some parameters are envelope-specific, all envelopes consist of the Attack Time and Decay Time parameters: The combined total of the Attack Time and Decay Time parameters is equal to the total length of the (synthesized or sampled) impulse response (determined by the Length parameter, see "Setting the Length of the Impulse Response" on page 174), unless the Decay time is reduced.



- You can change the curve shape by dragging the envelope curve directly. Use the small nodes attached to a line for finer adjustments to envelope curves. They are tied to the envelope curve itself, so you can view them as envelope handles. Moving the nodes vertically or horizontally will change the shape of the envelope curve.
- The large nodes are value indicators of the parameters that appear in the horizontal
 parameter bar below—Init Level, Attack Time, Decay Time, and so on. If you edit any
 numerical value, the corresponding node will move in the main display. Try this with
 each numerical parameter to establish which node is which. When you move the
 mouse pointer over one of these nodes, you'll see a pair of arrows. The arrows simply
 indicate the possible directions that the node can be moved.

Volume Envelope Parameters

The volume envelope lets you set the reverb's initial level and adjust how the volume will change over time. You can edit all of the volume envelope parameters numerically, and many of them can also be edited graphically using the techniques discussed in "Setting Envelope Parameters" on page 182.



The volume envelope includes the following parameters:

- *Init Level*: Sets the initial volume level of the impulse response attack. It is expressed as a percentage of the full-scale volume of the impulse response file. The attack phase is generally the loudest point of the impulse response. Set Init Level to 100% to ensure maximum volume for the early reflections.
- Attack Time: Determines the length of time before the decay phase of the volume envelope begins.
- Decay Time: Sets the length of the decay phase.
- Volume decay mode buttons: Click to choose the volume decay curve.
 - Exp: The output of the volume envelope is shaped by an exponential algorithm, in order to generate the most natural sounding reverb tail.
 - Lin: The volume decay will be more linear (and less natural sounding).
- End Level: Sets the end volume level. It is expressed as a percentage of the overall volume envelope. If you set this parameter to 0%, the reverb tail cuts off abruptly, which is great for gated reverb effects.

Filter Parameters

Space Designer's filter provides control over the timbre of the reverb. Its controls are distributed between two parts of the Space Designer interface: The main filter parameters are found in Space Designer's lower left corner, and the filter envelope appears on the Envelope and EQ display when its Filter button is engaged. You can select from several filter types, but you also have envelope control over the filter cutoff, independent from the volume envelope. Changes to the filter settings result in a recalculation of the impulse response, rather than a straight change to the sound as it plays through the reverb.

Main Filter Parameters



- Filter On/Off button: Switches the filter section on and off.
- Filter Mode knob: Selects between the four filter modes.
- Reso(nance) knob: Adjusting this parameter emphasizes frequencies above, around, or below the cutoff frequency. As you increase the resonance value, the sound will lose bass and become thinner. How much effect the resonance value has on the sound also depends on the selected filter mode, with steeper filter modes resulting in more pronounced resonance.

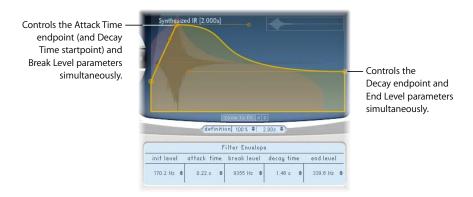
Setting the Filter Mode

The Filter Mode knob switches between four modes. Click on the desired LP (lowpass) 6 dB and 12 dB, BP (bandpass) or HP (highpass) value.

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

Filter Envelope Parameters

The filter envelope lets you control the filter's cutoff frequency over time. All of the filter envelope's parameters can be adjusted either numerically in the parameter area or graphically in the main display using the techniques discussed in "Setting Envelope Parameters" on page 182.



The parameters for the Filter Envelope are as follows:

- *Init Level:* Sets the initial cutoff frequency of the filter envelope.
- Attack Time: Determines the time required to reach the Break Level (see below) value.
- Break Level: Sets the maximum filter cutoff frequency the envelope reaches. It also acts as the break point between the attack and decay phases of the overall filter envelope. In other words, when this level has been reached after the attack phase, the decay phase will begin. You can create interesting filter sweeps by setting the Break Level to a value lower than that of the Init Level.
- *Decay Time*: Determines the time required (after the Break Level point) to reach the End Level value.
- End Level: Sets the filter end cutoff frequency.

Synthesizer Impulse Response Parameters

In Synthesizer IR mode, Space Designer generates a synthesized impulse response determined by the values of the Length, envelopes, filter, EQ, and spread parameters. To switch to Synthesizer IR mode, enable the Synthesized IR button in the impulse response parameters section.



Clicking the activated Synthesized IR button will *randomly* generate new impulse responses with slightly different reflection patterns. The current impulse response state will always be saved with a setting, allowing for an accurate reproduction of the reverberation sound when next loaded.

Density Envelope

The density envelope allows you to control the density of the synthesized impulse response over time. You can adjust the density envelope numerically in the parameter bar, and you can edit the Init Level, Ramp Time, and End Level parameters using the techniques described in "Setting Envelope Parameters" on page 182.

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



The density envelope offers the following parameters:

- *Init Level*: Sets the initial density (the average number of reflections in a given period of time) of the reverb. Lowering the density levels will result in audible reflections patterns and discreet echoes.
- Ramp Time: Adjusts the length of time elapsed between the Initial and End Density levels.

- End Level: Sets the density of the reverb tail. If you select an End Level value that is too low, the reverb tail will sound grainy. You may also find that the stereo spectrum is affected by lower values.
- Reflection Shape: Determines the steepness (shape) of the early reflection clusters as they bounce off the walls, ceiling, and furnishings of the virtual space. Small values result in clusters with a sharp contour, and large values result in an exponential slope and a smoother sound. This is handy when recreating rooms constructed of different materials. Reflection Shape, in conjunction with suitable settings for the envelopes, density, and early reflection will assist you in creating rooms of almost any shape and material.

Spread Parameters

The Spread and Xover knobs adjust the perceived stereo or surround width of a synthesized impulse response. While the Spread and Xover knobs are placed among the global parameters, they only function in Synthesized IR mode.



Note: As these parameters adjust stereo or surround processing, they have *no* impact when using the Space Designer as a mono plug-in.

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

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

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

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

EQ Parameters

Space Designer features a four-band EQ comprised of two parametric mid-bands plus two shelving filters (one low shelving filter and one high shelving filter).



The EQ has the following parameters:

- EQ On/Off button: Click to switch the entire EQ section on or off.
- Individual EQ buttons (1 through 4): Click to turn individual EQ bands on or off.
- Frequency: Sets the frequency for the selected EQ band.
- Gain: Adjusts the gain cut or boost for the selected EQ band.
- Q: Sets the Q-factor for the two parametric bands. The Q can be adjusted from 0.1 (very narrow) to 10 (very wide).

You can edit the EQ parameters numerically in the parameter bar, or graphically on the main 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.



- Drag any band to the right or left to adjust its frequency.
- Drag any band up to increase the Gain, down to lower the Gain.
- Position the mouse pointer directly on the (illuminated) pivot point of a parametric band and drag up to raise the Q or down to lower the Q.

Automating Space Designer

Space Designer cannot be fully automated as per most other Logic Studio plug-ins. This is because Space Designer needs to reload the impulse response (and recalculate the convolution) before audio can be routed through it.

You can, however, record, edit, and play back any movement of the following Space Designer parameters:

- Stereo Crossfeed
- Direct Output
- Reverb Output

Specialized 13

Logic Studio includes a bundle of specialized plug-ins designed to address tasks often encountered during audio production.

You should have a look at these specialized effects if you want to do one of the following:

- Eliminate or reduce noise below a threshold level (see "Denoiser" on page 192).
- Enhance the timing of audio recordings (see "Enhance Timing" on page 193).
- Add life to digital recordings by adding additional high frequency components (see "Exciter" on page 194).
- Vary recordings rhythmically (see "Grooveshifter" on page 195).
- Improve speech recordings made with your computer's internal microphone (see "Speech Enhancer" on page 196).
- Add an artificial bass signal, derived from the incoming signal (see "SubBass" on page 197).

Denoiser

The Denoiser eliminates or reduces any noise below a threshold volume level.

Denoiser Parameters



- *Threshold slider and field:* Sets the volume level (the threshold) below which the DeNoiser reduces the signal.
- Reduce slider and field: Sets the amount of noise reduction applied to sounds below
 the threshold. When reducing noise, remember that each 6 dB reduction is
 equivalent to halving the volume level (and each 6 dB increase equals a doubling of
 the volume level).
 - For example, if the noise floor of your recording is very high (more than –68 dB), reducing it to a level of –83 to –78 dB should be sufficient, provided this does not introduce any audible side effects. This effectively reduces the noise by more than 10 dB, to less than half of the original (noise) volume.
- Noise Type slider and field: Set to a value appropriate to the type of noise you want to reduce.
 - A value of 0 equals 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 Frequency knob: Adjusts how smoothing is applied to neighboring frequencies. If the Denoiser recognizes that only noise is present on a certain frequency band, the higher you set the Frequency Smoothing parameter, the more it changes the neighboring frequency bands to avoid glass noise.
- *Smoothing Time knob*: Sets the time required by the Denoiser to reach (or release) maximum reduction. This is the simplest form of smoothing.
- Smoothing Transition knob: Adjusts how smoothing is applied to neighboring volume levels. If the Denoiser recognizes that only noise is present in a certain volume range, the higher you set the Transition Smoothing parameter, the more it also changes similar level values to avoid glass noise.

• *Graphic display:* Shows how the lowest volume levels of your audio material (which should be mostly or entirely noise) are reduced. Changes to parameters are instantly reflected here, so keep an eye on it!

Using the Denoiser

Locate a section of the audio where only noise is audible, and set the Threshold value so that only signals at, or below, this level are filtered out. Then start playback and set the Reduce value as you listen to the audio, so that as much noise as possible is reduced, but as little of the desired signal is reduced.

The Denoiser uses FFT (Fast Fourier Transform) analysis to recognize frequency bands of lower volume and less complex harmonic structure, and then reduces them to the desired dB level. In principle, this method is completely discrete, as neighboring frequencies are also affected.

If you use the Denoiser too aggressively, however, the algorithm will produce artifacts, which are obviously artificial and therefore less desirable than the existing noise in most cases. If using the Denoiser produces these artifacts, you can use the three Smoothing knobs to reduce or eliminate them.

Enhance Timing

The Enhance Timing effect non-destructively enhances the timing of audio recordings.

Enhance Timing Parameters



- Intensity slider and field: Determines the amount of timing enhancement. Audio transients that don't fall on the grid divisions (determined by the value chosen in the Grid menu) are corrected.
- *Grid menu:* Allows you to choose between several grid divisions. As described above, the grid divisions serve as reference points for the timing correction process.

Using the Enhance Timing Effect

The Enhance Timing plug-in is designed to tighten up loose playing (of recorded audio) in a production. It can be used on a variety of material, and works in real time.

Obviously, this type of real-time quantization has some limitations. It will not work well on recordings of performances that have been played too far off the beat. The same is true for very complex, layered drum tracks. It will provide noticeable timing improvements on reasonably tight percussive and melodic material (played in an eighth or quarter note feel). If a large amount of timing correction is needed, and transients are shifted too far, you may notice a number of audio artifacts. As such, you should try to strike the balance between sound quality and timing enhancement.

Important: For technical reasons, the Enhance Timing plug-in only works on audio tracks, and must be inserted in the *top* Insert slot.

As a tip for triplets, try the 1/12 note setting for played 8th note triplets.

Exciter

The Exciter generates high frequency components that are not part of the original signal, using a nonlinear distortion process that resembles overdrive and distortion effects. Unlike those effects, however, the Exciter passes the input signal through a highpass filter before feeding it into the harmonics (distortion) generator. This results in the artificial harmonics added to the signal having frequencies at least one octave above the threshold of the highpass filter. The distorted signal is then mixed with the original, dry signal.

You can use the Exciter to add life to digital recordings. It is especially well suited to audio tracks with a weak treble frequency range. The Exciter is also useful for enhancing guitar tracks.

Exciter Parameters



- Frequency slider and field: Sets the cutoff frequency (in Hertz) of the highpass filter. The input signal passes through this filter before (harmonic) distortion is introduced.
- Frequency display: The graphic displays the frequency range that is used as the source signal for the process.

- *Input button*: When selected, the original (pre-effect) signal is mixed with the effected signal. If you disable Input, only the effected signal is heard.
- Harmonics knob and field: Sets the amount of the effected signal that is mixed with
 the original signal (expressed as a percentage). If the Input button is turned off, this
 has no effect on the signal. In most cases, higher Frequency and Harmonics values
 are preferable, because human ears cannot easily distinguish between the artificial
 and original high frequencies.
- Color 1 and Color 2 buttons: Click Color 1 to generate a less dense harmonic distortion spectrum. Click Color 2 for a more intense distortion. Color 2 also introduces more (unwanted) intermodulation distortions.

Grooveshifter

The Grooveshifter effect allows you to rhythmically vary recordings, imparting a swing feel to the track. Imagine a guitar solo played in straight eighth or sixteenth notes. The Grooveshifter can make this straightforward solo swing.

The reference tempo is the project tempo. The Grooveshifter will automatically follow all changes to the project tempo.

Note: The Grooveshifter is reliant on perfect matching of the project tempo with the tempo of the treated recording. Any tempo variations will deliver less precise results.



Grooveshifter Parameters

- *Tonal and Beat buttons:* Switch between two algorithms, each optimized for different types of audio material.
 - The Beat algorithm is optimized for percussive input material. THe Grain slider has no affect when Beat is chosen.
 - The Tonal algorithm is optimized for tonal input material. As this algorithm is based on granular synthesis, it offers an additional Grain slider, which allows you to set the size of the grains—and therefore, the precision of the analysis.
- Swing slider and field: Determines the amount that even beats will be delayed. A value of 50% means no swing, which is typical for most pop and rock music styles. The higher the value, the stronger the swing effect.

- *Grid buttons:* Determine the beat division used as a timing reference by the algorithm to analyze the audio material. Choose 1/8th if the audio material contains primarily eighth notes, and 1/16 if it consists mostly of sixteenth notes.
- Accent slider and field: Raises or lowers the level of even beats, accentuating them. Such accents are typical of a variety of rhythmic styles, such as swing or reggae.

Speech Enhancer

You can use the Speech Enhancer effect to improve speech recordings made with your computer's internal microphone (if applicable). It combines denoising, advanced microphone frequency remodeling, and multiband compression.



- Denoise slider and field: Determines (your estimation of) the noise floor in your recording, and therefore, how much noise should be eliminated. Settings towards 100 dB allow more noise to pass. Settings towards 0 dB will increasingly suppress background noise, but will also proportionately increase artifacts.
- *Mic Correction button:* Activate this button to improve the frequency response of recordings made with your built-in microphone. This creates the impression that an upmarket microphone was used.
- Mic Model menu: Choose the appropriate mic model. You can use the Speech
 Enhancer effect with other microphones, but microphone correction models are only
 offered for built-in Macintosh microphones. Should a non-Macintosh microphone be
 used, you will achieve the best results if Mic Correction is set to Generic.
- Voice Enhance button and Enhance Mode menu: Activation of this button turns on the multiband compression of the Speech Enhancer. Once active, you can choose between four Enhance Mode menu settings, making the recorded voice louder and more intelligible. Choose the setting that best matches your recording situation.

SubBass

The SubBass plug-in generates frequencies below those of the original signal—in other words, an artificial bass. The simplest use for the SubBass is as an octave divider, similar to Octaver effect pedals for electric bass guitars. Where such pedals can only process a monophonic input sound source of clearly defined pitch, SubBass can be used with complex summed signals as well. SubBass creates two bass signals, derived from two separate portions of the incoming signal. These are defined with the High and Low parameters.

Warning: Using the SubBass can produce extremely loud output signals! Choose moderate monitoring levels, and only use loudspeakers that are actually capable of reproducing the very low frequencies produced. Never try to force a loudspeaker to output these frequency bands with an EQ.

SubBass Parameters



- *High Ratio knob:* Adjusts the ratio between the generated signal and the original upper band signal.
- High Center knob: Sets the center frequency of the upper band.
- High Bandwidth knob: Sets the bandwidth of the upper band.
- Graphic display: Shows the selected upper and lower frequency bands.
- *Mix slider and field*: Adjusts the mix ratio between the upper and lower frequency bands.
- Low Ratio knob: Adjusts the ratio between the generated signal and the original lower band signal.
- Low Center knob: Sets the center frequency of the lower band.
- Low Bandwidth knob: Sets the bandwidth of the lower band.

- Dry slider and field: Sets the amount of dry (non-effected) signal.
- Wet slider and field: Sets the amount of wet (effected) signal.

Using the SubBass

Unlike a pitch shifter, the waveform of the signal generated by the SubBass is not based on the waveform of the input signal, but is sinusoidal (it uses a sine wave). Given that pure sine waves rarely sit well in complex arrangements, you can control the amount of (and balance between) the generated and original signals using the Dry and Wet sliders.

You define the two frequency bands (which the SubBass uses to generate tones) with the High and Low parameters. High Center and Low Center define the center frequency of each band, and High Bandwidth and Low Bandwidth define the bandwidth of each band.

The High Ratio and Low Ratio knobs define the amount that the generated signal is transposed for each band. This is expressed as a ratio of the original signal. As an example; Ratio = 2 transposes the signal down one octave.

Important: Within each frequency band, the filtered signal should have a reasonably stable pitch in order to be analyzed correctly.

In general, narrow bandwidths produce the best results, because they avoid unwanted intermodulations. Set High Center a fifth higher than Low Center, which means a factor of 1.5 for the center frequency. Derive the sub-bass to be synthesized from the existing bass portion of the signal, and transpose by one octave in both bands (Ratio = 2). Do not overdrive the process or you will introduce distortion. If you hear frequency gaps, move one or both Center frequency knobs, or widen the Bandwidth (of one or both frequency ranges) a little.

Tip: Be prudent when using the SubBass, and compare the extreme low frequency content of your mixes with other productions. It is very easy to go overboard with it.

Utility 14

The Utility plug-ins are handy tools that can help you with routine tasks and situations that you may encounter when producing music.

This includes the following tasks:

- Adjusting the input format of a channel (see "Down Mixer" on page 199).
- Adjusting the level or phase of input signals (see "Gain" on page 200 and "Multichannel Gain" on page 202).
- Integrating external audio effects into Logic Pro (see "I/O" on page 201).
- Generating a static frequency or sine sweep (see "Test Oscillator" on page 202).

Down Mixer

You can use the Down Mixer plug-in to adjust the input format of the surround master channel strip. This allows you to quickly check the surround mix in stereo, for example.

Channel mapping, panning, and mixing is handled "behind the scenes." You do, however, have some control over the mix:



- *Destination Format*: Choose the desired surround format from the menu, which includes To Stereo, To Quad, To LCRS, and others.
- *Generic level sliders*: These sliders control the respective channel levels. The number, and names, of sliders is dependent on the chosen plug-in format.

Gain

Gain lets you amplify (or reduce) the signal by a specific decibel amount. It is very useful when you are working with automated tracks during post-processing and want to quickly adjust levels. As examples: when you have inserted another effect that doesn't have its own gain control, or when you want to change the level of a track for a remix version.

Gain Parameters



- Gain slider and field: Sets the amount of gain.
- *Phase Invert Left and Right buttons:* When selected, inverts the phase of the left and right channels, respectively.
- Balance knob: Adjusts the balance of the incoming signal between the left and right channels.
- Swap L/R (Left/Right) button: When selected, swaps the left and right output channels. The swapping occurs after the Balance in the signal path.
- *Mono button:* When selected, outputs the summed mono signal on both the left and right channels.

Note: The Gain plug-in is available in $m \to m$, $m \to s$ and $s \to s$ configurations. In $m \to m$ and $m \to s$ modes, only one Phase Invert button is available. In the $m \to m$ version, the Stereo Balance, Swap Left/Right, and Mono parameters are disabled.

Using Phase Inversion

Inverting phase lets you combat time alignment problems, particularly those caused by recording with multiple microphones at the same time. When you invert the phase of a signal heard in isolation, it sounds identical to the original. When the signal is heard in conjunction with other signals, however, phase inversion has an audible effect. As an example, if you place microphones above and below a snare drum, you should invert the phase of the bottom microphone signal, so that it is in-phase with the top microphone signal.

I/O

The I/O plug-in allows you to use external audio effect units in a similar way to using the internal Logic Studio effects. This only makes sense if you are using an audio interface which provides discrete inputs and outputs (analog or digital), used to send signals to and from the external audio effect unit.

I/O Parameters



- Output Volume field and slider: Adjusts the volume of the output signal.
- Output menu: Assigns the respective output (or output pair) of your audio hardware to the plug-in.
- *Input menu*: Assigns the respective input (or input pair) of your audio hardware to the plug-in
- *Input Volume field and slider:* Adjusts the volume of the input signal.

Using the I/O Plug-in

The following section outlines the steps you should follow to integrate external audio effects in the Logic Pro signal path.

To integrate and use an external effects unit in Logic Pro:

- 1 Connect an output (or output pair) of your audio interface with the input (pair) on your effects unit.
- 2 Connect the output (or output pair) of your effects unit with an input (pair) on your audio interface.

Note: These can be either analog or digital connections if your audio interface and effects unit are equipped with either, or both.

- 3 Click an Insert slot of the channel you want to process with the external effects unit, and choose Utility > I/O.
- 4 In the I/O plug-in window, choose both the Output and Input (shown as numbers) that your effects unit is connected to.
- 5 Adjust the Input or Output volume as required.

When you start playback, the signal of the audio channel will be processed by the external effects unit.

Chapter 14 Utility 201

Multichannel Gain

The Multichannel Gain lets you control the gain (and phase) of each channel of a surround track or bus independently.



- Master slider: Sets the master gain for the combined channel output.
- Channel gain sliders: Each slider sets the gain for its channel.
- Phase Invert buttons: When selected, the phase of the selected channel is inverted.
- Mute buttons: When selected, the channel is muted from the overall output.

Test Oscillator

The Test Oscillator generates a static frequency or a sine sweep. The latter is a user-defined frequency spectrum tone sweep.

Test Oscillator Parameters



- Waveform buttons: Select the type of waveform to be used for test tone generation.
 - The Square Wave and Needle Pulse waveforms are available as either aliased or anti aliased versions. The latter when used in conjunction with the Anti Aliased button.
 - Needle Pulse is a single needle impulse waveform.
 - If the Sine Sweep button is active, the fixed oscillator settings in the Waveform section above are disabled.
- Frequency: Determines the frequency of the oscillator (default is 1 kHz).
- Level: Determines the overall output level of the Test Oscillator.

- Sine Sweep button: Activate to generate a user-defined frequency spectrum sine wave sweep.
- *Time field*: Determines the duration of the sweep.
- Start Freq and End Freq fields: Define the oscillator frequency at the beginning and end of the sine sweep.
- Sweep Mode (extended parameter): Choose either a Linear or Logarithmic sweep curve.
- *Trigger button: Click to trigger the sine sweep.* The behavior of the Trigger button can be switched via the menu below:
 - Single: Clicking the Trigger button triggers the sweep once.
 - Continuous: Clicking the Trigger button triggers the sweep indefinitely.

Using the Test Oscillator

If you insert the Test Oscillator into an Insert slot of an audio channel, you must pass audio through this channel, in order to generate a signal.

To use the Test Oscillator in an audio channel's Insert slot:

- 1 Place any audio region onto a track.
- 2 Insert the Test Oscillator into this track's channel, and start playback.

You can also insert the Test Oscillator plug-in in the Instrument slot of instrument channel strips. The Test Oscillator starts generating the test signal as soon as it is inserted. You can switch it off by bypassing it.

Chapter 14 Utility 203

The EVOC 20 PolySynth combines a vocoder with a polyphonic synthesizer, and can be played in real time.

The EVOC 20 PolySynth is a sophisticated vocoder, equipped with a polyphonic synthesizer, and capable of receiving MIDI note input. This allows the EVOC 20 PolySynth to be played, resulting in classic vocoder choir sounds, for example. Single notes and chords played with the polyphonic EVOC 20 PolySynth will sing with the articulation of the analysis audio source. During this process, the sonic characteristics and changes of the audio signal arriving at the analysis input (the audio track selected as a side chain) are imposed onto the output signal of the integrated synthesizer (the Synthesis section).

The signal path of the EVOC 20 PolySynth is shown in the block diagram on page 224.

Vocoder Basics

If you are new to vocoders you should read this section. It provides you with basic knowledge about vocoders and their functionality. You will also find tips on using vocoders, and achieving good speech intelligibility.

What Is a Vocoder?

The word vocoder is an abbreviation for VOice enCODER. A vocoder analyses and transfers the sonic character of the audio signal arriving at its analysis input to the audio signal present at its synthesis input. The result of this process is heard at the output of the vocoder.

The classic vocoder sound uses speech as the analysis signal and a synthesizer sound as the synthesis signal. This classic sound was popularized in the late 70s and early 80s. You'll probably know it from tracks such as "O Superman" by Laurie Anderson, "Funky Town" by Lipps Inc. and numerous Kraftwerk pieces—from "Autobahn" and "Europe Endless" up to "The Robots" and "Computer World."

Away from these "singing robot" sounds, vocoding has also been used in many films. As examples: the Cylons in Battlestar Galactica, and most famously, on the voice of Darth Vader from the Star Wars saga.

Vocoding, as a process, is not strictly limited to vocal performances. You could use a drum loop as the analysis signal to shape a string ensemble sound arriving at the synthesis input.

How Does a Vocoder Work?

The speech *analyzer* and *synthesizer* referred to above are actually two *filter banks* of bandpass filters. Bandpass filters allow a frequency band (a slice) in the overall frequency spectrum to pass through unchanged, and cut the frequencies which fall outside of the band's range.

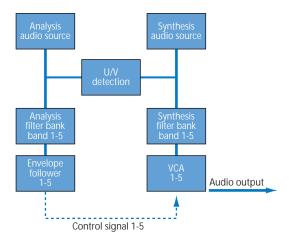
In the EVOC 20 plug-ins, these filter banks are named the Analysis and Synthesis sections. These filter banks have a matching number of corresponding bands—if the analysis filter bank has five bands (1, 2, 3, 4, and 5), there will be a corresponding set of five bands in the synthesis filter bank. Band 1 in the analysis bank is matched to band 1 in the synthesis bank, band 2 to band 2, and so on.

The audio signal arriving at the analysis input passes through the analysis filter bank where it is divided into up to 20 bands.

An envelope follower is coupled to each filter band. The envelope follower of each band tracks (follows) any volume changes in the portion of the audio source allowed to pass by the associated bandpass filter. In this way, the envelope follower of each band generates dynamic control signals.

These control signals are then sent to the synthesis filter bank where they control the levels of the corresponding synthesis filter bands. This is done via VCAs—Voltage Controlled Amplifiers. This allows the volume changes of the bands—and thus the changes of the original sound—in the analysis filter bank to be imposed on the matching bands in the synthesis filter bank.

The more bands a vocoder offers, the more precisely the original sound's character will be re-modeled.



How Does a Filter Bank Work?

If you removed all circuits responsible for transferring the sonic characteristics from the analysis to the synthesis signal from a vocoder, and dispensed with the detection of voiced or unvoiced signals, you'd be left with two filter banks—the analysis and synthesis filters. To use these musically, you would need to ensure that you could control the output level of each bandpass filter. With this level of control, you can apply unique and dramatic changes to the frequency spectrum.

Using the EVOC 20 PolySynth

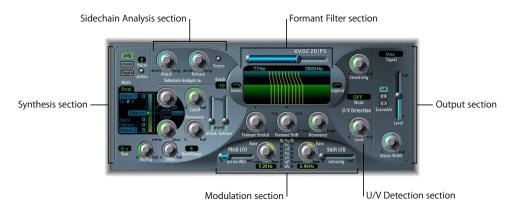
To make use of the EVOC 20 PolySynth, you need to insert the EVOC 20 PolySynth into an instrument channel's Instrument slot, and you also need to provide an audio signal as the analysis audio source.

You can do this by following these steps:

- 1 Select or create a new audio track in the Arrange window.
- 2 Insert (or record) an audio file—use a vocal part to start with—onto this audio track.
- Tip: It may be worthwhile setting up a Cycle area in the Arrange window, allowing you to continually cycle the audio part. This will make experimentation easier.
- 3 Insert the EVOC 20 PolySynth into the Instrument slot of an instrument channel.
- 4 In the Side Chain menu of the EVOC 20 PolySynth, choose the audio track that contains the audio file.
- 5 Ensure that the corresponding instrument track is selected in the Arrange window. The EVOC 20 PolySynth is now ready to accept incoming MIDI data, and has been assigned to see the output from the selected audio track via a Side Chain.
- 6 In the Mixer, mute the audio track (the vocal track) serving as the side chain input.
- 7 Start playback.
- 8 As the audio file is playing back, play your MIDI keyboard.
- 9 In the Mixer, adjust the volume levels of the EVOC 20 PolySynth and the audio track used for the Side Chain to taste.
- 10 Do a little experimentation with the knobs, sliders, and other controls. Have fun, and feel free to insert further effect plug-ins on the channel or busses to further enhance the sound.

EVOC 20 PolySynth Parameters

The EVOC 20 PolySynth interface is divided into six main sections.



- *Synthesis section:* Controls the polyphonic synthesizer of the EVOC 20 PolySynth. See "Synthesis Parameters" on page 210.
- Sidechain Analysis section: The parameters in this section define how the EVOC 20 PolySynth reacts to the analysis signal. See "Sidechain Analysis Parameters" on page 215.
- Formant Filter section: Configures the analysis and synthesis filter banks. See "Formant Filter Parameters" on page 217.
- Modulation section: The Modulation section offers two LFOs to control the Formant Shift and Pitch parameters of the EVOC 20 PolySynth. See "Modulation Parameters" on page 219.
- *U/V Detection section:* Detects the unvoiced portions of the sound in the analysis signal, improving speech intelligibility. See "Unvoiced/Voiced (U/V) Detection" on page 221.
- *Output section:* Configures the output signal of the EVOC 20 PolySynth. "Output Parameters" on page 223.

Synthesis Parameters

The EVOC 20 PolySynth is equipped with a polyphonic synthesizer. It is capable of accepting MIDI note input. The parameters of the Synthesis section are described below.

Mode Buttons

These buttons determine the number of voices used by the EVOC 20 PolySynth:



• When Poly is selected, the maximum number of voices is set via the numeric field alongside the Poly button.

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

- When Mono or Legato is selected, the EVOC 20 PolySynth 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).
- The Unison button enables or disables Unison mode. In this mode, each
 EVOC 20 PolySynth voice is doubled, which will cut polyphony in half (to a maximum
 of 8 voices) as indicated by the numeric Voices field. The doubled voices are detuned
 by the amount defined with the Analog parameter.
- In Unison-Mono mode (both the Unison and Mono or Legato buttons are active), up to 16 voices can be stacked and played monophonically. In this mode, the Voices field displays the number of stacked voices that sound at the same time.

Warning: Stacking voices in Unison-Mono mode will increase the EVOC 20 PolySynth's output volume. To avoid overloading the instrument channel output, adjust the EVOC 20 PolySynth's Level slider accordingly.

Oscillator Section

The EVOC 20 PolySynth is equipped with a two oscillator digital synthesizer which features a number of waveforms, and FM (Frequency Modulation). Further to these sound-generators in the Synthesis section is an independent noise generator.

Click here to switch between Dual and FM mode





There are two oscillator modes.

- *Dual*: Two oscillators make use of single-cycle digital waveforms to provide the synthesis sound source(s).
- FM: A two operator FM engine, with Oscillator 1 as a sine wave carrier, and Oscillator 2 as the modulator. Oscillator 2 can use any of the single-cycle digital waveforms.

You can switch between Dual and FM modes by clicking on the Dual or FM label(s) to the top left of the section shown in the screenshots above.

As you can see, there are some subtle differences between the two modes. This section will cover the common parameters first, and will then explain the mode-specific options.

Wave 1 Parameters

The footages below the Wave 1 label in both modes harks back to the days of pipe organs. The longer the pipe, the deeper the tone. This also applies to Wave 1. Simply click on the 16, 8, or 4 foot value to select the range in which Wave (oscillator) 1 functions. Your selection will be illuminated.

The numerical value beside the Wave 1 label indicates the currently selected waveform type. The EVOC 20 PolySynth features 50 waveforms with different sonic characteristics.

To switch between waveforms, do one of the following:

- Click-hold on the numerical waveform field and drag up or down.
 When the desired waveform number is visible, release the mouse button.
- Double-click the numerical field and input the desired value.

Note: When in FM mode, the waveform of Wave 1 is a fixed sine wave. The waveform parameter of Wave 1 does not have an effect in this mode.

Wave 2 Parameters

The numerical value beside the Wave 2 label indicates the currently selected waveform type. The EVOC 20 PolySynth features 50 single-cycle digital waveforms with different sonic characteristics.

To switch between waveforms, do one of the following:

- Click-hold on the numerical waveform field and drag up or down.
 - When the desired waveform number is visible, release the mouse button.
- Double-click the numerical field and input the desired value.

Noise Parameters

The noise generator provides a further sound source which can be used in addition to the two oscillators (Wave 1 and Wave 2).



- Level knob: Controls the amount of noise added to the signals of the two oscillators.
- Color knob: Controls the timbre of the noise signal. When the Color knob is turned fully-left, the noise generator creates a pure white noise. When turned fully-right, it generates blue noise (high-passed noise). White noise has always been used to create wind and rain sound effects. It has the same energy in each frequency interval. Blue noise sounds brighter, because its bass portion is suppressed by a highpass filter.

It is important to note that the noise generator in the Oscillator section is independent of the noise generator in the U/V Detection section. For further information on voiced and unvoiced signals, refer to "Unvoiced/Voiced (U/V) Detection" on page 221.

Tip: Turn Color full-right and Level a tiny bit up to achieve a more lively and fresh synthesis signal.

Dual Mode Parameters

The parameters specific to Dual mode are found in the Wave 2 section, and the Balance slider to the right.



- Semi parameter: Adjusts the tuning of the second oscillator (Wave 2) in semitone steps.
- Detune parameter: Fine-tunes Wave 1 and Wave 2 in cents. 100 cents equals a semitone step. Doing so will detune Wave 1 in conjunction with Wave 2 around the tuning zero point.
- Balance slider: Allows you to blend the two oscillator signals (Wave 1 and Wave 2).

FM Mode Parameters

The parameters specific to the FM mode are found in the Wave 2 section, and the FM Int slider to the right.



- Ratio c(oarse) parameter: Adjusts the coarse frequency ratio of the second oscillator in relation to the first oscillator.
- Ratio f(ine) parameter: Adjusts the fine frequency ratio of the second oscillator in relation to the first oscillator.
- FM Int slider: Determines the intensity of Wave 1's sine wave modulation by Wave 2. Higher FM Int settings will result in a more complex waveform with more overtones.

When combined, the Ratio and FM Int parameters form the resulting complex FM waveform, thus defining the harmonic content.

Tuning and Pitch Parameters



- Analog knob: Simulates the instability of analog circuitry found in vintage vocoders.
 Analog alters the pitch of each note randomly. This behavior is much like that of polyphonic analog synthesizers. The Analog knob controls the intensity of this random detuning.
- Tune: Defines the range of detuning.
- *Glide:* Glide determines the time it takes for the pitch to slide from one note to another (portamento).
- Bend Range: Determines the pitch bend modulation range in semitones.

Cutoff and Resonance



- *Cutoff:* Sets the cutoff frequency of the lowpass filter. As you turn this knob to the left, an increasing number of high frequencies is filtered from the signal.
- Resonance: Turning up Resonance leads to an emphasis of the frequency area surrounding the frequency defined by the Cutoff parameter. The filter is used for rough signal shaping, before the signal is articulated by the vocoding circuits.
- Tip: Set Cutoff as high as possible, and add a little Resonance to achieve a nice, brilliant high-end.

Envelope Parameters



The EVOC 20 PolySynth features an Attack/Release envelope generator used for level control of the Oscillator section.

- Attack slider: Determines the amount of time that it takes for the oscillators of the Synthesis section to reach their maximum level.
- Release slider: Determines the amount of time that it takes for the oscillators of the Synthesis section to reach their minimum level.

Sidechain Analysis Parameters

The parameters in the Sidechain Analysis section control various aspects of the analysis signal. They require precise control to ensure the best possible intelligibility and tracking.



Attack

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

Note: A long Attack time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulate vocoder effect. Set Attack to the lowest possible value to enhance articulation.

Release

The Release parameter determines how quickly each envelope follower (coupled to each analysis filter band) reacts to falling signals. Longer Release times cause the analysis input signal transients to sustain longer at the vocoder's output.

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

Freeze

When enabled, the current analysis sound spectrum is held infinitely. The frozen analysis signal can capture a particular characteristic of the source signal which is then imposed as a complex sustained filter shape on the Synthesis section. While Freeze is enabled, the analysis filter bank ignores the input source, and the Attack and Release parameters have no effect.

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

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

Bands

The Bands parameter determines the number of frequency bands used by the EVOC 20 PolySynth.

The greater the number of bands, the more precisely the sound can be reshaped. As the number of bands is reduced, the source signal's frequency range is divided up into fewer bands—and the resulting sound will be formed with less precision by the synthesis engine.

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

You may find that a good compromise between sonic precision—allowing incoming signals (speech and vocals, in particular) to remain intelligible—and resource usage, is around 10 to 15 bands.

Formant Filter Parameters

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



High and Low Frequency

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

- To adjust the Low Frequency value, simply click-hold on the silver slider to the left of the blue bar, and drag to the right (or left). The value range is 75 to 750 Hz.
- To adjust the High Frequency value, simply click-hold on the silver slider to the right of the blue bar, and drag to the left (or right). The value range is 800 to 8000 Hz.
- To adjust both sliders simultaneously, click on the area between the slider halves (directly on the blue bar) and drag to the left or right.
- You can make changes to the High and Low Frequency values directly by editing the numerical fields below the blue bar.

Lowest and Highest

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

- In the Bandpass setting, the frequencies below/above the lowest/highest bands are ignored for both analysis and synthesis.
- In the Highpass (or Lowpass) setting, all frequencies below the lowest (or above the highest) bands will be considered for analysis and synthesis.

Formant Stretch

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

With Formant Stretch set to 0, the width and distribution of the bands in the synthesis filter bank is equal to the width of the bands in the analysis filter bank. Low values narrow the width of each band, while high values widen them. The control range is from 0.5 to 2 (expressed as a ratio of the overall bandwidth).

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

Formant Shift

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

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

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

Formant Stretch and Formant Shift are also useful if the frequency spectrum of the synthesis signal does not complement the frequency spectrum of the analysis signal. You could create a synthesis signal in the high frequency range from an analysis signal that mainly modulates the sound in a lower frequency range, for example.

Resonance

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

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

Modulation Parameters



The Modulation section offers two LFOs to control the Formant Shift and Pitch parameters of the EVOC 20 PolySynth. The LFOs can run free or synchronized to the project's tempo.

- *Pitch LFO*: Controls Pitch modulation (Vibrato) of the built-in synthesizer's oscillators. It is hardwired to accept data from the modulation wheel of your MIDI keyboard (or corresponding MIDI data) to control modulation intensity.
- *Shift LFO:* Controls the Formant Shift parameter of the synthesis filter bank to produce dynamic phasing-like effects.

Wave Buttons



These buttons in this section allow you to select the waveform type used by Pitch LFO and Shift LFO. A selection of Triangle, falling and rising Sawtooth, Square up and down around zero (bipolar, good for trills), Square up from zero (unipolar, good for changing between two definable pitches), a random stepped waveform (S&H), and a smoothed random waveform is available for each LFO.

Intensity and Int via Whl

The Intensity slider controls the amount of Formant Shift modulation by the Shift LFO.



The Int via Whl slider for the Pitch LFO features a multi-part slider. The intensity of LFO pitch modulation can be controlled by the modulation wheel of an attached MIDI keyboard. The upper half of the slider determines the intensity when the modulation wheel is set to its maximum value, and the lower half when set to its minimum value. By clicking and dragging in the area between the two slider segments, you can simultaneously move both.

Rate Knobs

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

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

Unvoiced/Voiced (U/V) Detection

Human speech consists of a series of voiced sounds (tonal sounds) and unvoiced sounds (noisy sounds). The main distinction between voiced and unvoiced sounds is that voiced sounds are produced by an oscillation of the vocal cords, while unvoiced sounds are produced by blocking and restricting the air flow with lips, tongue, palate, throat, and larynx.

Should speech containing voiced and unvoiced sounds be used as a vocoder's analysis signal, but the synthesis engine doesn't differentiate between voiced and unvoiced sounds, the result will sound rather toothless. To avoid this, the synthesis section of the vocoder must produce different sounds for the *voiced* and *unvoiced* parts of the signal.

For that reason, the EVOC 20 PolySynth includes an Unvoiced/Voiced detector. This unit detects the unvoiced portions of the sound in the analysis signal and then substitutes the corresponding portions in the synthesis signal with Noise, a mixture of Noise + Synth, or with the original signal (Blend). If the U/V Detector detects voiced parts, it passes this information to the Synthesis section, which uses the normal synthesis signal for these portions.

Tip: Speech intelligibility is highly dependent on high frequency content, as human hearing is reliant on these high frequencies to determine syllables within words. Bear this fact in mind when using the EVOC 20 PolySynth, and take care with filter frequency settings in the Synthesis and Formant Filter sections. To aid intelligibility, it may be worthwhile using equalization to boost particular frequencies in the mid to high frequency range, before processing the signal with the EVOC 20 PolySynth. Please see the "Tips for Better Speech Intelligibility" on page 224 for further information.



Sensitivity

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

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

Mode

Mode selects the sound sources that can be used to replace the unvoiced content of the input signal. Possible settings are Off, Noise, Noise + Synth, or Blend.

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

Level

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

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

Output Parameters

This section covers the various parameters available in the EVOC 20 PolySynth output section.



Signal

This menu offers the choice of Voc(oder), Syn(thesis), and Ana(lysis). These settings allow you to determine the signal that you wish to send to the EVOC 20 PolySynth main outputs. To hear the vocoder effect, the Signal parameter should be set to Voc. The other two settings are useful for monitoring purposes.

Ensemble

The three Ensemble buttons switch the ensemble effects on or off. Ensemble I is a special chorus effect. Ensemble II is a variation, creating a fuller and richer sound by using a more complex modulation routine.

Level

The Level slider controls the volume of the EVOC 20 PolySynth output signal.

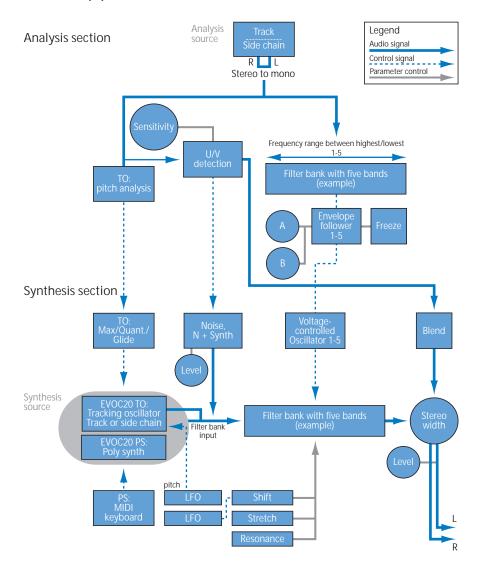
Stereo Width

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

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

Block Diagram

This block diagram illustrates the signal path in the EVOC 20 TrackOscillator and EVOC 20 PolySynth.



Tips for Better Speech Intelligibility

The classic vocoder effect is very demanding, with regard to the quality of *both* the *analysis* and *synthesis* signals. Furthermore, the vocoder parameters need to be set carefully. Following, are some tips on both topics.

Editing the Analysis and Synthesis Signals

The following section outlines how you can edit the analysis and synthesis signals to achieve better speech intelligibility.

Compressing the Analysis Signal

The less the level changes, the better the intelligibility of the vocoder. You should therefore compress the analysis signal in most cases.

Enhancing High Frequency Energy

The vocoder, in a way, always generates the intersection point of the analysis and synthesis signals. To explain: If there's no treble portion in the analysis signal, the resulting vocoder output will also lack treble. This is also the case when the synthesis signal features a lot of high frequency content. This is true of each frequency band. As such, the vocoder demands a stable level in *all* frequency bands from *both* input signals, in order to obtain the best results.

Due to the way human beings hear, the intelligibility of speech is highly dependent on the presence of high frequency content. To aid in keeping speech clear, it may be worthwhile using equalization to boost or cut particular frequencies in analysis signals before processing them with the vocoders.

If the analysis signal consists of vocals or speech, a simple shelving filter should be sufficient. It doesn't require much processing power, and efficiently boosts the highmid and treble range, which is so important for speech intelligibility.

If the *synthesis* signal lacks treble energy, it can be generated with a distortion effect. The Overdrive effect is perfect for this purpose (see "Overdrive" on page 56).

Avoiding Sonic Artifacts

A common problem with vocoder sounds are sudden signal interruptions (ripping, breaking sounds) and rapidly triggered noises, during speech pauses.

Release Parameter in the Analysis Section

The Release parameter defines the speed that a given synthesis frequency band can decrease in level, if the signal level of the respective analysis band decreases abruptly. The sound is smoother when the band levels decrease slowly. To achieve this smoother character, use higher Release values in the analysis section of the interface. Longer release times result in a washy sound.

Short Attack values are no problem. They may, in fact, even be desirable when a fast reaction to impulse signals by the vocoder is required.

Gating Background Noises in the Analysis Signal

If the analysis signal is compressed, as recommended, the level of breath, rumble, and background noises will rise. These background noises can cause the vocoder bands to open, but this is normally not intended. In order to eliminate these noises, it's therefore a good idea to employ a noise gate before compression and treble boosting. If the analysis signal is gated appropriately, you may find that you want to reduce the Analysis Release value.

When gating speech and vocals, the Hysteresis parameter is important. Threshold defines the level, above which the gate will open. Hysteresis defines a lower Threshold level, under which the gate will close. The value is relative to the Threshold level.



The graphic above shows a Threshold setting, which is well-adapted to compressed speech. Unwanted triggering by low or high frequency noise is avoided by the Noise Gates' dedicated sidechain filters. The Hold, Release, and Hysteresis values shown are typical level envelopes, suitable for most vocal and speech signals.

Achieving the Best Analysis and Synthesis Signals

For good speech intelligibility, please keep these points in mind:

- The spectra of the analysis and synthesis signals should overlap almost completely. Low male voices with synthesis signals in the treble range do not work well.
- The synthesis signal must be constantly sustained, without breaks. The track should be played legato, as breaks in the synthesis signal will stop the vocoders output. Alternatively, the Release parameter of the synthesis signal (not to be confused with the Release time of the analysis section) can be set to a longer time. Nice effects can also be achieved by the use of a reverberation signal as a synthesis signal. Note that the two latter methods can lead to harmonic overlaps.
- Do not overdrive the vocoder. This can happen easily, and distortion will occur.
- Enunciate your speech clearly, if the recording is to be used as an analysis signal. Spoken words, with a relatively low pitch, work better than sung vocals—even if the creation of vocoder choirs is your goal! Pronounce consonants well. A nice example is the rolled *R* of "We are the Robots," by Kraftwerk, a classic vocoder track. This pronunciation was specifically made to cater to the demands of the vocoder.

Feel free to do what you like when setting the Formant parameters. The intelligibility of speech is surprisingly little affected by shifting, stretching, or compressing the formants. Even the number of frequency bands used has a minimal impact on the quality of intelligibility. The reason for this is our ability to intuitively differentiate the voices of children, women, and men, whose skulls and throats vary vastly by nature. Such physical differences cause variations in the formants which make up their voices. Our perception (recognition) of speech is based on an analysis of the *relationships* between these formants. In the EVOC 20 plug-ins, these stay intact, even when extreme formant settings are used.

Vocoder History

You may be surprised you to learn that the voder and vocoder date back to 1939 and 1940, respectively.

Homer Dudley, a research physicist at Bell Laboratories, New Jersey (USA) developed the Voice Operated reCOrDER as a research machine. It was originally designed to test compression schemes for the secure transmission of voice signals over copper phone lines.

It was a composite device consisting of an analyzer and an artificial voice synthesizer. These were the:

- Parallel bandpass vocoder: A speech analyzer and resynthesizer, invented in 1940.
- Vocoder speech synthesizer: A voice model played by a human operator, invented in 1939. This valve-driven machine had two keyboards, buttons to recreate consonants, a pedal for oscillator frequency control, and a wrist-bar to switch vowel sounds on and off.

The analyzer detected the energy levels of successive sound samples, measured over the entire audio frequency spectrum via a series of narrow band filters. The results of this analysis could be viewed graphically as functions of frequency against time.

The synthesizer reversed the process by scanning the data from the analyzer and supplying the results to a number of analytical filters, hooked up to a noise generator. This combination produced sounds.

The Voder was demonstrated at the 1939 World Fair, where it caused quite a stir. In World War II, the vocoder (now called VOice enCODER) proved to be of crucial importance, scrambling the transoceanic conversations between Winston Churchill and Franklin Delanore Roosevelt.

Werner Meyer-Eppler, the director of Phonetics at Bonn University, recognized the relevance of the machines to electronic music after Dudley visited the University in 1948. Meyer-Eppler used the vocoder as a basis for his future writings which, in turn, became the inspiration for the German "Elektronische Musik" movement.

In the 1950s, a handful of recordings ensued.

In 1960, the Siemens Synthesizer was developed in Munich. Among its many oscillators and filters, it included a valve-based vocoding circuit.

In 1967, a company called Sylvania created a number of digital machines that used time-based analysis of input signals, rather than bandpass filter analysis.

In 1971, after studying Dudley's unit, Bob Moog and Wendy Carlos modified a number of synthesizer modules to create their own vocoder for the *Clockwork Orange* sound track.

Peter Zinovieff's London-based company "EMS" developed a standalone—and altogether more portable—vocoder. EMS are probably best known for the "Synthi AKS" and VCS3 synthesizers. The EMS Studio Vocoder was the world's first commercially available machine, released in 1976. It was later renamed the EMS 5000. Among its users were Stevie Wonder and Kraftwerk. Stockhausen, the German "Elektronische Musik" pioneer, also used an EMS vocoder.

Sennheiser released the VMS 201 in 1977, and EMS released the EMS 2000, which was a cut-down version of its older sibling.

1978 saw the beginning of mainstream vocoder use, riding on the back of popularity created through the music of Herbie Hancock, Kraftwerk, and a handful of other artists. Among the manufacturers who jumped into vocoder production at this time are: Synton/Bode, Electro-Harmonix, and Korg, with the VC-10.

In 1979, Roland released the VP 330 ensemble/vocoder keyboard.

The late 70s and early 80s were the heyday of the vocoder. Artists who used them included: ELO, Pink Floyd, Eurythmics, Tangerine Dream, Telex, David Bowie, Kate Bush, and many more.

On the production side, vocoders could (and can still) be picked up cheaply in the form of kits from electronics stores.

From 1980 through to the present, EMS in the UK, Synton in Holland and PAiA in the USA were, and remain, the main flyers of the vocoding flag.

In 1996, Doepfer in Germany and Music and More joined the vocoder-producing fraternity.

Throughout the 1990s, a number of standalone software-based vocoders have appeared.

The 16-voice EFM1 is a simple, but powerful, frequency modulation synthesizer.

It produces the typically rich bell and digital sounds that frequency modulation (FM) synthesis has become synonymous with.



At the core of the EFM1 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 EFM1 is divided into several areas.

- The top section contains the global Transpose, Tune, Glide, Voices, and Unison parameters.
- The FM engine consists of the Modulator and Carrier parameters (raised, darker sections), and the FM controls, including the modulation envelope and LFO shown in the mushroom-shaped area in the center.

- The bottom section houses the Output section, and features the Sub Osc Level and Stereo Detune parameters, plus the volume envelope, Main Level, and Velocity controls. A Randomize field is shown to the lower right.
- The extended parameters panel (accessed by clicking the disclosure triangle at the lower left) allows you to assign MIDI controllers to the FM Amount (FM depth, in other words) and Vibrato parameters.

Global Parameters

These parameters impact on the overall instrument sound produced by the EFM1.

Transpose

The base pitch is set with the Transpose parameter. You can transpose the EFM1 by ± 2 octaves.

Tune

Tune will fine-tune the FFM1 + 50 cents. A cent is 1/100th of a semitone.

Unison

Enabling the Unison button will layer two complete EFM1 voices, making the EFM1 sound larger and fatter. In Unison mode, the EFM1 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 to 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 to 16).

Randomize

The Randomize facility (shown at the lower right of the interface) generates new sounds with each click on the Randomize button. The amount of randomization (or variance from the original sound) is determined by the value set in the numeric field. You should use values below 10% if you only want to randomly tweak the current sound.

Modulator and Carrier

The modulator and carrier parameters are outlined below.

Harmonic

In FM synthesis, the basic overtone structure is determined by the tuning relationship of the modulator and volume envelope. This is often expressed as a tuning ratio. In the EFM1, 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 volume envelope to any of the first 32 harmonics. The tuning relationship (or ratio) greatly changes the base sound of the EFM1, 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 volume envelope 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 volume envelope 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 like the waveform selector of an analog synthesizer.

Fine

Fine tune adjusts the tuning 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.

Modulator Wave

In classic FM synthesis, sine waves are used as modulator and carrier waveforms. To extend its sonic capabilities, the EFM1 modulator provides a number of additional digital waveforms.

When turned completely to the left, 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.

Fixed Carrier Button

This button allows you to disconnect the carrier frequency from keyboard, pitch bend, and LFO modulations.

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FM Parameters

These parameters affect the frequency modulation aspects of the EFM 1.

FM (Intensity)

The modulator oscillator modulates the volume envelope 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 EFM1 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.

The Output Section

The EFM1 provides several level controls, as discussed below.

Sub Osc Level

For added bass response, the EFM1 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 EFM1's FM engine output.

Stereo Detune

Stereo Detune adds a rich and diverse chorus-like effect to the sound of the EFM1. This is achieved by doubling the EFM1 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.

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.

Main Level

The Main Level control adjusts the overall output level of the EFM 1. Turning it clockwise makes the EFM1 output louder. Turning it counter clock-wise will decrease the output level.

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Velocity

The EFM1 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 EFM1 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 EFM1 to respond to velocity. Turning the control clockwise will increase velocity sensitivity, and with it, dynamic changes to the sound that the EFM1 is able to produce.

MIDI Controller Assignments

The EFM1 extended parameter section allows you to assign any MIDI controller to the following parameters:

- FM intensity
- Vibrato

Simply choose the desired controller in the Ctrl FM and Ctrl Vibrato menus, and set the modulation or vibrato amount with the sliders below the menus.

Note: The EFM 1 also responds to pitch bend data: Pitch bend is hardwired to the overall pitch of the EFM1.

ES E 17

This chapter discusses the eight-voice polyphonic ES E synthesizer.

The ES E (ES Ensemble) is designed for pad and ensemble sounds. It is great for adding atmospheric beds to your music, with minimal CPU overhead. All ES E parameters are discussed in the following section.



- 4, 8, 16 buttons: Determine the ES E's octave transposition.
- Wave knob: The left-most setting of the Wave parameter causes the oscillators to
 output sawtooth signals—which can be frequency modulated 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 knob: If Wave is set to sawtooth, this parameter defines the amount of frequency modulation, resulting in a vibrato or siren effect—depending on the 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 ("breaking up"). Given this potential artifact, 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 knob: Controls the frequency of the pitch (sawtooth), or pulse width modulation.
- *Cutoff knob:* Sets the cutoff frequency of the resonance-capable, dynamic lowpass filter.

- Resonance knob: Sets the resonance of the ES E's dynamic lowpass filter.
- AR Int knob: The ES E features one simple envelope generator per voice—offering an Attack and a Release parameter. The AR Int parameter defines the amount of cutoff frequency modulation (applied by the envelope generator).
- *Velo Filter knob*: Sets the velocity sensitivity of the cutoff frequency modulation (applied by the envelope generator). This parameter has no effect if AR Int is set to 0.
- Attack slider: Sets the attack time of the envelope generator.
- Release slider: Sets the release time of the envelope generator.
- *Velo Volume knob:* Determines the amount of velocity sensitivity—with each note being louder, if struck more firmly.
- Volume knob: Sets the output level of the ES E.
- Chorus I, Chorus II, and Ensemble button: Switch on/off any of the three ES E chorus/ensemble effect variations.

ES M 18

The monophonic ES M (ES Mono) is a good starting point if you're looking for bass sounds that punch through your mix.

The ES M compact synthesizer features an automatic fingered portamento mode, making bass slides easy. It also features an automatic filter compensation circuit that delivers rich, creamy basses, even when using higher resonance values. All ES M parameters are discussed in the following section.



- 8, 16, 32 buttons: Set the ES M's octave transposition.
- Glide knob: 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 knob:* Crossfades between a sawtooth wave and a 50% rectangular wave, which is heard one octave lower.
- *Cutoff knob:* Sets the cutoff frequency of the resonance-capable dynamic lowpass filter. Its slope is 24 dB/Octave.
- Resonance knob: Sets the resonance of the dynamic lowpass filter. Increasing the
 Resonance value results in a rejection of bass (low frequency energy) when using
 lowpass filters. The ES M compensates for this side-effect internally, resulting in a
 more bassy sound.
- *Int knob*: 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) knob:* Sets the decay time of the filter envelope. It is only effective if Int is not set to 0.
- *Velo (Filter) knob*: Determines the velocity sensitivity of the filter envelope. This parameter is only effective if *Int* is not set to 0.
- *Decay (Volume) knob:* Sets the decay time of the dynamic stage. The attack, release, and sustain times of the synthesizer are internally set to 0.
- *Velo (Volume) knob:* Determines the velocity sensitivity of the dynamic stage.
- Vol knob: Sets the master volume of the ES M.
- Overdrive knob: Sets the overdrive/distortion level for the ES M output. Be careful: The overdrive effect significantly increases the output level.
- Bender Range sliders (extended parameters): Set the sensitivity of the pitch bender in semitones.

ES P 19

This chapter introduces you to the 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. All ES P parameters are discussed in the following section.



- 4, 8, 16 buttons: The 4, 8, and 16 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 knob*: 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 lowpass 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 knob:* Controls the rate of the oscillator frequency or cutoff frequency modulation.
- Frequency knob: Sets the cutoff frequency of the resonance-capable dynamic lowpass filter.
- Resonance knob: Sets the resonance of the dynamic lowpass filter. Increasing the
 Resonance value results in a rejection of bass (low frequency energy) when using
 lowpass filters. The ES P compensates for this side-effect internally, resulting in a
 more bassy sound.
- 1/3, 2/3, 3/3 buttons: 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 knob: 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 knob*: The cutoff frequency modulation by the ADSR envelope generator is velocity sensitive. The amount of velocity sensitivity is set by the Velo Filter knob.
- Volume knob: Sets the master volume of the ES P.
- *Velo Volume knob*: Determines the amount of velocity sensitivity, with each note being louder if struck more firmly.
- A (Attack) slider: Determines the attack time of the envelope generator.
- *D (Decay) slider:* Determines the decay time of the envelope generator.
- *S (Sustain) slider:* Determines the sustain level of the envelope generator.
- *R (Release) slider:* Determines the release time of the envelope generator.
- Chorus knob: Sets the intensity of the integrated chorus effect.
- Overdrive knob: Sets the overdrive/distortion level of the ES P output. Caution: The overdrive effect significantly increases the output level.

ES1 20

This chapter introduces the 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.

The ES1 Parameters



2', 4', 8', 16', 32' Buttons

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 "Router" on page 245). Modulating the pulse width with a slowly cycling LFO, for example, allows periodically mutating, fat bass sounds.

Sub

The sub oscillator delivers square waves (one and two octaves below the frequency of the main oscillator), as well as a pulse wave (two octaves below the frequency of the main oscillator). In addition to pure square waves, the waveform switch allows selections between different mixes, and phase relationships of these waves, resulting in different sounds. You can also use white noise, or switch the sub oscillator OFF. You can feed a Side Chain signal (from any track) into the synth's filter (select EXT). You can select the Side Chain source track from the Side Chain menu at the top of the plug-in window.

Mix

This slider defines the mix relationship between the main and sub oscillator signals. When the sub oscillator wave is switched to OFF, its output is completely removed from the mix.

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

This section outlines the filter parameters available to the ES1.

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.

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ADSR via Vel

The main envelope generator (ADSR) modulates the cutoff frequency over the duration of a note. The intensity of this modulation can be set to positive or negative values, and can respond to velocity information. If you play pianissimo (Velocity = 1), the modulation will take place as indicated by the lower arrow. If you strike with the hardest fortissimo (Velocity = 127), the modulation will take place as indicated by the upper arrow. The blue bar between the arrows shows the dynamics of this modulation. You can adjust the modulation range and intensity simultaneously by grabbing the bar and moving both arrows at once. Note that as you do so, they retain their relative distance from one another.

Level via Vel

The upper arrow works like a main volume control for the synthesizer. The greater the distance from the lower arrow (indicated by the blue bars), the more the volume is affected by incoming velocity messages. The lower arrow indicates the level when you play pianissimo (velocity =1). You can adjust the modulation range and intensity simultaneously by grabbing the bar and moving both arrows at once. Note that as you do so, they retain their relative distance from one another. In order to retain the maximum possible resolution for velocity sensitivity, even when set to a low volume, the ES1 has an additional Out Level parameter—available in the Controls view.

Amplifier Envelope Selector

The AGateR, ADSR, and GateR buttons 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 247). A value of 0 disables the *Glide* function.

LFO Waveform

The LFO offers several waveforms: triangle, ascending and descending sawtooth, square wave, sample & hold (random), and a lagged, smoothly changing random wave. You can also assign a Side Chain signal (any audio track) as a modulation source (*EXT*). Choose the Side Chain source track via the Side Chain menu at the top of the plug-in window.

Rate

This defines the speed (frequency) of modulation. If you set values to the left of zero, the LFO phase is locked to the tempo of the project—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

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- 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 nonlinear. 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.

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MIDI Controller List

Controller Number	Parameter Name
12	Oscillator pitch buttons
13	Oscillator waveform
14	Mix slider
15	Waveform of sub oscillator
16	Drive slider
17	Cutoff slider
18	Resonance slider
19	Slope buttons
20	ADSR via Vel: lower slider
21	ADSR via Vel: upper slider
22	Attack slider
23	Decay slider
24	Sustain slider
25	Release slider
26	Key slider
27	Amplifier Envelope Selector buttons
28	Level via Velocity: lower slider
29	Level via Velocity: upper slider
30	Chorus parameter
31	Modulation envelope target
102	Modulation Envelope form slider
103	Modulation envelope: Int via Vel parameter: lower slider
104	Modulation envelope: Int via Vel parameter: upper slider
105	LFO rate
106	LFO waveform
107	LFO modulation target
108	LFO: Int via Whl: lower slider
109	LFO: Int via Whl: upper slider
110	Glide slider
111	Tune parameter
112	Analog parameter
113	Bender Range parameter
114	Out Level parameter
115	Voices parameter

ES2 21

The ES2 synthesizer combines a powerful tone generation system with extensive modulation facilities.

The ES2 provides three oscillators, which can be synchronized and ring-modulated. Pulse-width modulation is also possible. Oscillator 1 can be modulated in frequency by Oscillator 2, and is capable of producing FM-style synthesizer sounds.

In addition to the classic analog synthesizer waveforms, the ES2 oscillators also provide 100 single-cycle waveforms, known as Digiwaves. Each has a totally different sonic color. You can even cross-fade or step between them, using the modulation matrix, resulting in sounds that are reminiscent of wavetable synthesizers.

The modulation matrix (known as the *Router*) can be used in addition to a number of hard-wired modulation routings. The concept of combining any modulation *source* with any modulation *target* (to perform modulation *processes*), is almost as old as the synthesizer itself. Most important to this concept is a huge set of modulation *targets* and *sources*—inserted into modulation *channels*. The ES2 offers 10 Router channels.

Two dynamic multi-mode filters—which can be used in parallel and series—are capable of delivering fat analog synthesizer sounds.

The ES2 Unison mode can be used *both* monophonically and polyphonically, making it easy to convincingly recreate the huge sounds of classic analog synthesizers such as the Roland Jupiter 8, the SCI Prophet V, and Oberheim OB 8.

You can directly mix a sine wave (from Oscillator 1) into the dynamic stage to further fatten the sound.

Dedicated macro controls make it quick and easy to adjust several ES2 sound generation parameters simultaneously. This can be done with the mouse, or by using knobs and sliders on your MIDI keyboard—just like a hardware synthesizer!

Last, but not least, integrated Distortion, Chorus, Phaser, and Flanger effects are built right-in to the ES2.

The ES2 Parameters



If given just a few words to explain the principles behind a subtractive synthesizer, it would go something like this: The oscillator generates the oscillation (or waveform), the filter takes away the unwanted overtones (of the waveform), and the dynamic stage sets the volume of the permanent oscillation (the filtered waveform) to zero—as long as no keyboard note is pressed.

In an analog synthesizer, these three sections are commonly called the VCO, VCF, and VCA, with VC being the abbreviation for Voltage Controlled, and the other letters standing for Oscillator, Filter, and Amplifier, respectively.

The basic parameters of a synthesizer are controlled (modulated) by voltages: pitch in the oscillator, timbre in the filter, loudness in the amplifier.

These voltages are generated by modulation sources. In the ES2, the Router determines which parameter is controlled by which modulation source.

Finally, the synthesizer's sound is refined by effects such as distortion or chorus.

This simple signal path is followed throughout, to introduce you to the ES2's sound generation modules.

Global Parameters

These parameters impact on the overall instrument sound produced by the ES2. You can find the global parameters to the left of the oscillators, and above the filter section. In surround instances, two additional global parameters are shown in the Extended Parameters section.



Tune

Tune sets the pitch of the ES2 in cents. 100 cents equals a semitone step. At a value of 0 c (zero cents), a' is tuned to 440 Hz or concert pitch.

Analog

Analog alters the pitch of each note, plus the cutoff frequency in a random fashion. Much like polyphonic analog synthesizers, all three oscillators used by each synthesizer voice maintain their specific deviation, but are shifted by the same amount randomly. Medium values simulate the tuning instabilities of analog synthesizer circuitry which can be useful in achieving that much sought after warmth of the ES2's analog hardware counterparts.

If the ES2 is set to mono or legato, Analog is only effective with Unison activated. In this situation, Analog sets the amount of detuning between the stacked (unison) voices.

If voices is set to 1 and Unison is not activated, the Analog parameter has no effect.

Read more on these parameters in "Keyboard Mode (Poly/Mono/Legato)" on page 252.

CBD

Fine Tuning detunes Oscillators 1, 2, and 3 in cents (1/100th percentages of a semitone). The detuning results in beats (phasing), the speed of which is determined by the difference between the two oscillator frequencies (given that these frequencies are nearly identical). The higher the pitch, the faster the phasing beats are. High notes may therefore seem to be further out of tune than lower notes.

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The CBD (Constant Beat Detuning) parameter matches this natural effect by detuning the lower frequencies in a ratio proportionate to the upper frequencies. Besides disabling CBD altogether, four values are at your disposal: 25%, 50%, 75%, 100%. If you choose 100%, the phasing beats are (almost) constant across the entire keyboard range. This value, however, may be too high, as the lower notes might be overly-detuned at the point where the phasing of the higher notes feels right. Try lower values for CBD (and detune, of course) in cases where the bass appears to be a little too far out of tune.

The reference pitch for CBD is C3 (middle C): Its detuning stays the same, regardless of the CBD value.

Glide

The Glide parameter controls the portamento time. This is the time it takes for the pitch to travel from one note to another. Glide behavior depends on the value of the Keyboard Mode parameter setting.

If Keyboard Mode is set to Poly or Mono, and Glide is set to a value other than 0, portamento is active. If Keyboard Mode is set to Legato, and Glide is set to a value other than 0, you need to play legato (press a new key while holding the old one) to activate portamento. If you do not play legato, portamento will be inactive. This behavior is also called fingered portamento.

Bend Range

Bend Range determines the pitch range for pitch bend modulation. The range is ± 36 semitones. There are separate range settings for upwards and downwards bends, plus an optional Link mode.

Keyboard Mode (Poly/Mono/Legato)

A *polyphonic* instrument allows several notes to be played simultaneously—as you would with an organ or piano, as examples. Many synthesizers are *monophonic*, especially the older ones. This means that only one note can be played at a time, much like a brass or reed instrument. This shouldn't be viewed as a disadvantage in any way, because it allows playing styles that are not possible with polyphonic keyboard instruments.

You can switch between Monophonic and Polyphonic modes by clicking on the Poly and Mono buttons. Legato is also monophonic, but with one difference: The envelope generators are only retriggered if you play staccato (release each key before playing a new key). If you play legato (press a new key while holding the old one), the envelope generators are only triggered with the first note you play legato, and then continue their curve until you release the last legato played key. If you switch to Mono, legato or staccato playing does not have this impact: The envelope generators are retriggered with every new note that is played.

Note: If you switch to Legato, you need to play legato to actually hear the Glide parameter taking effect.

Note: On several monophonic synthesizers, the behavior in Legato mode is referred to as single trigger, while Mono mode is referred to as multi trigger.

Voices

The maximum number of notes that can be played simultaneously is determined by the Voices parameter. Maximum value for Voices is 32.

The value of this parameter has a significant impact on the computer processing resources demanded by the ES2 when played at its maximum polyphony. Reduce this value to the number of voices that you actually *require* for the part. Setting it to a higher value places higher overheads on the CPU, and wastes resources.

Unison

A forte of polyphonic analog synthesizers has always been unison mode. Traditionally, in unison mode, classic analog polysynths run monophonically, with all voices playing a single note simultaneously. As the voices of an analog synthesizer are never perfectly in tune, this results in an extremely fat chorus effect with great sonic depth. Switch the ES2 to Mono or Legato and switch on Unison in order to achieve and hear this effect. The intensity of the unison effect depends on the number of Voices selected. Remember that the amount of processing power required is correlated to the number of voices! The intensity of detuning (voice deviation) is set via the Analog parameter.

In addition to this classic monophonic unison effect, the ES2 also features a polyphonic unison effect. In Poly/Unison, each note played is effectively doubled, or more correctly, the polyphony value of the Voices parameter is halved. These two voices are then used for the single triggered note.

Switching on Poly and Unison has the same effect as setting the ES2 to Mono, Unison, and Voices = 2, except that it can be played polyphonically.

Osc Start

The oscillators can run freely, or they can begin at the same phase position of their waveform cycle each time you hit a key (every time the ES2 receives a note on message).

- When Osc Start (Oscillator Start) is set to free, the initial oscillator phase startpoint is
 random, with each note played. This gives the sound more life and a less static feel—
 just like an analog hardware synthesizer. On the other hand, the output level may
 differ each time you play a note, and the attack phase may sound less punchy.
- If you set Osc Start to soft, each initial oscillator phase will start at a zero crossing every time a note is played. This mimics the sonic character of a normal digital synthesizer.

• If Osc Start is set to hard, each initial oscillator phase begins at the highest possible level in its waveform cycle every time a note is played. This punch is only audible if the ENV3 Attack Time is set to a minimal value—a very fast attack. This setting is highly recommended for electronic percussion and hard basses.

Note: Osc Start soft and hard result in a constant output level of the initial oscillator phase every time the sound is played back. This may be of particular importance when using the Bounce function of Logic Pro, at close to maximum recording levels.

Surround Range and Diversity

In surround instances, the ES2 extended parameters provide the Surround Range and Surround Diversity parameters:



- Surround Range: Determines the range of the surround angle. This can be from 0 to 360°. Put another way, this determines the breadth of the surround field. You can modulate the movement of sounds—within this surround range—through use of the Diversity target in the Router (see "Diversity" on page 277).
- Surround Diversity: Determines how the output signal is spread across your surround speakers. If you choose a value of 0, only the speakers that are closest to the original signal's position will carry the signal. A diversity value of 1 distributes an identical amount of signal to all speakers.

Oscillator Parameters

The following section describes the parameters that you can set individually for each oscillator. You can find these parameters in the silver area to the right of the ES2 interface.



Muting Oscillators

By clicking on the green numbers to the right of the oscillators, you can mute and unmute them independently. This saves processor power.

Frequency Knobs

The Frequency knobs set the pitch in semitone steps over a range of ± 3 octaves. As an octave consists of 12 semitones, the ± 12 , 24, and 36 settings represent octaves. You can click on these options to quickly set the corresponding octave.

The value display works as follows: the left numbers show the semitone (s) setting, the right numbers show the cent (c, 1 cent = 1/100th semitone) setting. You can adjust these two values independently. As an example: an oscillator with the value 12 s 30 c sounds an octave (12 semitones) and 30 cents higher than an oscillator with the value 0 s 0 c.

Note: The fifth (seven semitones), and all settings that correspond with harmonics of an oscillator set to 0 semitones (19, 28 semitones) result in harmonic sounds.

Wave

Each of the three oscillators features a rotary knob that allows you to select a waveform. This is responsible for the basic harmonic content and tone color of a sound. Oscillators 2 and 3 are almost identical to each other, but differ from Oscillator 1. Oscillator 1 is capable of generating a sine wave, the frequency of which can be modulated in the audio range, for true FM synthesis sounds. Oscillators 2 and 3 can be synchronized to, or be ring-modulated with, Oscillator 1. They also feature rectangular waves with freely definable fixed pulse widths plus pulse width modulation (PWM) facilities. Via the Router, the rectangular and pulse waves of Oscillator 1 can be modulated in width in conjunction with the synchronized and ring-modulated rectangular waves of Oscillators 2 and 3.

Note: The Filter button disables the entire filter section. This makes listening to the pure oscillator waveforms easier.

Oscillator 1 Waveforms

Oscillator 1 outputs standard waveforms—pulse, rectangular, sawtooth, and triangular waves—or, alternately, any of the 155 available Digiwaves. It can also output a pure sine wave.



The sine wave can be modulated in frequency by Oscillator 2 in the audio frequency range. This kind of linear frequency modulation is the basis on which FM synthesis works. FM synthesis was popularized by synthesizers such as Yamaha's DX7 (the architecture of which is much more complex, when it comes to FM synthesis).

A click on the oscillator number disables the output of Oscillator 1.

Note: Even when Oscillator 1 is turned off, it is still available for use as a modulation and synchronization source for Oscillators 2 and 3.

Digiwaves

The ES2 not only features the most popular synthesizer waveforms, but also a selection of 100 additional waveforms, called Digiwaves.

To select a Digiwave:

- Set the Waveform knob to Sine, then do one of the following:
 - Control-click the Sine label, and choose the desired waveform from the pop-up menu.



- Click-hold the Sine label and move the mouse vertically.
- Tip: You can select the Digiwave numerically when you hold Shift.

The number of the Digiwave is a parameter that can be modulated. By modulating the OscWave target, you can scroll through the list of Digiwaves. Choose a sufficiently low modulation intensity and speed to hear the single Digiwaves being crossfaded. The Digiwaves of the three oscillators can be modulated individually or together. The modulation targets are discussed in "OscWaves" on page 274, through to "OscWaveB" on page 275.

Given the ES2's ability to modulate Digiwaves, you can produce sounds reminiscent of classic wavetable synthesizers from PPG and Waldorf (and Korg's Wavestation).

Linear Frequency Modulation

The principle of linear frequency modulation (FM) synthesis was developed in the late sixties and early seventies by John Chowning. It's such a flexible and powerful method of tone generation that synthesizers were developed, based solely on the idea of FM synthesis. The most popular FM synthesizer ever built is Yamaha's DX7. FM synthesis is also found in other models of the Yamaha DX range and some Yamaha E-Pianos. In the discipline of pure FM synthesis, the ES2 can't be compared with these synthesizers, but it can certainly achieve some of their signature sounds.



Between the Sine setting (when the sine symbol is selected) and the FM symbol of the Oscillator 1 knob, a stepless control range—which determines the amount of frequency modulation—is available. This parameter is also available as a modulation address.

Note: Osc1Wave is optimized for subtle FM sounds, utilizing lower FM intensity amounts. For more extreme FM modulations, the Router offers the Osc1WaveB target. See the "Modulation Targets" on page 273.

The frequency of Oscillator 1 can be modulated by the output signal of Oscillator 2. Whenever it outputs a positive voltage, the frequency of Oscillator 1 will increase. Whenever it is negative, its frequency will decrease.

The effect is similar to an LFO modulation, being used to create a vibrato (a periodic modulation of the frequency) or a slow siren effect. In comparison to an LFO, Oscillator 2 does not oscillate slowly. In the audio domain, it oscillates a little faster than Oscillator 1 itself. Thus, Oscillator 1's oscillation also accelerates and slows down within a single phase, resulting in a distortion of the basic sine shape of Oscillator 1. This sine wave distortion has the side benefit of a number of new harmonics becoming audible.

The effect of frequency modulation depends on *both* the modulation intensity and frequency ratio of both oscillators. It also depends on the waveform used by the modulating oscillator (Oscillator 2). The modulation that takes place varies according to the waveform selection for Oscillator 2—it might even be an oscillation that is synchronized to Oscillator 1! Given the 100 available Digiwaves, countless combinations of modulation intensities and frequency ratios, the frequency modulation of the two oscillators delivers an infinite pool of spectra and tone colors.

The Waveforms of Oscillators 2 and 3

Basically, Oscillators 2 and 3 supply the same selection of analog waveforms as Oscillator 1: sine, triangular, sawtooth, and rectangular waves. The pulse width can be scaled steplessly between 50% and the thinnest of pulses, and can be modulated in a number of ways (see the "Pulse Width Modulation" on page 258).

Oscillators 2 and 3 also offer the selection of:

- A rectangular wave, synchronized to Oscillator 1
- A sawtooth wave, synchronized to Oscillator 1
- A ring modulator, which is fed by the output of Oscillator 1 and a square wave from Oscillator 2.
- Colored noise for Oscillator 3

Synchronization and ring modulation facilitate the creation of very complex and flexible harmonic spectra. The principle of Oscillator synchronization is described on page 259, and ring modulation on page 259.

Pulse Width Modulation

Oscillators 2 and 3 allow you to scale the width of the pulses to any value. The spectrum and tone color generated by these oscillators depends on the pulse width. The pulse width can be modulated. You can even modulate the pulse width of the square and pulse wave of Oscillator 1, the pulse width of the synchronized pulse waves of Oscillator 2 and 3, and the square wave of Oscillator 2's ring modulator.



This width modulation is controlled in the Router (the Modulation Matrix). The pulse width is defined by the waveform rotary control. The graphic below shows a pulse wave, with the pulse width modulated by an LFO. You can clearly see how the width of the pulses change over time.

Note: A pulse wave (with its width modulation controlled by an LFO set to a sine wave) makes a single oscillator sound vivid, undulating, and rich with overtones. Sonically, this is somewhat like the sound of two slightly detuned, phasing oscillators. The effect sounds great with sustained bass and pad sounds. Select the intensity and speed of the modulation with care, as the overall volume (and level of the first partial) decreases, and slight detuning occurs, when the pulses become very thin (below 10%). Pulse width modulations via velocity sensitive envelope generators sound very dynamic—a great effect that is especially suitable for percussive bass sounds.

Sync

The rectangular and sawtooth waveforms also feature a Sync option. In this mode, the frequency of Oscillator 2 (or 3, respectively) is synchronized to the frequency of Oscillator 1.



This does not mean that their frequency controls are simply disabled. They still oscillate at their selected frequencies, but every time that Oscillator 1 starts a new oscillation phase, the synchronized oscillator is also forced to restart its phase from the beginning. Between the pulses of Oscillator 1, the synchronized oscillator runs freely.

Synchronized oscillator sounds are especially cool when the frequency of the synchronized oscillator is modulated by an envelope generator. This way, the number of phases within a section (phase) of the synchronization cycle always changes, and so does the spectrum. Typical oscillator sync sounds tend towards the aggressive, screaming leads that synthesizer manufacturers like to talk about.

Ring (Ring Modulation)

The waveform control of Oscillator 2 also features the Ring setting. In this mode, Oscillator 2 outputs the signal of a ring modulator. This ring modulator is fed with the output signal of Oscillator 1 and a square wave of Oscillator 2. The pulse width of this square wave can be modulated.

A ring modulator has two inputs. At its output you will find the sum and difference frequencies of the input signals. If you ring-modulate a sine oscillation of 200 Hz with a sine oscillation of 500 Hz, the output signal of the ring modulator will consist of a 700 Hz (sum) and a 300 Hz (difference) signal. Negative frequencies result in a change to the phase polarity of output signals. With sawtooth and rectangular input signals, the output signal is much more complex, as these harmonically-rich waveforms produce a number of extra side bands.

Note: Ring modulation is a powerful tool for inharmonic, metallic sounds, as the spectra resulting from its use are inharmonic at almost every frequency ratio. The ring modulator was the tool of choice for bell-like sounds, and dates back to the early days of the synthesizer (see "Ringshifter" on page 142).

White and Colored Noise (Oscillator 3 Only)

Unlike Oscillator 2, Oscillator 3 is not capable of producing ring modulated signals or sine waves. Its sonic palette however, is bolstered by the inclusion of a noise generator. By default, Oscillator 3's noise generator generates white noise. White noise is defined as a signal that consists of all frequencies (an infinite number) sounding simultaneously, at the same intensity, in a given frequency band. The width of the frequency band is measured in Hertz. Its name is analogous to white light, which consists of a mixture of all optical wavelengths (all rainbow colors). Sonically, white noise falls between the sound of the consonant *F* and breaking waves (surf). Synthesis of wind and seashore noises, or electronic snare drum sounds, requires the use of white noise.

Oscillator 3 has more up its sleeve than the output of neutral sounding white noise—you also can make it hiss or rumble. Even better, you can modulate this sound color in real-time without using the main filters of the ES2.

If the waveform of Oscillator 3 is modulated (modulation target: Osc3Wave), the color of the noise will change. It can be filtered by a dedicated highpass or lowpass filter with a descending slope of 6 dB/octave. At negative values, the sound becomes darker (red); The lowpass filter can be tuned down to 18 Hz with a setting of –1. When Osc3Wave is modulated positively, the noise becomes brighter (blue): At a value of +1 for Osc3Wave, the highpass filter's cutoff frequency is set to 18 kHz. This filtering of the noise signal is independent of the main filters of the ES2, and can be automatically changed in real-time.

Oscillator Mix Field—the Triangle

By grabbing, and moving the cursor in the Triangle, you can crossfade between the three oscillators. This is pretty self-explanatory. Moving the cursor along one of the Triangle's sides will crossfade between two oscillators, with the third oscillator being muted.



The position of the cursor can be controlled via the vector envelope, just like the cursor position in the Track Pad (the Square), which is covered in "The Square" on page 291.

Note that the vector envelope features a loop function. This addition extends its usefulness, allowing you to view it as a luxurious pseudo-LFO with a programmable waveform. It can be used for altering the positioning of the Triangle and Square cursors. Read more about this in "Vector Mode Menu" on page 292, and "The Vector Envelope" on page 293.

Triangle Values in Control View

Internally, the position of the cursor in the Triangle is described by two parameters (coordinates, actually) which are relevant when it comes to automating the oscillator mix. These parameters, called OscLevelX and OscLevelY—are visible in the Controls view. Don't confuse them with the X and Y positions of the Square.

If you intended to edit a region containing oscillator mix automation data in the Hyper Editor, you would use the following MIDI controller values. Take a look at the information below to get a feel for how they operate.

To listen to Oscillator 1 only:

Set OscLevelX to 1.0 (MIDI: 127) and OscLevelY to 1.154 (MIDI: 127).

To listen to Oscillator 2 only:

Set OscLevelX to 0.0 (MIDI: 0) and OscLevelY to 0.577 (MIDI: 64).

To listen to Oscillator 3 only:

Set OscLevelX to 1.0 (MIDI: 127) and OscLevelY to 0.0 (MIDI: 127).

To listen to Oscillator 1 and 2 only:

Set OscLevelX to 0.5 (MIDI: 64) and OscLevelY to 0.866 (MIDI: 95).

To listen to Oscillator 1 and 3 only:

Set OscLevelX to 1.0 (MIDI: 127) and OscLevelY to 0.577 (MIDI: 64).

To listen to Oscillator 2 and 3 only:

Set OscLevelX to 0.5 (MIDI: 64) and OscLevelY to 0.288 (MIDI: 32).

To set all oscillators to the same level:

Set OscLevelX to 0.667 (MIDI: 85) and OscLevelY to 0.577 (MIDI: 64).

Filters

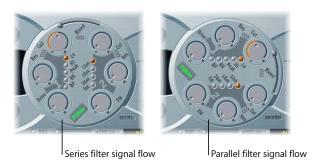
The ES2 features two dynamic filters which are equivalent to the Voltage Controlled Filters (VCF) found in the world of analog synthesizers. The two filters are not identical. Filter 1 features several modes: lowpass, highpass, bandpass, band rejection, peak. Filter 2 always functions as a lowpass filter. Unlike Filter 1, however, Filter 2 offers variable slopes (measured in dB/octave).



The Filter button bypasses (switches off) the entire filter section of the ES2. Disabling the filters makes it easier to hear adjustments to other sound parameters, as the filters always heavily affect the sound. If the word Filter is green, the filters are engaged. Disabling the filters reduces processor load.

Series and Parallel Filter Signal Flow

You can rotate the entire (circular) filter segment of the ES2 user interface. Click the button labelled Parallel or Series to do so. The label and position/direction of the filter controls clearly indicate the current signal flow.



In the graphic displayed to the left, the filters are cabled in series. This means that the signal of the Oscillator Mix section (Triangle) passes through the first filter, and then this filtered signal passes through Filter 2—if Filter Blend is set to 0 (middle position). See "Filter Blend: Cross-Fading the Filters" on page 263 for a detailed description of this parameter.

The mono output signal of Filter 2 is then fed into the input of the dynamic stage (the equivalent of a VCA in an analog synthesizer), where it can be panned in the stereo or surround spectrum, and then fed into the effects processor.

In the graphic to the right, the filters are cabled in parallel. If Filter Blend is set to 0, you'll hear a 50/50 mix of the source signal routed via Filter 1 and Filter 2, which is fed into the mono input of the dynamic stage. There it can be panned in the stereo or surround spectrum, and then fed into the effects processor.

Filter Blend: Cross-Fading the Filters

You can cross-fade between the two filters. When cabled in parallel, you'll find this quite obvious to look at and understand: If Filter Blend is set to its top position, you'll only hear Filter 1. If Filter Blend is set to its lowest position, you'll only hear Filter 2. In between these positions, the filters are cross-faded.



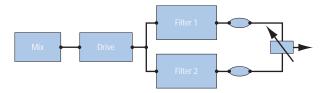
More often, you'll want to cable the filters in series, where you can also cross-fade the filters. This is achieved through the use of controllable side chains (bypassing lines). In this serial cabling scenario, the distortion circuits controlled by the Drive parameter also need to be considered, as the distortion circuits are positioned before, or inbetween the filters, dependent on Filter Blend.

Note: The Filter Blend parameter can be modulated dynamically in the Router!

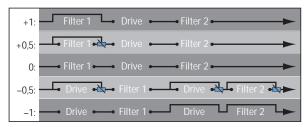
Filter Blend and Signal Flow

No matter whether parallel or series filter configurations are chosen, a Filter Blend setting of -1 results in only Filter 1 being audible. A Filter Blend setting of +1 will limit audibility to Filter 2. This is reflected in the user interface.

In conjunction with the overdrive/distortion circuit (Drive) and a series cabling configuration, the ES2's signal flow is far from commonplace. The graphics illustrate the signal flow between the Oscillator Mix stage (the Triangle) and the dynamic stage, which is always controlled by ENV 3. The signal flow through the filters, the overdrives and the sidechains is dependent on the Filter Blend setting.



Filter Blend in parallel filter mode.



Filter Blend in series filter mode. Between 0 and –1 two distortion circuits are active—one pre each filter. Filter Blend cross-fades up to three bypassing lines simultaneously.

Filter Blend and Series Filter Configuration Tips

- With positive values for Filter Blend, Filter 1 is partially bypassed.
- With negative values for Filter Blend, Filter 2 (the lowpass) is partially bypassed.
- At zero and positive values for Filter Blend, there is only one overdrive circuit for both filters.
- With negative values, another overdrive circuit is introduced, distorting the output signal of the Oscillator Mix stage before it is fed into the first filter—where Filter Blend = -1.

Note: If Drive is set to 0, no distortion occurs.

Filter Blend and Parallel Filter Configuration Tip

The overdrive/distortion circuit is always wired *after* the Oscillator Mix stage, and *before* the filters. The filters receive a mono input signal from the overdrive circuit's output. The outputs of both filters are mixed to mono via Filter Blend.

Note: If Drive is set to 0, no distortion occurs.

Drive

The filters are equipped with separate overdrive modules. Overdrive intensity is defined by the Drive parameter. If the filters are connected in parallel, the overdrive is placed before the filters. If the filters are connected in series, the position of the overdrive circuits depends on the Filter Blend parameter—as described above.



Polyphonic Distortions in the Real World

The ES2 features a *distortion* effect, equipped with a tone control, in the *Effects* section. Given the inclusion of this effect, you may be wondering what benefit the *Drive* function in the Filter section brings?

The distortion circuit in the Effects section affects the sum of the entire polyphonic synthesizer performance. Thus, more complex chords (other than major chords, parallel fifths and octaves) sound rough, when using distortion. Every rock guitarist knows this. Due to these intermodulation distortions, distorted guitar playing is generally performed by using few voices or parallel fifths and octaves.

The Filter Drive affects every voice individually—and when every voice in the ES2 is overdriven individually (like having six fuzz boxes for the guitar, one for each string), you can play the most complex harmonies over the entire keyboard range. They'll sound clean, without unwanted intermodulation effects spoiling the sound.

Furthermore, appropriate Drive parameter settings lead to a different tonal character. To explain, the way analog filters behave when overdriven forms an essential part of the sonic character of a synthesizer. Each synthesizer model is unique with regard to the way its filters behave when overdriven. The ES2 is very flexible in this area, allowing for the most subtle fuzz through to the hardest distortions.

In series mode, the distortion always takes place *before* the lowpass filter (Filter 2). As the lowpass filter (Filter 2) can filter (cut) away the overtones introduced by the distortion, the Drive feature can be seen (and used) as another tool for deforming the oscillator(s) waveforms.

To check out how the overdrive circuit between the filters works, program a sound as follows:

- Simple static waveform (a sawtooth)
- Filter set to Series mode
- Filter Blend set to 0 (center position)

- Filter 1 set to Peak Filter mode
- High Resonance value for Filter 1
- Modulate Cutoff Frequency 1 manually or in the Router.
- Set Drive to your taste.
- Filter away (cut) the high frequencies with Filter 2 to taste.

The sonic result resembles the effect of synchronized oscillators. At high resonance values, the sound tends to scream. Modulate the Resonance of Filter 1, if you wish.

Filter Parameters

This section covers the ES2 filter parameters in detail.

Cutoff and Resonance

With every lowpass filter (in the ES2: Lo mode for Filter 1 and all of Filter 2's modes), all frequency portions above the Cutoff Frequency (Cut) are suppressed, or cut off, hence the name. The Cutoff Frequency controls the brilliance of the signal. The higher the Cutoff Frequency is set, the higher the frequencies of signals that are allowed to pass through the lowpass filter.



Note: The dynamic lowpass filter is the most essential module in any subtractive synthesizer. This is why Filter 2 always operates in lowpass mode.

Resonance (Res) emphasizes the portions of the signal which surround the frequency defined by the Cutoff Frequency value. This emphasis can be set so intensively in Filter 2, that the filter begins to oscillate by itself. When driven to self-oscillation, the filter outputs a sine oscillation (sine wave). This self-oscillation can be supported by the *Filter Reset* parameter. See "Flt Reset" on page 267 for details.

Tip: If you are new to synthesizers, experiment with a simple saw wave, using Oscillator 1, and Filter 2 (lowpass filter, Filter Blend = +1) on its own. Experiment with the Cutoff Frequency and Resonance parameters. You'll quickly learn how to emulate a number of recognizable sounds, and will pick up the basic principles of subtractive synthesis intuitively.

The Chain Symbols

Manipulating the Cutoff and Resonance controls in real time is one of the key ingredients in the creation of expressive synthesizer sounds. You'll be pleased to know that you can control two filter parameters at once by dragging on one of the three little chain symbols in the filter graphic.

- The chain between Cut and Res of Filter1 controls Resonance (horizontal mouse movements) and Cutoff (vertical mouse movements) of the first filter simultaneously.
- The chain between Cut and Res of Filter2 controls Resonance (horizontal mouse movements) and Cutoff (vertical mouse movements) of the second filter simultaneously.
- The chain between Filter1's Cut and Filter2's Cut controls Cutoff (vertical mouse movements) of the first filter, and Cutoff (horizontal mouse movements) of the second filter simultaneously.

Flt Reset

If you increase the filter's Resonance parameter to higher values, it will begin to internally feed back and will start to self-resonate. This resonance results in a sine oscillation (a sine wave), which you may be familiar with, if you've used subtractive synthesizers before.



In order to start this type of oscillation, the filter requires a trigger. In an analog synthesizer, this trigger may be the noise floor or the oscillator output. In the digital domain of the ES2, noise is all but eliminated. As such, when the oscillators are muted, there is no input signal routed to the filter.

When the Filter Reset button is engaged, however, each note starts with a trigger that makes the filter resonate immediately. The Filter Reset button is located in the upper right corner of the ES2 interface.

Filter Slope

A filter can not completely suppress the signal portion outside the frequency range defined by the Cutoff Frequency parameter. The slope of the filter curve expresses the amount of rejection applied by the filter (beneath the cutoff frequency) in dB per octave.



Filter 2 offers three different slopes: 12 dB, 18 dB and 24 dB per octave. Put another way, the steeper the curve, the more severely the level of signals below the *cutoff* frequency are affected in each octave.

Fat

Increasing the Resonance value results in a rejection of bass (low frequency energy) when using lowpass filters. The Fat button (located below the filter slope buttons) compensates for this side-effect, delivering a richer sound.

Filter Mode (Lo, Hi, Peak, BR, BP)

Filter 1 can operate in several modes, allowing the filtering (cutting away) and/or emphasis of specific frequency bands.



- A *lowpass filter* allows frequencies that fall below the cutoff frequency to pass. When set to *Lo*, the filter operates as a lowpass filter. The slope of Filter 1 is 12 dB/octave in *Lo* mode.
- A *highpass filter* allows frequencies above the cutoff frequency to pass. When set to *Hi*, the filter operates as a highpass filter. The slope of Filter 1 is 12 dB/octave in *Hi* mode.
- In *Peak* mode, Filter 1 works as a *Peak Type Filter*. This allows for a level increase in a frequency band, the width of which is controlled by the *Resonance* parameter.
- The abbreviation *BR* stands for *band rejection*. In this mode, the frequency band (a range of adjacent frequencies) directly surrounding the cutoff frequency is rejected, whilst the frequencies outside this band can pass. The *Resonance* parameter controls the width of the rejected frequency band.

• The abbreviation *BP* stands for *bandpass*. In this mode, only the frequency band directly surrounding the cutoff frequency can pass. All other frequencies are cut. The *Resonance* parameter controls the width of the frequency band that can pass. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the band.

Filter 2 FM

The cutoff frequency of Filter 2 can be modulated by the sine wave of Oscillator 1, which means that it can be modulated in the audio frequency range.



The effect of such filter modulations in the audio spectrum is unpredictable, but the results tend to remain harmonic as long as the modulation intensity doesn't get too high. FM defines the intensity of frequency modulation. This parameter can be modulated in real-time: In the Router, this modulation is abbreviated as LPF FM.

A clean sine wave, at the frequency of Oscillator 1, is always used as the modulation source.

Note: Don't confuse this type of filter frequency modulation with the FM feature of Oscillator 1, which can be modulated by Oscillator 2, as described in "Linear Frequency Modulation" on page 257. If a frequency modulation of Oscillator 1 by Oscillator 2 is used, it does not influence the (sine wave) signal used to modulate the cutoff frequencies.

Filter 2 can be driven to self-oscillation. If you set a very high value for Resonance, it will produce a sine wave. This self-oscillating sine wave will distort at the maximum Resonance value. If you mute all oscillators, you'll only hear this sine oscillation. By modulating the Cutoff Frequency, you can produce effects similar to those produced by modulating the frequency of Oscillator 1 with Oscillator 2.

Handling Processing Power Economically

The ES2 is designed to make the most efficient use of computer processing power. Modules and functions that are not in use don't use processing power. This principle is maintained by all elements of the ES2.

As examples: If only one of the three oscillators is in use, and the others are muted, less processing power is required. If you do not modulate Digiwaves, or if you disengage the filters, processing power is saved. When it comes to filtering, here are some hints that will help you to use processing power most efficiently:

- If you can achieve the same lowpass-filtered sound with Filter 1 as with Filter 2, use Filter 1. Filter 1 uses less processing power, although it differs a little, sonically.
- Filter FM uses additional processing power. If you don't need it, don't use it.
- Modulation of the Filter Blend parameter requires quite a bit of additional processing power, as soon as it is engaged in the Router.
- Drive requires additional processing power. This is especially the case when it comes to filters wired in series and Filter Blend settings with two distortion circuits. See "Filter Blend and Signal Flow" on page 263 for details.

Dynamic Stage (Amplifier)

The dynamic stage defines the level—which means the perceived volume—of the played note. The change in level over time is controlled by an *envelope generator*.



ENV3 and the Dynamic Stage

ENV3 is hard-wired to the dynamic stage—envelope generator 3 is always used for control over the level of the sound. For detailed explanations of the envelope parameters, see "The Envelopes (ENV 1 to ENV 3)" on page 287.

Router Modulation Target: Amp

The dynamic stage can be modulated by any modulation source in the Router. The modulation target is called AMP in the Router.

Note: If you select AMP as the target, LFO1 as the source, and leave via set to Off in the Router, the level will change periodically, based on the current Rate of the LFO—and you'll hear a tremolo.

Sine Level



The Sine Level knob (located next to the Filter 2 section) allows the mixing of a sine wave (at the frequency of Oscillator 1) directly into the dynamic stage, independent of the filters. Even if you have filtered away the basic partial tone of Oscillator 1 with a highpass filter, you can reconstitute it through use of this parameter. Please note:

- In cases where Oscillator 1 is frequency modulated by Oscillator 2 (if you have turned up FM with the waveform selector), *only* the pure sine wave is mixed into the dynamic section, not the distorted FM waveform.
- Low frequency modulations of Oscillator 1's pitch, set in the Router, affect the sine wave frequency mixed in here.

Note: Sine Level is well suited for adding warmth and a fat bass quality to the sound. Thin sounds can be fattened up with this function, given that Oscillator 1 actually plays the basic pitch.

The Router

The ES2 features a modulation matrix, called the *Router*. If the vector envelope is displayed, click the Router button to view the Router.



Any modulation *source* can be connected to any modulation *target*—much like an old-fashioned telephone exchange or a studio patchbay. The modulation intensity—how strongly the target is influenced by the source—is set with the associated vertical slider.

Note: To set the modulation intensity to zero, just click on the little zero symbol (the small circle) right beside via.

The intensity of the modulation can also be modulated: The *via* parameter determines yet another modulation source, which defines the amount or intensity of the modulation.

Ten such modulations of source, via, and target can take place simultaneously, in addition to those which are hard-wired outside the Router. The bypass (b/p) parameter allows the disabling/enabling of individual modulation routings without losing settings.

Note: Some modulations aren't possible, due to technical reasons. As an example, the envelope times can be modulated by parameters that are only available during a note-on message. Therefore, there are situations where the envelopes are not available as *targets*. Furthermore, the LFO 1 can't modulate its own frequency. Values that are not available are grayed out.

The Range of Via

The modulation intensity is set with the vertical slider. This is self-explanatory, as long as the "via" parameter is set to off. This ensures that the modulation intensity is constant, if not affected by any other controller (such as the modulation wheel or aftertouch).

As soon as you select a value other than off for "via," the slider is divided into two halves. The lower half defines the minimum intensity of the modulation, when the via controller is set to its minimum value. The upper half defines the maximum modulation intensity when the via modulator (the modulation wheel, in this case) is set to its maximum value. The area between the two slider halves defines the range that is controlled by the via controller.

You can grab the area between the two slider halves with the mouse and drag both halves simultaneously. If this area is to small to be grabbed with the mouse, just click on a free part of the slider track and move the mouse up or down to move the area.

In the example below, the lower half of the slider knob defines the vibrato intensity when the modulation wheel is turned down. The upper half defines the vibrato intensity that takes place when the modulation wheel is set to its maximum value.



Note: To invert the effect of the via modulation source, simply activate the "via invert" (inv) parameter in the Router.

A Modulation Example

Say you've chosen these settings:

Target: Pitch 123 via: Wheel Source: LFO1

• Modulation intensity: Slider position, set as desired

In this configuration, the modulation source—LFO1—is used to modulate the frequency (pitch) of all three Oscillators (Pitch 123). (Pitch 123) is the modulation target in this example. You'll hear a vibrato (a modulation of the pitch) at the speed of LFO 1's Rate. The modulation intensity is controlled by the (modulation) wheel, which is determined by the via parameter. This provides you with control over the depth of vibrato (pitch modulation) via the modulation wheel of your keyboard. This type of configuration is well-known in countless sound settings (patches).

It does not matter which of the ten Router Channels you use.

You can select the same target in several Router Channels, in parallel. You can freely use the same sources as often as you like, and the same via controllers can be set in one or multiple Router channels.

Modulation Targets

The following targets are available for real-time modulation.

Note: These modulation targets are also available for the X and Y axes of the X/Y modulator (the Square). See "The Square" on page 291.

Pitch 123

This target allows the parallel modulation of the frequencies (pitch) of all three oscillators. If you select an LFO as the source, this target leads to siren or vibrato sounds. Select one of the envelope generators with zero attack, short decay, zero sustain, and short release as the source for tom and kick drum sounds.

Pitch 1

This target allows modulation of the frequency (pitch) of Oscillator 1. Slight envelope modulations can make the amount of detuning change over time, when Oscillator 1 is sounding in unison with another (unmodulated) Oscillator. This is useful for synthesizer brass sounds

Pitch 2

This target allows modulation of the frequency (pitch) of Oscillator 2.

Pitch 3

This target allows modulation of the frequency (pitch) of Oscillator 3.

Detune

This target controls the amount of detuning between all three oscillators.

Note: The sensitivity of all the pitch modulation targets described above depends on the modulation intensity. This sensitivity scaling allows for very delicate vibrati in the cent range (one cent equals 1/100 semitone), as well as for huge pitch jumps in octave ranges.

- Modulation intensity from 0 to 8: steps are 1.25 cents.
- Modulation intensity from 8 to 20: steps are 3.33 cents.
- Modulation intensity from 20 to 28: steps are 6.25 cents.
- Modulation intensity from 28 to 36: steps are 12.5 cents.
- Modulation intensity from 36 to 76: steps are 25 cents.
- Modulation intensity from 76 to 100: steps are 100 cents.

This leads to the following rules of thumb for modulation intensity settings.

- Modulation intensity of 8 equals a pitch shift of 10 cents.
- Modulation intensity of 20 equals a pitch shift of 50 cents, or one quarter tone.
- Modulation intensity of 28 equals a pitch shift of 100 cents, or one semitone.
- Modulation intensity of 36 equals a pitch shift of 200 cents, or two semitones.
- Modulation intensity of 76 equals a pitch shift of 1,200 cents, or one octave.
- Modulation intensity of 100 equals a pitch shift of 3,600 cents, or three octaves.

OscWaves

Dependent on the Waveforms set in the three oscillators, this target can be used to modulate:

- The pulse width of rectangular and pulse waves
- The amount of frequency modulation (Oscillator 1 only)
- Noise color (Oscillator 3 only)
- The position of the Digiwaves

OscWaves affects all oscillators simultaneously. The Osc1Wave, Osc2Wave, and Osc3Wave targets only affect the specified Oscillator. Check out the ensuing paragraphs to see what wave modulation does in the three oscillators.

For further information on the effects of these modulations, please read "Pulse Width Modulation" on page 258. Also take a look at "Linear Frequency Modulation" on page 257, "White and Colored Noise (Oscillator 3 Only)" on page 260, and "Digiwaves" on page 256.

Osc1Wave

Dependent on the *waveform* selected, you can control the pulse width of rectangular and pulse waves of Oscillator 1, the amount of frequency modulation (with Oscillator 1 being the carrier and Oscillator 2 being the modulator), or the position of the Digiwave. The pulse width of the rectangular and pulse waves is not restricted to two fixed values in Oscillator 1.

Note: In classic FM synthesizers, the amount of FM is controlled in real time by velocity sensitive envelope generators. Select one of the ENVs as the source for such sounds.

Osc2Wave

As per Osc1Wave, except that Oscillator 2 does not feature FM. Please note that pulse width modulation also works with both the synchronized rectangular and ring modulated rectangular waves.

Osc3Wave

Oscillator 3 is as per Osc1Wave and Osc2Wave, but it does not feature FM or ring modulation. Oscillator 3 features Noise, the color of which can be modulated with this parameter.

OscWaveB

The transitions between Digiwaves during a wavetable modulation are always smooth. Depending on the modulation intensity, an additional *OscWaveB* target can be used to continuously modulate the shape of the transitions from smooth to hard. It applies to all *Oscillators*.

Osc1WaveB

If wavetable modulation is active for a Digiwave using Osc1Wav, you can use this target to modulate the shape of the transition.

Osc1 FM mode: When compared with the hardwired Osc1-FM and the Osc1Wave modulation target, the Osc1WaveB modulation target offers much higher FM intensities.

Osc2WaveB

If wavetable modulation is active for a Digiwave using *Osc2Wav*, you can use this target to modulate the shape of the transition.

Osc3WaveB

If wavetable modulation is active for a Digiwave using *Osc3Wav*, you can use this target to modulate the shape of the transition.

SineLevl

SineLevl (Sine Level) allows modulation of the *sine wave* level of Oscillator 1, which can be mixed directly into the input of the dynamic stage—without being affected by the filters. The parameter defines the level of the first partial tone of Oscillator 1. See "Sine Level" on page 271.

OscLScle

OscLScle (Osc Level Scale) allows modulation of the levels of *all three* oscillators simultaneously. A modulation value of *0* mutes all oscillators, while a value of *1* raises the gain of the entire mix by 12 dB. The modulation is applied *before* the overdrive stage, allowing for dynamic distortions.

Osc1LevI

Osc1LevI (Osc 1 Level) allows modulation of Oscillator 1's level.

Osc2LevI

Osc2Levl (Osc 2Level) allows modulation of Oscillator 2's level.

Osc3LevI

Osc3LevI (Osc 3Level) allows modulation of Oscillator 3's level.

Cutoff 1

This target allows modulation of the Cutoff Frequency of Filter 1. See "Cutoff and Resonance" on page 266.

Resonance 1 (Reso 1)

This target allows modulation of the Resonance of Filter 1. See "Cutoff and Resonance" on page 266.

Cutoff 2

This target allows modulation of the Cutoff Frequency of Filter 2.

Resonance 2 (Reso2)

This target allows modulation of the Resonance of Filter 2.

LPF FM

A sine signal, at the same frequency as Oscillator 1, can modulate the *Cutoff frequency* of Filter 2 (which always works as a lowpass filter). LPF FM (Lowpass Filter Frequency Modulation) allows the modulation of Filter 2's FM modulation intensity. This filter FM parameter is described in "Filter 2 FM" on page 269.

Cut 1+2

Cut 1+2 (Cutoff 1 and 2) modulates *both* filter's *Cutoff* frequencies in parallel, much like applying the same modulation to Cutoff 1 and Cutoff 2 in two *Router* channels.

Cut1inv2

Cut1inv2 (Cutoff 1 normal and Cutoff 2 inverse) simultaneously modulates the *Cutoff* frequencies of the first and second filters *inversely* (in opposite directions). Put another way, while the first filter's *Cutoff* frequency is rising, the *Cutoff* of the second filter will fall—and vice versa.

Note: In cases where you have combined Filter 1, defined as a highpass filter, and Filter 2 (which always works in lowpass mode) in *Serial* mode, both will serve as a bandpass filter. Modulating the *Cut1 inv 2* target will result in a modulation of the bandpass filter's bandwidth in this scenario.

FltBlend

FltBlend (Filter Blend) modulates the *FilterBlend* (the cross-fading of the two filters), as described in "Filter Blend and Signal Flow" on page 263.

Note: If FilterBlend is set as a target in one or several Router channels, the modulation data for both filters will be calculated—even if the FilterBlend parameter is set to 1.0 or +1.0. As such, you should be careful when choosing FilterBlend as a modulation target because it may increase the need for processing power.

Amp

This target modulates the dynamic stage, or level (the voice's volume). If you select Amp as the target and modulate it with an LFO as the source, the level will change periodically, and you will hear a tremolo.

Pan

This target modulates the panorama position of the sound in the stereo or surround spectrum. Modulating Pan with an LFO will result in a stereo or surround tremolo (auto panning).

In Unison Mode, the panorama positions of all voices are spread across the entire stereo or surround spectrum. Nevertheless, pan can still be modulated, with positions being moved in parallel.

Note: The extended Surround Range parameter defines the angle range resulting from modulation values. As an example: if pan is modulated by the maximum amount of an LFO (using a sawtooth waveform), a Surround Range value of 360 ° results in circular movements of the voice output.

Diversity

This parameter (only available in surround instances of the ES2) allows you to dynamically control how much the voice output is spread across the surround channels. Negative values reduce this effect. See the Surround chapter of the *Logic Pro 8 User Manual* for details on surround terminology and use.

Lfo1Asym

Lfo1Asym (Lfo1 Asymmetry) can modulate the selected waveform of LFO 1. In the case of a square wave, it changes its pulse width. In the case of a triangle wave, it sweeps between triangle and sawtooth. In the case of a sawtooth wave, it shifts its zero crossing.

Lfo1Curve

This target modulates the waveform smoothing of the square and random wave. In the case of a triangle or sawtooth wave, it changes their curves between convex, linear, and concave.

Scaled Modulations

All following modulation targets result in a scaled modulation, which means that the modulation value isn't simply added to the target parameter's value, but rather that the target parameter value will be multiplied. This works as follows: a modulation value of 0.0 results in no effect, while a modulation value of +1.0 equals a multiplication by 10, and a modulation value of -1.0 equals a multiplication by 0.04.

LFO1Rate

This target modulates the frequency (speed, rate) of LFO 1.

Note: Say you've created a vibrato with another Router channel by modulating the Target Pitch 123 with LFO 1. If desired, you can have LFO 1's speed (the vibrato speed) automatically accelerated or slowed down. To do so, modulate the LFO1Rate target with one of the envelope generators (ENV). Select LFO 2 as a source and reduce its Rate, in order to periodically accelerate and slow down the vibrato.

Env2Atck

Env2Atck (Envelope 2 Attack) modulates the Attack time of the second envelope generator.

Env2Dec

Env2Dec (Envelope 2 Decay) modulates the Decay time of the second envelope generator.

In cases where you've selected ENV2 Dec as the target and Velocity as the source, the duration of the decaying note is dependent on how hard you strike the key. Selecting Keyboard as the source will result in higher notes decaying more quickly (or slowly).

Env2Rel

Env2Rel (Envelope 2 Release) modulates the Release time of the second envelope generator.

Env2Time

Env2Time (Envelope 2 All Times) modulates *all* of ENV2's time parameters: Attack time, Decay time, Sustain time, and Release time.

Env3Atck

Env3Atck (Envelope 3Atck) modulates the Attack time of the third envelope generator.

Env3Dec

Env3Dec (Envelope 3 Decay) modulates the Decay time of the third envelope generator.

Env3Rel

Env3Rel (Envelope 3 Release) modulates the Release time of the third envelope generator.

Env3Time

Env3Time (Envelope 3 All Times) modulates all of ENV3's time parameters: Attack time, Decay time, Sustain time, and Release time.

Glide

This target modulates the duration of the Glide (portamento) effect.

Note: If you modulate Glide with Velocity selected as the source, the velocity used to (how hard) strike the key will define how long it takes for the played notes to "find their way" to the target pitch. See "Glide" on page 252.

Modulation Sources

Some modulation sources are *unipolar*, delivering values between 0 and 1. Others are *bipolar*, and output values between –1 and +1. The following modulation sources are available:

LFO₁

LFO 1 is described in "The LFOs" on page 285.

LFO₂

LFO 2 is described in "The LFOs" on page 285.

ENV1

Envelope Generator 1 is described in "The Envelopes (ENV 1 to ENV 3)" on page 287.

ENV₂

Envelope Generator 2 is described in "The Envelopes (ENV 1 to ENV 3)" on page 287.

ENV3

Envelope Generator 3 is described in "The Envelopes (ENV 1 to ENV 3)" on page 287.

Note: Envelope Generator 3 always controls the level of the overall sound.

Pad-X, Pad-Y

These modulation sources allow you to define the axes of the Square, for use with the selected modulation target. The cursor can be moved to any position in the Square, either manually or controlled by the vector envelope. See "The Square" on page 291 and "The Vector Envelope" on page 293.

Max

If you select Max as a source, the value of this source will permanently be set to +1. This offers interesting options in conjunction with via, as the possible values available for via control the modulation intensity.

Kybd

Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0). Five octaves below and above, an output value of -1 or +1, respectively is sent.

Note: This could be used to control the Cutoff frequencies of the filters in parallel with the keyboard position—as you played up and down the keyboard, the Cutoff frequencies would change. Modulate the Cut 1+2 target with the Keyboard source to do so. At a modulation intensity of 0.5, the Cutoff frequencies scale proportionally with the pitches played on the keyboard.

Velo

If you select Velo (Velocity), velocity sensitivity serves as modulation source.

Bender

The pitch bender serves as a bipolar modulation source, if Bender is selected. This is also true when the Oscillators' Bend Range parameter is set to 0.

ModWhl

The modulation wheel serves as an unipolar modulation source, if ModWhl is selected.

Note: For most standard applications, you'll probably use the wheel as the *via* controller. Traditionally, it can be (and is) used for control over the intensity of periodic LFO modulations. Used here, it can be employed for direct, static modulations, such as controlling the Cutoff frequencies of the filters (Target = Cut 1+2).

Note: The Least Significant Byte (LSB) controller for the modulation wheel is recognized correctly, as well.

Touch

Aftertouch serves as modulation source. The ES2 reacts to poly pressure (polyphonic aftertouch). It uses the sum of channel pressure and the note-specific poly pressure value.

Note: If you set the *target* to Cut 1+2, the cutoff frequencies will rise and fall, dependent on how firmly you press a key on your touch-sensitive MIDI keyboard after the initial keystrike.

WhI+To

The modulation wheel and aftertouch serve as modulation sources.

MIDI Controllers A-F

MIDI controllers available in the mod matrix are named Ctrl A–F and can be assigned to arbitrary controller numbers (via the MIDI Controllers Assignment menus at the bottom of the ES2 interface).

Note: Earlier ES2 versions offered the Expression, Breath, and MIDI Control Change Messages 16 to 19 as modulation sources. These MIDI controllers are the default values for the assignment and guarantee backwards compatibility.

The values of the MIDI Controllers Assignment menus are only updated if the default setting is loaded, or if a setting that was saved with a project is loaded. If you simply step through settings, the assignment will remain unchanged.

The MIDI Controllers Assignment menus allow you to assign your favorite MIDI controllers as Ctrl A, Ctrl B, and so on.

All MIDI Controller Assignment menus feature a *Learn* option. If this is selected, the parameter will automatically be assigned by the first appropriate incoming MIDI data message. The Learn mode features a 20 second time-out facility: If the ES2 does not receive a MIDI message within 20 seconds, the parameter will revert to its original MIDI controller assignment.

Note: If none of the controller assignments (Ctrl A–F) is assigned to Expression, the Expression CC message (Ctrl #11) controls the output volume.

Note: The Vector Stick (Joystick) of the Korg Wavestation synthesizer generates Controllers 16 and 17, for example. If you use this instrument as your master keyboard, you can control any two ES2 parameters directly with its Joystick.

Note: In the MIDI specification, for all controllers from 0 to 31, there also is a LSB-Controller defined (32 to 63). This Least Significant Byte-controller allows for a resolution of 14 bit instead of 7 bit. The ES2 recognizes these control change messages correctly; the breath or expression controllers, for example.

RndN01

RndNO1 (Note On Random1) outputs a random modulation value between –1.0 and 1.0 (same range as an LFO), that changes when a note is triggered or re-triggered. The (random) note-on modulation remains constant throughout the duration of the note until the next note-on trigger.

Note: There is no value change when playing legato, while in legato mode.

RndN02

RndNO2 (Note On Random2) behaves like Note On Random1 with the exception that it glides to the new random value using the Glide time (inclusive of modulation). It also differs from NoteOnRandom1 in that the (random modulation) value changes when playing legato, while in legato mode.

SideCh

SideCh (Side Chain modulation) uses a Side Chain (tracks, inputs, busses) to create a modulation signal. The Side Chain source can be selected in the upper gray area of the window. It is fed to the internal envelope follower, which creates a modulation value based on the current Side Chain input signal level.

Via—Controlling the Modulation Intensity

Some modulation sources are *unipolar*, delivering values between 0 and 1. Others are *bipolar*, and output values between -1 and +1. The following *sources* may be used to modulate the modulation intensity.

LFO1

The modulation undulates at the speed and waveform of LFO1, which controls the modulation intensity.

LFO2

The modulation undulates at the speed and waveform of LFO2, which controls the modulation intensity.

ENV1

ENV1 controls the modulation intensity.

ENV₂

ENV2 controls the modulation intensity.

ENV3

ENV3 (the level envelope) controls the modulation intensity.

Pad-X, Pad-Y

Both axes of the Square (the vector envelope) are available as *via sources* as well, enabling you to control modulation intensities with them.

Kybd

Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0). Five octaves below and above, an output value of -1 or +1, respectively is sent.

If you select Pitch 123 as the target, modulate it with the LFO1 source, and select Keyboard as the via value, the vibrato depth will change, dependent on key position. Put another way, the vibrato depth will be different for notes higher or lower than the defined Keyboard position.

Velo

If you select *Velo (Velocity)* as the *via* value, the modulation intensity will be velocity sensitive—modulation will be more or less intense dependent on how quickly (hard) you strike the key.

Bender

The pitch bender controls the modulation intensity.

ModWhl

If you select ModWhl (Modulation Wheel) as the via value, the modulation intensity will be controlled by your MIDI keyboard's modulation wheel.

The Least Significant Byte (LSB) Controller for the modulation wheel is recognized correctly, as well.

Touch

If you select Touch (Aftertouch) as the via value, the modulation intensity will be touch sensitive—modulation will be more or less intense dependent on how firmly you press the key of your touch-sensitive MIDI keyboard after the initial keystrike (aftertouch is also known as pressure sensitivity).

WhI+To

Both modulation wheel and aftertouch control the modulation.

MIDI Controllers A-F

MIDI controllers available in the mod matrix are named Ctrl A–F, rather than Expression, Breath, and General Purpose 1–4 (MIDI Control Change Messages 16 to 19 are also known as General Purpose Slider 1/2/3/4). These can be assigned to arbitrary controller numbers via menus in the Controller Assignments section at the bottom of the interface (press the MIDI button to view the A to F menus).

The default values for the assignment guarantee backwards compatibility. The values of these assignment menus are only updated if the default setting is loaded, or if a setting that was saved with a project is loaded. If you simply step through settings, the assignment will remain unchanged.

This facility is especially helpful if you've always wanted to use Controller #4 (foot pedal), for example, as a modulation source. This feature allows you to assign your favorite MIDI real-time controllers as Ctrl A, Ctrl B, and so on.

All parameters that allow you to select a MIDI controller feature a *Learn* option. If this is selected, the parameter will automatically be assigned by the first appropriate incoming MIDI data message. The Learn mode features a 20 second time-out facility: If the ES2 does not receive a MIDI message within 20 seconds, the parameter will revert to its original MIDI controller assignment.

Note: As the new entry is added to the top of the list, existing automation data is increased by one. Should further assignments be made, this will increment by one for each entry.

Note: If none of the controller assignments (Ctrl A–F) is assigned to Expression, the Expression CC message (Ctrl #11) controls the output volume.

Note: The Vector Stick (Joystick) of the Korg Wavestation synthesizer generates Controllers 16 and 17, for example. If you use this instrument as your master keyboard, you can control modulation intensities directly with its Joystick.

Note: In the MIDI specification, for all controllers from 0 to 31, there also is a LSB-Controller defined (32 to 63). This Least Significant Byte-controller allows for a resolution of 14 bit instead of 7 bit. The ES2 recognizes these control change messages correctly, for instance the controllers for breath or expression.

RndN01

RndN01 (NoteOnRandom1) controls the modulation intensity (See RndNO1 on page 281).

RndN02

RndNO2 (NoteOnRandom2) controls the modulation intensity (See RndNO2 on page 282).

SideCh

A Side Chain source (tracks, inputs, busses) is used to create the modulation signal.

The LFOs

LFO is the abbreviated form of Low Frequency Oscillator. In an analog synthesizer, LFOs deliver modulation signals below the audio frequency range—in the bandwidth that falls between 0.1 and 20 Hz, and sometimes as high as 50 Hz. LFOs serve as modulation sources for periodic, cyclic modulation effects. If you slightly modulate the pitch of an audio oscillator at a rate (speed, LFO frequency) of, say, 3–8 Hz, you'll hear a vibrato. If you modulate the cutoff frequency of a lowpass filter, you'll hear a wah wah effect, and modulating the dynamic stage results in a tremolo.

The ES2 features two LFOs, the outputs of which are available as sources in the Router.



- LFO 1 is polyphonic, which means that if used for any modulation of multiple voices, they will not be phase-locked. Furthermore, LFO 1 is key-synced: Each time you hit a key, the LFO 1 of this voice is started from zero. To explain, when used on polyphonic input (a chord played on the keyboard) the modulation is independent for each voice (note). Where the pitch of one voice may rise, the pitch of another voice might fall and the pitch of a third voice may reach its minimum value.
- LFO 2 is monophonic, which means, that the pitch of all voices will rise and fall synchronously, if you modulated the Pitch123 target with the LFO2 source, for example.

Both LFOs feature a number of waveforms. LFO 1 can fade in or out automatically, without the need to engage a separate envelope generator. The LFO parameters are detailed below:

EG (LFO1)

At its center position—which can be accessed by clicking the middle mark—the modulation intensity is static: it won't be faded in or out at all. At positive values, it is faded *in*. The higher the value, the longer the delay time is. At negative values, it is faded *out*. The lower (onscreen) the slider is positioned, the shorter the fade out time is.

The function is abbreviated as EG because the fading in or out is internally performed by an ultra-simple envelope generator.

Most commonly, this is used for delayed vibrato—many instrumentalists and singers intonate longer notes this way. To set up a delayed vibrato: Place the slider at a position in the upper half (Delay) and modulate the Pitch123 target with the LFO1 source. Set a slight modulation intensity. Select a Rate of about 5 Hz and the triangular wave as the LFO waveform.

Tip: Chaotic and fast modulations of the oscillators frequencies (Pitch123) by LFO 1 with a delayed Sample&Hold selected as the waveform, a high Rate, and short fade out, make the attack phase of the note sound Moog Rogue-ish—and quite similar to the attack phase of brass instruments.

Rate

This parameter defines the frequency or speed of the modulation. The value is displayed in Hertz (Hz) beneath the slider.

Wave

This is where you select the desired LFO waveform. Check out the waveforms while a modulation of Pitch123 is engaged and running. You should find the symbols quite self-evident.

Triangular Wave

The triangular wave is well suited for vibrato effects.

Sawtooth Wave and Inverted Sawtooth

The sawtooth is well suited for helicopter and space gun sounds. Intense modulations of the oscillator frequencies with a negative (inverse) sawtooth wave leads to bubbling and boiling, underwater sounds. Intense sawtooth modulations of lowpass filters (such as Filter 2) create rhythmic effects.

Rectangular Waves

The rectangular waves make the LFO periodically switch between two values. The upper rectangular wave switches between a positive value and zero. The lower wave switches between a positive and a negative value set to the same amount above/below zero.

Note: An interesting effect you may wish to try out is achieved by modulating Pitch123 with a suitable modulation intensity that leads to an interval of a fifth. Choose the upper rectangular wave to do so.

Sample & Hold

The two lower waveform settings of the LFOs output *random* values. A random value is selected at regular intervals, defined by the LFO rate. The upper waveform delivers exact steps of randomization. At its lower setting, the random wave is smoothed out, resulting in fluid changes to values.

The term Sample & Hold (abbreviation—S & H) refers to the procedure of taking samples from a noise signal at regular intervals. The voltage values of these samples are then *held* until the next *sample* is taken. When converting analog audio signals into digital signals, a similar procedure takes place: Samples of the voltage of the analog audio signal are taken at the rate of the sampling frequency.

Tip: A random modulation of Pitch 123 leads to the effect commonly referred to as a random pitch pattern generator or sample and hold. Check out very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies!

Rate (LFO 2)

The LFO2 Rate (frequency) control allows for the free (in the upper half of the slider range) or project-tempo synchronized (in the lower half of the slider range) running of LFO 2. The rate is displayed in Hertz or rhythmic values, dependent on whether the project tempo synchronization is engaged or not. Rates range from speeds of 1/64-notes through to a periodic duration of 32 bars. Triolic and punctuated values are also possible. LFO 2 is ideally suited for rhythmic effects which retain perfect synchronicity, even during tempo changes to the project.

The Envelopes (ENV 1 to ENV 3)

In addition to the complex vector envelope, described in "The Vector Envelope" on page 293, the ES2 also features three envelope generators per voice. On both the front panel and as a *source* in the *Router*, they are abbreviated as ENV 1, ENV 2, and ENV 3, respectively.



Note: The roots of the term envelope generator and its basic functionality are described in "Envelopes" on page 626.

The feature sets of ENV 2 and ENV3 are identical. ENV 3 defines the changes in level over time for each note played. You can regard ENV 3 as being hard-wired to the Router's AMP modulation target.

The parameters of ENV 2 and ENV 3 are identical—but ENV 3 is always used for control over level.

Unlike many other synthesizers, there is no hard-wired connection between any of the envelope generators and the cutoff frequencies of the ES2 filters. Modulation of the cutoff frequencies must be set separately in the Router. In the default setting, this is already the case—in the Router channel just below the Filter (see graphic).

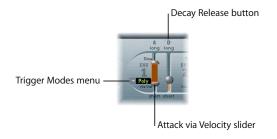


Set up a Router channel as follows, in order to establish this type of modulation: Set target to Cutoff 1, Cutoff 2, or Cut 1+2, set source to, say, ENV 2. Once set as described, the slider of the Router channel will function as the filter's EG Depth parameter.

Note: Both ENV 2 and ENV 3 are velocity sensitive, making it unnecessary to set via to Velo in the Router channel: You can leave via switched off.

The Parameters of ENV 1

At first glance, ENV 1 appears to be rather poorly equipped. Its few parameters, however, are useful for a vast range of synthesizer functions.



Trigger Modes: Poly, Mono, Retrig

In Poly mode, the envelope generator behaves as you would expect on any polyphonic synthesizer: Every voice has its own envelope.

In Mono and Retrig modes, a single envelope generator modulates all voices in parallel—identically, in other words.

• If ENV 1 is set to Mono, all notes must be released before the envelope can be triggered again. If you play legato, or any key remains depressed, the envelope won't start its attack phase again.

• In Retrig mode, the envelope will be triggered by any key you strike, no matter whether other notes are sustained or not. Every sustained note is affected by the retriggered envelope.

The design of early analog polysynths led to polyphonic instruments where all voices passed through a single lowpass filter. This design was primarily due to cost factors. The best known example of these polyphonic instruments were the Moog Polymoog, the Yamaha SK20, and Korg Poly 800. The sole lowpass filter of such instruments is controlled by a single envelope generator. To simulate this behavior, use the Mono or Retrigger modes.

Say you've modulated the Cutoff 2 target with a percussive source, such as ENV1, which is set to Retrig. If you play and sustain a bass note, this note will receive a percussive filter effect every time you hit another key. The newly struck key is also shaped by the same filter. Playing a sound set up in this way feels like you're playing a polyphonic synthesizer with *one* filter. This is despite the fact that the ES2's filters remain polyphonic and can be simultaneously modulated by different polyphonic sources.

Tip: If you want to simulate the percussion of a Hammond Organ, you will also need to use the Mono or Retrigger modes.

Decay/Release

ENV 1 can be set to act as an envelope generator with an Attack time and Decay time parameter *or* with an Attack time and Release time parameter.

Switching between both modes is achieved by clicking on the D or the R above the right ENV 1 slider.

- In its Attack/Decay mode, the level will fall to zero after the attack phase has
 completed, no matter whether you sustain the note or not. It will decay at the same
 speed even if you release the key. The decay time is set with the Decay time slider,
 abbreviated as D.
- In its Attack/Release mode, the envelope level remains at its maximum after the attack phase is over, as long as the key remains depressed. Following the release of the key, its level decreases over the time set with the R slider—the abbreviation for Release time.

Attack Time and Attack Via Vel

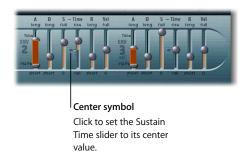
The Attack time slider is divided into two halves. The lower half defines the attack time when the keys are struck hard (at maximum velocity). The upper half defines the attack time at minimum velocity.

You can grab the area between the two slider halves with the mouse and drag both halves simultaneously. If this area is to small to be grabbed with the mouse, just click in a free part of the slider track, and move the mouse up or down to move the area.

The Parameters of ENV 2 and ENV 3

The feature sets of ENV 2 and ENV 3 are identical, but it is always the task of ENV 3 to define the level of each note—to modulate the dynamic stage. ENV 3 is available for simultaneous use as a source in the Router as well. The envelope's time parameters can also be used as modulation targets in the Router.

Note: See "Envelopes" on page 626 for information on the basic functionality and meaning of envelope generators.



Attack Time

As per the Attack slider of ENV 1, the Attack time sliders of ENV 2 and ENV 3 are divided into two halves. The lower half defines the attack time when the keys are struck at maximum velocity. The upper half defines the attack time at minimum velocity.

You can grab the area between the two slider halves with the mouse and drag both halves simultaneously. If this area is to small to be grabbed with the mouse, just click in a free part of the slider track, and move the mouse up or down to move the area.

Decay Time

The Decay time parameter defines the length of time it takes for the level of a sustained note to fall to the Sustain level, after the attack phase is over. If Sustain level is set to its maximum, the Decay parameter has no effect. When the Sustain level is set to its minimum value, Decay defines the duration or fade-out time of the note.

The Decay parameter appears as a modulation target in the Router for ENV 2 and ENV 3 individually (ENV2Dec, ENV3Dec).

Tip: On pianos and plucked string instruments, high notes decay faster than low notes. In order to simulate this effect, modulate the Decay Time target with the Kybd source in the Router. The Router channel slider should be set to a negative value.

Sustain and Sustain Time

When the Sustain Time (rise) parameter is set to its center value—which can be achieved by clicking the center symbol shown above—the Sustain level behaves like the Sustain parameter of any synthesizer ADSR envelope.

In this position, the Sustain level (abbreviated as S) defines the level that is sustained for as long as the key remains depressed, following the completion of the Attack time and Decay time phases.

The Sustain Time slider defines the time it takes for the level to rise to its maximum—or to fall to zero—after the decay phase is over. Settings in the lower half of its range (Fall) determine the speed at which the level decays from the Sustain level to zero. The lower the slider position, the faster the decay speed. Settings in the upper half of its range (Rise) determine the speed at which the level rises from the Sustain level to its maximum value. The higher the slider position, the faster the speed of the rise.

Release Time

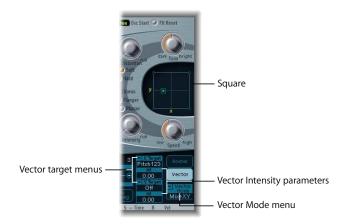
As with any synthesizer ADSR envelope, the Release time parameter (R) defines the time the level takes to decay to zero after the key is released.

Vel

The Vel (Velocity Sensitivity) parameter defines the velocity sensitivity of the entire envelope. If it is set to maximum, the envelope will only output its maximum level when the keys are struck at maximum velocity.

The Square

The Square has two axes: The X and Y axes have positive and negative value ranges. They are bipolar, in other words. By touching and moving the cursor with the mouse, the values of both axes are continuously transmitted. As you can modulate one freely selectable parameter with the X value, and another freely selectable parameter with the Y value, you can use the mouse like a joystick.



As an alternative to this real-time control, the position of the cursor can be modulated by the vector envelope—just like the mix between the three oscillators in the Triangle. The Loop function of the vector envelope generator allows for cyclic movements. This opens a number of doors, allowing it to operate as a two-dimensional, luxurious pseudo-LFO with a programmable waveform. More on this is found in "The Vector Envelope" on page 293.

Vector Mode Menu

The Vector Mode menu below the vector envelope button allows you to disable the control of the Square cursor by the vector envelope. The menu also defines whether or not the Triangle (the oscillator mixer) shall be controlled by the vector envelope.

- Off: The vector envelope does not influence the Triangle or the Square. It's simply switched off. This allows you to set and control the cursors of the Triangle and the Square in real-time.
- *Mix:* The vector envelope controls the Triangle (the oscillator mix), but not the Square.
- XY: The vector envelope controls the Square, but not the Triangle.
- *Mix+XY*: The vector envelope controls both the Square and Triangle.

Note: Like all of the ES2's parameters, the movements of the cursors in the Triangle and Square can be recorded and automated by Logic Pro. This automation data can be edited and looped in Logic Pro. This is completely independent of the cyclic modulations of the vector envelope. Vector modulation of the Square and Triangle should be disabled for this type of use (Vector mode = off).

Vector Target—Modulation Destinations

The Vector X and Vector Y target menus determine the effect of cursor movements in the Square. The modulation targets are identical to those available in the Router. Please see "Modulation Targets" on page 273 for descriptions. The position of the cursor in the Square is also available in the Router, as the Pad-X and Pad-Y Source and via options.

Vector Int—The Modulation Intensity

The maximum intensity, sensitivity, and polarity of the modulation is set with the Vector X Int and Vector Y Int parameters.

The Vector Envelope

The Triangle and Square are the most special and unusual elements of the ES2's graphical user interface. Whilst the Triangle controls the mix of the three oscillators, the X and Y axes of the Square can modulate any (modulation) target.



The vector envelope can control the movement of the cursors in the Triangle and the Square in realtime. Each voice is equipped with its own vector envelope, which is triggered from its startpoint with every new keystrike (or with every MIDI note-on message, to be more precise).

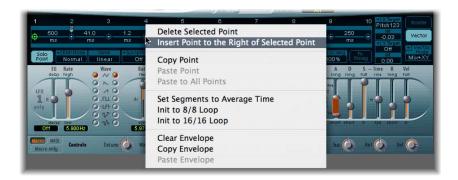
The concept of the vector envelope, the Square, and the Triangle may seem strange at first glance; but when combined with the other ES2 synthesis options, you will end up with sounds that are truly unique—and quite literally—moving.

Envelope Points, Times, and Loops

The vector envelope consists of up to 16 points on the time axis. Each point can control the position of both the Triangle and Square's cursors.

The points are numbered sequentially. Point 1 is the starting point. In order to edit a point, simply select it—by clicking on it.

Note: A number of vector envelope editing commands can be quickly accessed in a shortcut menu. Control-click anywhere on the vector envelope to open it.



Sustain Point

Any point can be declared the *sustain point*. Given that the note played is sustained long enough and there's no loop engaged, any envelope movement will stop when this sustain point is reached. It will be sustained until the key is released (until the MIDI note-off command).

To define a point as the *sustain point*, click on the turquoise strip above the desired point. The selected point will be indicated by an S between the point and its number, on the turquoise strip.



Loop Point

Any point can be declared loop point. Given that the note is sustained long enough, the envelope can be repeated in a loop.

The looped area is the time span between sustain point and loop point. In between, you can define several points which describe the movements of the Square and Triangle cursors.

In order to define a point as the loop point, click on the turquoise strip below the desired point. A loop point is indicated by an *L* in the strip below.



In order to see and define the loop point, the loop must be activated. See "Loop Mode" on page 298.

Tip: With loop activated, the vector envelope works like a multi-dimensional, polyphonic LFO with a programmable waveform.

Vector Envelope Times

With the exception of the first point, which is tied to the beginning of each played note, every point has a Time parameter. This parameter defines the period of time required for the cursor to travel from the point which preceded it. The times are normally displayed in milliseconds (ms).

To adjust a time value, you can click directly on the numerical value and use your mouse as a slider.



Default Setting of the Vector Envelope

The default setting of the vector envelope consists of three points. Point 1 is the startpoint, point 2 is defined as the sustain point, and point three is the end point, by default.

The impact of the vector envelope on the Oscillator Mix Triangle or on the Square is switched off by default. This allows the ES2 to behave as a synthesizer without a vector envelope generator. This traditional starting point is more convenient when creating patches from scratch.

There are two ways to switch off the vector envelope:

• You can switch on the Solo Point parameter (described on page 296). If it is on, only the Triangle and Square cursor positions of the currently selected point are active.

• You can disable the vector envelope altogether (or only the Triangle or Square) as described in the "Vector Mode Menu" on page 292.

Setting and Deleting Points

The more points you set, the more complex the vector envelope movements that can be designed. You can:

- Create a new point by Shift-clicking between two existing points. The segment that
 previously existed between the two old points is divided at the mouse position. The
 sum of the two new segment times is equal to the time used by the old undivided
 segment. As such, the ensuing points retain their absolute time positions. In addition,
 the existing cursor positions in the Triangle and Square are fixed, ensuring that the
 creation of new points does not affect any previously defined movements.
- Delete points by clicking on them while holding Control.

Setting Vector Envelope Times

By clicking a time value and moving the mouse, you will alter the envelope time—the time it takes for the vector envelope to travel from the point before this time value to the point after this time value. You have two ways of doing this.

- Simple vertical dragging of the time parameter results in reaching all following points later (or sooner, respectively) in time.
- Dragging with Control held, you will shorten or lengthen the time of the following point by the same amount. This ensures that the adjacent, and all following, points retain their absolute time positions.

Resetting the Values of a Point

Reverting to default cursor positions in the Triangle and the Square is done by:

- Clicking in the Triangle while holding down the Option key. This sets all Oscillators to output the same level. The cursor is set to the middle of the Triangle.
- Clicking onto the Square while holding down the Option key sets the cursor to the center of the Square. Both axis values are set to zero.

Solo Point

The Solo Point button basically switches off the entire vector envelope generator. If Solo Point is set to on, no dynamic modulations are applied by the vector envelope. In this scenario, the currently visible cursor positions of the Triangle and Square are permanently in effect. These cursor positions match the currently selected vector envelope point.

If you select another vector envelope point (by clicking on it), you will engage its Triangle and Square cursor positions immediately. If Solo Point is set to on, the newly selected point will become the solo point.

Note: You can independently switch off the vector modulation of the Square by setting Vector Mode off, as outlined on page 292.

Env Modes Normal and Finish

If the Env Mode menu is set to Normal, the release phase (the phase after the sustain point) will begin as soon as you release the key (note off). The release phase will start from the vector envelope point where you released the key.

- If the loop is switched off, and the vector envelope reaches the sustain point, the sustain point will be played for as long as you hold the key.
- If the loop is switched on (see "Loop Mode" on page 298), and the loop point is positioned before the sustain point, the loop will be played for as long as you hold a key.
- If the loop is switched on, and the loop point is positioned after the sustain point, the loop will be played after the release of a key.

In Env Mode Finish, the vector envelope will not immediately commence the release phase when you release the key. Rather, it will play all points (for their full duration) until the last point is reached, regardless of whether you hold the key or release it.

- If the loop is switched off, the sustain point will be ignored. The vector envelope will end on its last point, regardless of whether you hold the key or release it.
- If the loop is switched on, the vector envelope will play all points until it reaches the loop point, and then play the loop for as long as the sound ends. It does not matter if the loop point is before or after the sustain point *S*.
- If the loop is switched on, and Loop Count is set to a value other than "infinite," the vector envelope will continue on to following points after the selected number of loop repeats. If Loop Count is set to "infinite," the number of segments after the loop is irrelevant. See "Loop Count" on page 299.

Curve

The Curve parameter sets the shape of the transition from point to point. You can choose between nine convex and nine concave shapes. There are also the two extreme forms; "hold+step" and "step+hold," which allow stepped modulation. Where "step+hold" jumps at the beginning of the transition time, "hold+step" jumps at the end.

Note: You can use "hold+step" to create stepped vector grooves with up to 15 steps.

Vector Envelope Loops

The vector envelope can—like any envelope—run in one shot (as long as the note is sustained) mode. It can also run several times or in an infinite cycle, much like an LFO. You can achieve this through the use of *loops*.

Note: The loop parameters might remind you of the loop parameters available for samples. Just to avoid any misunderstandings: The vector envelope only supplies control signals used for moving the cursor positions of the Triangle and Square. The audio of the ES2 is *not* looped at all.

Loop Mode

The ES2 features these loop modes:

- Off: If the Loop Mode mode is set to Off, the vector envelope runs in one shot mode from its beginning to its end—given that the note is held long enough. The other loop parameters are disabled.
- Forward: When Loop Mode is set to Forward, the vector envelope runs to the sustain point and begins to repeat the section between the loop point and sustain point periodically, always in a forward direction.
- *Backward:* When Loop Mode is set to Backward, the vector envelope runs to the sustain point and begins to repeat the section between the sustain point and loop point periodically, always in a backward direction.
- Alternate: When Loop Mode is set to Alternate, the vector envelope runs to the sustain point and returns to the loop point and back to the sustain point periodically, alternating in both a backward and forward direction.

Loop Rate

Just as every LFO has a speed (or *rate*) parameter, the loop can be set to cycle at a defined Loop Rate. And just like an LFO, the vector envelope Loop Rate can be synchronized to the project tempo automatically.



- If you switch the Loop Rate to as set, the duration of the loop cycle is equal to the sum of the times between the sustain and loop points. Click on the field labeled as set (below the Rate slider) to select.
- If you set the Loop Rate to one of the rhythmic values (sync, left half of the slider, 32 bars up to 64th Triplet Note), the Loop Rate fits to the project tempo.
- You also can set the Loop Rate in the small panel to the right half of the slider (free). The value indicates the number of cycles per second. Use the mouse as a slider to adjust.

Note: If Loop Rate is not switched to "as set," and the loop is activated (Loop Mode Forward, Backward, or Alternate), the times of the points between the loop and sustain points as well as the value for Loop Smooth are indicated as a percentage of the loop duration, rather than in milliseconds.

Loop Smooth

When Loop Mode is set to Forward or Backward, there will inevitably be a moment when a transition from the sustain point to the loop point occurs. In order to avoid abrupt cursor position changes, this transition can be smoothed through use of the Loop Smooth parameter.

- If Loop Rate is set to Sync or Free, the loop smoothing time will be displayed as a percentage of the loop cycle duration.
- If Loop Rate is set to as set, the loop smoothing time will be displayed in milliseconds (ms).

Loop Count

The loop cycle of the vector envelope doesn't need to be performed infinitely—you can have it cycle just a few times. Following the defined number of repetitions, the vector envelope will run from the sustain point onwards, just as with Loop Mode off. Use the mouse as a slider to set the Loop Count value. Possible values are 1 to 10 and "infinite."

Time Scaling

You can stretch and compress the entire vector envelope. As an example, to double the vector envelope's speed, it's not necessary to halve the time values of every point. All you need to do is set Time Scaling to 50%.

- The range for the Time Scaling parameter is from 10% to 1000%. It is scaled logarithmically.
- If the Loop Rate is "as set," the scaling also affects the loop. If not (Loop Rate = free or sync), the setting will not be affected by Time Scaling.

Fix Timing—Normalizing Time Scaling and Loop Rate

By clicking the Fix Timing button next to the Time Scaling parameter, the Time Scaling value will be multiplied by all time parameters, and Time Scaling will be reset to 100%. There will be no audible difference. This is simply a normalizing procedure, most similar to the normalizing function of the region playback parameters in Logic Pro.

In cases where a loop which was synchronized to the project tempo had been engaged (Loop Rate = sync), clicking Fix Timing will also switch the Loop Rate to "as set," thus preserving the absolute rate.

Effect Processor

The ES2 is equipped with an integrated effect processor. Any changes to this processor's effects settings are saved as an integral part of each sound program.



Despite the inclusion of this integrated effects processor, please feel free to process the ES2 with the other effect plug-ins included in Logic Studio. The sound and parameter set of the integrated effects unit is reminiscent of classic pedal effects, designed for the electric guitar. The use of guitar pedal effects on classic analog synthesizers was a standard practice amongst gigging musicians.

Distortion

At its Soft setting, the distortion circuit is somewhat like a tube overdrive, whilst Hard sounds like a fully-transistorized fuzz box. The Distortion control defines the amount of distortion, and Tone controls the treble portion of the output of the distortion process.

Chorus, Phaser, Flanger

These classic modulation effects and their parameters (Intensity and Speed) simulate the sound of analog effects of this kind, with one exception: They don't produce as much noise.

- A *chorus* effect is based on a delay line, the output of which is mixed with the original, dry signal. The short delay time is modulated periodically, resulting in pitch deviations. The modulated deviations, in conjunction with the original signal's pitch, produce the chorus effect.
- A flanger works in a fashion similar to that of a chorus, but with even shorter delay times, and the output signal being fed back into the input of the delay line. This feedback results in the creation of harmonic resonances which wander cyclically through the spectrum, giving the signal a metallic sound.
- A *phaser* is based on a mix of a delayed and an original signal. The delayed element is derived from an all-pass filter, which applies a frequency-dependent delay to the signal. This is expressed as a phase angle. The effect is based on a comb filter, which is basically an array of inharmonic notches (rather than resonances, as with the flanger), which also wander through the frequency spectrum.

Using Controls and Assigning Controllers

The section at the bottom of the ES2 interface provides three modes, accessed by clicking the buttons on the left:



- *Macro*: Shows a number of macro parameters which affect groups of other parameters.
- *MIDI*: Allows you to assign MIDI controllers to particular Router channels (see "MIDI Controllers A–F" on page 281).
- *Macro Only:* Replaces the ES2 interface with a dedicated (and much smaller) view that is limited to the macro parameters.

Macro Parameters

The macro parameters provide quick access to several linked (and related) parameters. As you alter any of the macro controls, you will see one, two, or more parameters in the ES2 interface update. As an example, adjusting the Detune macro control can simultaneously affect the Analog and Coarse and Fine Tune parameters.



Important: The impact of each macro control is completely dependent on the parameter values of the current setting. In some patches, a number of macro controls may have no effect.

Another bonus of the macro parameters is that they are settings-compatible with ES2-based GarageBand instruments. In other words, you can use the ES2 and some GarageBand synthesizer settings interchangeably.

Random Sound Variations

The ES2 offers a unique feature that allows you to vary the sound parameters randomly. You can define the amount of random variation, and can restrict the variations to specific sonic elements. The random sound variation feature will inspire and aid (or occasionally amuse) you when creating new sounds.



Clicking the RND button randomly alters the sound. The process is triggered by a single click and can be repeated as often as you like.

Note: To avoid possible misunderstandings: This feature has nothing to do with random real-time modulations. The random feature changes the parameters randomly with each mouse click. Real-time random modulations are performed with the random waveforms of the LFOs and by the Analog parameter, for random pitch settings.

RND Int

RND Int (Random Intensity) defines the amount of random parameter alteration. As you move the slider to the right, you will increase the amount of random variation.

The random sound variation feature always alters the parameters as they are currently set, not based on the memorized setting file. As such, clicking RND repeatedly will result in a sound which increasingly differs from the original setting. If want to check out several slight alterations of the current setting, you can reload the original setting, after each random alteration.

RND Destination

Some aspects of your sound may already be ideal for the sound you had in mind. As such, it may not be desirable to alter them. Say your sound setting has a nice percussiveness, and you'd like to try a few sonic color variations while retaining this percussive feel. To avoid the random variation of any attack times, you can restrict the variation to the oscillator or filter parameters, with the envelope parameters excluded from the variation process. To do so, set the RND Destination to Waves or Filters.

Please note:

- The Master Level, Filter Bypass as well as the three Oscillator On/Off parameters and the Vector/Router display options are never randomized.
- During randomizations of the vector envelope, the Solo Point will always be set to off.

You can restrict the random sound variation to the parameter groups listed below:

ΑII

All ES2 parameters, with the exception of the parameters mentioned above, will be altered.

All except Router and Pitch

All ES2 parameters, with the exception of all Router parameters and the basic pitch (*semitone* settings of the oscillators), will be altered. The oscillator fine tuning will be varied. This will result in more musically useful sounds.

All except Vector Env

All ES2 parameters, with the exception of all vector envelope parameters, will be altered. This maintains the rhythmic feel of a given setting.

Waves

Only the Wave and Digiwave parameters of the Oscillators will be altered. The other Oscillator parameters (tuning, mix, and modulations in the *Router*) are excluded.

DigiWaves

Digiwaves will be selected in all Oscillators. The Digiwave number will be altered. The other oscillator parameters (tuning, mix, and modulations in the Router) are excluded.

Filters

The filter parameters are varied. The parameters included are: *Filter Structure* (series or parallel wiring), *Blend*, *Filter Mode*, *Cutoff Frequency*, and *Resonance* for Filter 1 and 2, *Fatness*, *Filter FM* of Filter 2.

Envs

All envelope parameters of all three envelopes ENV 1, ENV 2 and ENV 3 are varied. The vector envelope is *excluded*.

LFOs

All LFO parameters of all LFOs are varied.

Router

All Router parameters of all Router channels are varied with all Intensities, Target, via, and Source parameters.

FX

All effects parameters are varied.

Vector Envelope

All vector envelope parameters are varied, including the XY routing of the Square.

Vector Env Mix Pad

The Oscillator mix levels (Triangle cursor positions) of the vector envelope points are altered. The rhythm and tempo of the modulation (the time parameters of the points) will not be altered.

Vector Env XY Pad Options

The Square cursor positions of the vector envelope points are altered. The XY routing won't be altered. The rhythm and tempo of the modulation (the time parameters of the points) will not be altered.

You can specify alterations in a single direction by choosing either:

- Vector Env XY Pad X only
- Vector Env XY Pad Y only

Vec Env Times

Only the time parameters of the vector envelope points will be altered.

Vec Env Structure

The vector envelope structure will be altered: All times, the sustain point, the number of points and all loop parameters.

Vec Env Shuffle Times

The vector envelope shuffle times (within loops) are altered: This includes the Loop Smooth value, if Loop Mode is set to Forward or Backward.

Note: It's recommended that you save any good sounds resulting from the RND process, as you work. Do this under a new name in the Settings menu of the plug-in window.

Tutorials

You will find the settings for these tutorials in the Tutorial Settings folder in the Settings menu (in the header of the ES2 window).

Sound Workshop

The Sound Workshop will guide you—from scratch—through the creation of commonly used sounds. The following tutorial section will also guide you through the sound creation process, but starts you off with a number of templates.

Sound Design From Scratch, Filter Settings, Digiwaves

The Analog Saw Init tutorial setting is designed for use as a starting point, when programming new sounds from "scratch." Professional sound designers like to use "scratch" settings when programming entirely new sounds, usually as follows: An unfiltered sawtooth wave sound without envelopes, modulations, or any gimmicks. This type of setting is also useful when getting to know a new synthesizer. It allows you to access all parameters without having to consider any pre-set values.

- Start with the filters, the heart of any subtractive synthesizer. Check out the four lowpass filter types 12 dB, 18 dB, 24 dB and fat (Filter 2) with different values for Cut (Cutoff Frequency) and Res (Resonance). Define Env 2 as filter envelope. This modulation wiring is pre-set in the Router.
- Set Filter Blend to its left-most position, which will allow you to listen to Filter 1 in isolation. In many circumstances, you'll probably prefer Filter 2, but Filter 1 has its advantages. In addition to the lowpass filter with 12 dB/octaves slope (Lo), Filter 2 also offers: a highpass, peak, bandpass (BP), and band rejection mode (BR). Filter 1's lowpass sounds "softer," when compared with Filter 2. It is best-suited to sounds where the filter effect is/should be less audible (Strings, FM sounds). Distorted TB-303 style sounds are more easily achieved with Filter 1.
- This setting is also ideal for checking out the oscillator waveforms. The analog
 waveforms can be set in the Editor view. In order to select the Digiwaves, set Osc 1
 Wave to Digiwave.

Three Detuned Sawtooth Oscillators and Unison Mode

"Fat" synthesizer sounds have always been popular, and are likely to remain so, given their use in modern trance, techno, R n' B, and other styles. The Analog Saw 3 Osc setting features three detuned oscillators, and sounds fat as it is. The following will introduce you to some additional tools to fatten the sound even more.

- Check out the 3-oscillator basic sound with different filter and envelope settings.
- Check out the chorus effect at different Intensities and Speeds.

Engage Unison mode and select a higher setting for Analog. As the sound is
polyphonic, each note is doubled. The number of notes that can be played
simultaneously will be reduced from 10 to 5. This will make the sound rich and broad.
Combining Unison and higher values for Analog will spread the sound across the
stereo or surround spectrum.

In many factory settings, the Unison mode is active. This demands a lot of processing power. If your computer isn't fast enough, you can switch off the Unison mode and insert an Ensemble effect in a bus, for use with several plug-ins. This will save processing power. Other ways to save CPU resources are by freezing or bouncing several software instrument tracks.

Extremely Detuned Monophonic Analog Sounds, Effects

The Analog Unison setting is a fat, heavily detuned, un-filtered basic sound. As with the example above, three sawtooth oscillators are used, but they are further detuned. The combination of Unison and Analog (set to a high value) is essential—but this time, monophonic mode is used to stack ten voices. Without further effects, the result is a huge lead sound, much like those used in countless dance and trance productions. With appropriate filter and envelope settings, electro sounds (that are ideal for arpeggiation and sequencing) can easily be set up.

- Set the Cutoff Frequency of Filter 2 to 0. This will activate the preset filter envelope. Feel free to check out different envelope settings.
- Switch Osc 1 to sound one or two octaves lower.
- Increase Drive or Distortion.
- Set Env 2 to be velocity sensitive. This allows for velocity sensitive filter modulations.
- Insert a delay effect in the instrument channel strip of the ES2. In order to delay several software instrument channels, you might prefer to insert the effect into a bus, which is accessed via each channel's Send.

Logic Studio incorporates reverb and delay effects, which are essential for many synthesizer sounds. These aren't integrated into the ES2, ensuring that no processing power is wasted unnecessarily.

Clean Bass Settings With One Oscillator Only

Not every sound needs to comprise of several oscillators. There are numerous simple, effective, sounds which make use of a single oscillator. This is especially true of synthesizer bass sounds, which can be created quickly and easily with the basic "Analog Bass clean" setting.

The basic sound is a rectangular wave, transposed down by one octave. The sound is filtered by Filter 2. What's special about this sound is its combination of Legato and Glide (portamento). When you play staccato, no glide effect will occur. When you play legato, the pitch will smoothly glide from one note to another. All keys must be released before striking a new key, in order to retrigger the envelopes.

- Check out different filter and envelope settings.
- Replace the rectangular wave with a sawtooth.
- Vary the Glide settings.

It's best to make your edits while a bass line is playing. Create a monophonic MIDI region, with most notes played staccato, but some legato. This can provide some interesting results with very long Glide values.

Distorted Analog Basses

In the "Analog Bass distorted" setting, Filter 1 is engaged, with high settings for Drive and Distortion. This filter is better suited to the creation of distorted analog sounds than Filter 2.

- Check out Filter 2 by setting Filter Blend to its right-most position. You'll hear that Filter 1 works better with distorted sounds.
- In order to control the filter modulation, move the green sliders of the first modulation channel in the Router. This controls the modulation intensity.

FM Intensity and Frequency

The FM Start setting is great for familiarizing yourself with linear Frequency Modulation (FM) synthesis. You'll hear an un-modulated sine sound, generated by Oscillator 1. Oscillator 2 is switched on, and set to produce a sine oscillation as well, but its level is set to 0: Just push the cursor in the triangle in the uppermost corner.

In the ES2—Oscillator 1 is always the carrier, and Oscillator 2, the modulator. IN other words, Oscillator 2 modulates Oscillator 1.

 Adjust the intensity of the frequency modulation by slowly moving the wave selector from Sine to FM. You will hear a typical FM spectrum, with the carrier and modulator set to the same frequency.

- Alter the modulator frequency (Oscillator 2) by adjusting Fine Tune from 0 c to 50 c. You'll hear a very slow frequency modulation, that can be compared to the effect of an LFO. The frequency modulation, however, takes place in the audio spectrum. It is adjusted in semitone steps by the frequency selector. Check out the entire range from -36 s to +36 s for Oscillator 2. You'll hear a broad spectrum of FM sounds. Some settings will remind you of classic FM synthesizer sounds.
- Select other waveforms for Oscillator 2. Sine is the classic, standard FM waveform, but other waveforms lead to interesting results as well, especially the Digiwaves.
- You will achieve further interesting results by altering the carrier (Oscillator 1) frequency. Check out the entire range: from –36 s to +36 s semitones here, as well. The odd intervals are especially fascinating. Note that the basic pitch changes when doing so.

Controlling FM Intensity by an Envelope and FM Scaling

In the FM Envelope setting, you can control the FM intensity with an envelope, generated by Envelope 2. The modulation target is the range which falls between Sine and FM—in the Oscillator wave selector. The first Router channel is used for this. You can control a wider range through the use of additional modulations, which have been pre-prepared for you. All you need to do is set their values. As these modulations work without velocity sensitivity, you can set them in the Editor view—by moving both the lower and upper fader halves to their top-most positions.

- Set the second modulation channel to 1.0. You'll hear how the modulation now "wanders" through a broader sound range.
- Set modulation channels 3 and 4 to a value of 1.0 as well, and listen to the increase in the sound range.
- Following such drastic augmentations to the modulation range, the sound will become uneven across the keyboard. In the lower and middle ranges, it sounds nice, but in the upper key range the FM intensity appears to be too severe. You can compensate for this effect by modulating the Osc 1 Wave target by keyboard position (kybd)—in modulation channels 5 and 6. This results in a keyboard scaling of the FM intensity.
- As the sound range is so vast (due to the 4 modulations), two modulation channels
 are required to compensate for this. Set the lower slider halves to their lowest
 positions. Good keyboard scaling is essential for any FM sound.

FM With Drive and Filter-FM

The FM Drive setting illustrates how dramatically the character of FM sounds can be altered by applying Drive and Filter FM. The results are reminiscent of the feedback circuits of classic FM synthesizers.

- Check out different Drive and Filter FM settings.
- Lower the Cutoff Frequency of Filter 2 to 0. Envelope 2 modulates Filter 2. This modulation routing is already present in the setting.

FM With Digiwaves

In the FM Digiwave setting, a Digiwave is used as an FM modulator. This results in a bell-like spectra from only two operators. Using traditional FM synthesis, this type of timbre could normally only be produced with a larger number of sine oscillators.

In order to create a fatter, undulating, and atmospheric quality to the sound, the polyphonic Unison mode has been engaged. Filter and amplitude envelopes have been preset to shape the sound.

- Check out the variety of Digiwaves, as FM modulation sources.
- Check out different Analog parameter values.

FM With Wavetables

You can program the most vivid FM sounds when the modulation source morphs between different Digiwaves. The morphing in the FM Digiwave setting is controlled by LFO 2. The tempo of LFO 2 (and therefore the morph) depends on the sequencer tempo (here: 2 bars).

- Set LFO 2 to different waveforms. Lag S/H (smooth random), in particular, should be fun.
- Check out different FM intensities and oscillator frequencies.
- Alter the modulation intensity of the first modulation channel (LFO2 modulates Osc2 Wave) and the LFO 2 rate.

Distorted FM in Monophonic Unison

The FM Megafat setting is well-suited for distorted basses and guitar-like sounds. This sound gets rather "rude" in the upper key range. This cannot be compensated for with key scaling, but not every sound has to be "nice" across the entire keyboard range!

- Check out extreme detunings by adjusting the Analog parameter.
- Check out the Flanger with this sound.
- Engage the filter envelope by lowering the Cutoff Frequency of Filter 2 down to 0.
- Add some Glide to lead sounds.
- As always, when it comes to FM: You can dramatically alter the sound by varying the frequencies of the oscillators. Make sure you check out the odd intervals, as well.

FM With Unusual Spectra

If you're unconcerned with the pitch of your sound, you can get the weirdest spectra out of odd frequency ratios (oscillator intervals).

The FM Out of Tune setting offers a bell-like sound, reminiscent of a ring modulator. It was achieved through a setting of 30 s 0 c, with the modulator set to a value of 0 s 0 c. Sounds like this were very commonly used in the electronic music of the eighties, and have undergone a resurgence in popularity in modern ambient and trance music styles.

You can further develop the sound by applying filtering, envelope modulations, and effects. There is, however, one small problem—the sound is out of tune.

- Use Oscillator 3 as a reference for the tuning of the FM sound, by moving the cursor in the Triangle.
- You'll notice that the sound is 5 semitones too high (or 7 semitones too low, respectively).
- Transpose both oscillators 1 and 2 five semitones (500 ct) lower. Transposing them upwards is not practical, as you'd need to select 37 s 0 c for Oscillator 1, which maxes out at 36 s 0 c.
- It's important to maintain the frequency ratio (interval) between Oscillators 1 and 2.

 This means that Oscillator 1 will sound at 25 s 0 c and Oscillator 2 at -5 s 0 c.

Slow and Fast Pulse Width Modulations With Oscillator 2

Pulse Width Modulation (PWM) is one of the most essential features of any sophisticated analog synthesizer.

- Choose the PWM Start setting, and move the Wave control slowly back and forth between the rectangular and the pulse wave symbols. Both are green. What you will hear is a (manual) pulse width modulation.
- Choose the PWM Slow setting. Here, LFO 1 controls the pulse width modulation source, not your manual movements. The result should be quite similar.
- Raise the LFO 1 rate from its pre-set value of 0.230 to 4.400. The result is a classic, fast PWM.
- In this, and the next step, the PWM will be set so that it sounds slower in the lower keyboard range, and faster in the upper range. This is desirable for many sounds, such as synthetic strings. First, reduce the LFO 1 Rate to 3,800.
- Change the modulation intensity of the second Router channel (target = LFO1 Rate, Source = Kybd) to 0.46. This will alter the scaling of the PWM, making it sound faster in the treble range. You can also hear this type of effect in the PWM Scaled setting.
- **Tip:** Avoid Drive and Distortion with PWM sounds.

Pulse Width Modulation With Two Oscillators, PWM Strings

In order to make the sound fatter, add Oscillator 3, which can also be pulse width modulated. In fact, even the first oscillator can deliver PWM. In the PWM 2 Osc setting, both oscillators are detuned quite significantly. Develop your own personalized PWM string sound, using this setting as your base.

- Adjust the Chorus intensity. You'll probably choose higher values, which make the sound rather broad.
- Program Envelope 3 according to your taste. You should, at the very least, raise the
 attack and release times. Define it to react to velocity, if you prefer. If you want to use
 the sound for something other than a simple pad, a shorter Decay Time and a lower
 Sustain Level of about 80 to 90% may be more appropriate.

- Reduce the Cutoff Frequency and Resonance of Filter 1 to make the sound softer.
- Save the new setting.
- Compare the result with the PWM 2 Osc setting. You'll hear that the sound has undergone a remarkable evolution.
- Compare it to PWM Soft Strings, which was created as described above. You'll probably notice a few similarities.

Ring Modulation

A ring modulator takes its two input signals and outputs the sum and difference frequencies of them.

In the ES2, Oscillator 2 outputs a ring modulator, which is fed with a square wave of Oscillator 2 and the wave of Oscillator 1, when Ring is set as Oscillator 2's waveform.

Odd intervals (frequency ratios) between the oscillators, in particular, result in bell-like spectra, much like those heard in the RingMod Start setting.

As discussed in the "FM Out of Tune" section, on page 309, the third oscillator can be used as a tuning reference, in order to maintain a kind of basic tuning. On occasion, you may find that it's nice to leave the sound out of tune—for use as a source of overtones and harmonics for another basic wave, supplied by Oscillator 3.

Try to program an atmospheric bell sound. Use your imagination, but here are some hints that may help:

- Experiment with the various frequency ratios of Oscillators 1 and 2. You may want to use the 29 s 0 c/21 s 0 c ratio, which doesn't sound out of tune at all. Ring modulation is not only useful for bell-like sounds, it's also good for a great variety of spectra which tend to sound pretty weird at lower frequency settings. Also try alterations to the fine tuning of the Oscillators.
- Check out an Intensity of 50% and a Rate, set to around 2/3 of the maximum value, for the Chorus effect.
- Set the Attack and Release Times of Envelope 3 to taste.
- Check out Drive and Filter FM, if you like your sounds a little "out of control."
- The rest is up to you!

Oscillator Synchronization

If you select the synced square and sawtooth waveforms for Oscillators 2 and 3, they will be synchronized with Oscillator 1. In the Sync Start setting, only Oscillator 2 is audible and Oscillator 3 is switched off.

Typical sync sounds feature dynamic frequency sweeps over wide frequency ranges. These frequency modulations (the sweeps) can be applied in various ways.

• Try the pre-programmed pitch modulation, assigned to the modulation wheel, first.

- In the second Router channel, an envelope pitch modulation has been preprogrammed (target = Pitch 2, Source = Env 1). Setting the minimum value to 1.0 results in a typical sync envelope. Also check out shorter Decay Times for Envelope 1.
- In order to avoid a sterile, lifeless sound (after the decay phase of the envelope), you may also want to modulate the oscillator frequency with an LFO. Use the third Router channel: set the minimum modulation applied by LFO 1 to about 0.50.
- Substitute the synchronized square wave with the synced sawtooth wave, and see if you like the results.

Note: Pulse width modulation is also available via the synchronized square wave of Oscillators 2 and 3. A modulation of the wave parameters (of these two Oscillators) results in a PWM, when the synced square wave is selected.

First Steps in Vector Synthesis

This tutorial section provides some useful hints for vector envelope programming. In the Vector Start setting, the "mix" of the oscillators is controlled by the vector envelope. Fach oscillator has been set to a different waveform.

- Switch from the Router view to Vector view.
- In its basic (default) setting, the vector envelope has 3 envelope points. Point 1 is the start point, Point 2 the sustain point, and Point 3 is the target in the release phase. By clicking the points, you can see that the mix is always set to 100% for Oscillator 1, in the Triangle.
- Click Point 2, and move the Triangle cursor to Oscillator 2. You'll hear a square wave, instead of Oscillator 1's sawtooth.
- Engage the vector envelope by switching the Solo Point parameter off. As long as it is switched on, you will only hear the selected point, with no dynamic modulation. Having switched Solo Point off, you'll hear the sound moving from saw to square, with every triggered note.
- Alter the pre-set time of 498 ms, between points 1 and 2.
- While holding Shift, click between points 1 and 2. This will create a new Point 2, and the point formerly known as Point 2 will become Point 3. The total time span between Point 1 and Point 3 is divided into the times between Points 1 and 2, and 2 and 3. The division takes place at the click location. If you clicked at the exact midpoint, the new time spans are equal.
- Grab the newly created Point 2, and move its cursor in the Triangle to Oscillator 2.
- Grab Point 3, and move its cursor in the Triangle to Oscillator 3. Listen to the three oscillators morphing from sawtooth to square to a triangular wave at the final sustain point.
- Grab Point 4 (the end point) and move its cursor in the Triangle to Oscillator 1 (if it is not already there). Listen to how the sound returns to Oscillator 1's sawtooth wave, following the release of the key.

Vector Synthesis—XY Pad

The vector envelope example starts where the first one left off. You have a simple vector envelope consisting of 4 points, which is set to modulate the oscillator mix (the Triangle).

In this example, the vector envelope will be used to control two additional parameters: The Cutoff Frequency of Filter 2 and Panorama. These are pre-set as the X and Y targets in the Square. Both have a value of 0.50.

- Switch on Solo Point, in order to more easily listen to the settings for the single points.
- Click Point 1. You will only hear Oscillator 1's sawtooth.
- Move the cursor in the Square to the hard left, which results in a low Cutoff Frequency for Oscillator 2.
- Click Point 2. You will only hear Oscillator 2's rectangular wave.
- Move the cursor in the Square all the way down, which results in the right-most Panorama position.
- Click Point 3. You will only hear Oscillator 3's triangular wave.
- Move the cursor in the Square all the way up, which results in the left-most Panorama position.
- Switch on Solo Point. The sound begins with a strongly filtered sawtooth wave and turns into an-unfiltered square wave. It initially sounds from the right, and then moves to the left while morphing into a triangular wave. After releasing the key, the saw sound will be heard.

Vector Synthesis Loops

The basic sound of the Vector Loop setting (without the vector envelope) consists of three elements:

- Oscillator 1 delivers a metallic FM spectrum, modulated by Oscillator 2's wavetable.
- Oscillator 2 outputs cross-faded Digiwaves (a wavetable), modulated by LFO 2.
- Oscillator 3 plays a PWM sound at the well-balanced, and keyboard-scaled, speed of LFO 1.

Unison and Analog make the sound fat and wide.

These heterogenic sound colors will be used as sound sources for the vector loop.

A slow, forward loop is pre-set. It moves from Oscillator 3 (PWM sound, Point 1) to Oscillator 1 (FM sound, Point 2), then to Oscillator 3 again (PWM, Point 3), then to Oscillator 2 (Wavetable, Point 4) and finally, it returns to Oscillator 3 (PWM, Point 5). Points 1 and 5 are identical, avoiding any transition from Point 5 to Point 1 in the forward loop. This transition could be smoothed out with Loop Smooth, but this would make the rhythmic design more difficult to program.

The distances between the points of the vector envelope have been set to be rhythmically exact. Given that Loop Rate has been engaged, the time values are not displayed in ms, but as percentages. There are four time values (each at 25%), which is a good basis for the transformation into note values.

- Switch off the vector envelope by setting Solo Point to on. This allows you to audition the individual points in isolation.
- Take the opportunity to alter the cursor positions in the Square according to your taste. As in the example above, the X/Y axes of the Square control the Cutoff Frequency of Filter 2, and the Panorama position. Adjustments to these make the sound more vivid.
- Activate the vector envelope by setting Solo Point to off. Check the result, and finetune the cursor positions in the Square.
- Alter the Loop Rate from the pre-set value of 0.09 up to 2.00. You will hear a periodic
 modulation, much like that of an LFO. At this point, the modulation is not
 synchronized with the project tempo. To synchronize the loop speed with the project
 tempo, move the Rate cursor to the very left, and set a note or bar value.
- You can create faster rhythmic note values by clicking between two points and setting the new time values (resulting from the division which occurs) to, say, 12.5%.

Bass Drum With Self-Oscillating Filter and Vector Envelope

Electronic kick drum sounds are often created with modulated, self-oscillating filters. This approach can also be taken with the ES2, particularly when the vector envelope is used for filter modulation. An advantage of the vector envelope, in comparison with conventional ADSR envelopes, is its ability to define/provide two independent decay phases. The distortion effect applies the right amount of drive without losing the original sonic character of the drum sound.

Note: In order to make the Vector Kick setting really punchy, you must activate Flt Reset. This is necessary, because all Oscillators are switched off in this setting, and the filter needs a little time to start oscillating. At the start of each note, Flt Reset sends a very short impulse to the filter—making it oscillate from the outset!

Through tweaks to the Vector Kick setting, you'll probably be able to create any dancefloor kick drum sound your heart desires. These are the parameters which allow for the most efficient and significant variations:

- Filter2 slopes 12 dB, 18 dB, 24 dB
- · Distortion Intensity, Soft/Hard
- Envelope 3's Decay Time (D)
- Vector Envelope Time 1 > 2 (Pre-set: 9.0 ms)
- Vector Envelope Time 2 > 3 (Pre-set: 303 ms)
- Vector Time Scaling

Percussive Synthesizers and Basses With Two Filter Decay Phases

As with the Vector Kick, the Vector Perc Synth setting uses the vector envelope to control the filter cutoff frequency (with two independently adjustable decay phases). This would not be possible with a conventional ADSR envelope generator. Try creating further percussive synthesizers and basses by varying these parameters:

- Vector Envelope Time 1 > 2 (= Decay 1),
- Vector Envelope Time 2 > 3 (= Decay 2),
- · Vector Time Scaling,
- Cursor positions in the Square for points 1, 2, and 3 (= Cutoff Frequency),
- Choosing other waveforms.

Templates for the ES2

Welcome to a brief programming tour of the ES2!

While working on the factory preset programming for the ES2, a number of testers, sound programmers and other people involved in the project indicated that it would be nice to start their programming work from templates, rather than entirely from scratch.

Needless to say, creating templates which cover all sound genres is something of a mission impossible. As you spend time familiarizing yourself with the ES2's architecture, you'll start to understand why ...

Nevertheless, this programming tour for the ES2 was included as a part of the toolbox to help you learn and understand the ES2's architecture, through experimentation. You'll find that this approach is fun. You'll also discover, as you're working through a number of simple operations, that results will come quickly when starting to create your personal sound library.

As you become more familiar with the ES2, and what its myriad of functions and parameters do, you can create your own templates—for use as starting points when designing new sounds.

Clean Stratocaster (Slap Strat)

The target of this preset was the sound of a Stratocaster, with the switch between bridge and middle pickup in the middle position (in phase). It attempts to emulate the noisy twang, typical of this sound's characteristics.

This might be a useful template for emulations of fretted instruments, harpsichords, clavinets, and so on.

Have a look at its architecture:

Osc 1 and 3 provide the basic wave combination within the Digiwave field. Changing the Digiwaves of both (in combination) delivers a huge number of basic variations—some also work pretty well for electric piano-type keyboard sounds.

Osc 2 adds harmonics with its synced waveform, so you should only vary its pitch or sync waveform. There are a couple of values which can be changed here, which will give you a much stronger, more balanced signal.

An old trick, which delivers a punchy attack, was used—that the use of a naked wave wouldn't deliver—even with the best and fastest filters available: You use an envelope (in this case, Env. 1) for a quick "push" of a wavetable's window (or all wavetables together, where it makes sense).

So set up Envelope 1's Decay time for this short push, moving the wave selectors for all Oscillators on the attack. (... actually it makes no sense on the synced sawtooth Oscillator, No. 2—it just works this way ...)

So you can vary the punchiness of the content between:

- Envelope 1's contribution to the overall attack noise, changing decay speed (a slow one gives you a peak, a long one gives you a growl, as it is reading a couple of waves from the wavetable).
- Modulation destination: you can always assign this to each of the Oscillators separately.
- Start point (you vary the wave window start with minimum/maximum control of EG1/Osc.waves modulation: negative values for a start wave before the selected wave, positive starts from a position behind the selected wave and rolls the table back) ...
- Feel free to experiment with this wavetable-driving trick. The growl effect works pretty well for brass sounds, and some organs absolutely shine with a little click, courtesy of a wavetable push.

Envelope 2, which controls the filter, provides a slight attack when used for "slapped" characteristics. Setting it to the fastest value eliminates the wah-like attack (and don't worry, there's enough punch left).

For playing purposes, you'll find that LFO 2 is used as a real time source for vibrato. It is assigned to the mod wheel and pressure.

Don't concern yourself too much with the different wheel and pressure settings. Feel free to change them!

Velocity is set up to be very responsive because many synthesizer players don't strike keys like a piano player would with a weighted-action "punch." As such, you should play this patch softly, or you may find that the slap tends to sweep a little. Alternately, you can adjust the sensitivity of the filter modulation's velocity value to match your personal touch.

Also feel free to increase the Voices to maximum; six strings should be enough for a quitar, but for held or sustained notes, a few extra voices may come in handy.

The Big Twirl, Basically (Wheelrocker)

This quite ordinary organ patch doesn't hold any deep, high-end sound design secrets: it is just a combination of three oscillators—with mixed wave levels. You'll probably find a different combination, which more closely matches your vision of what an organ sounds like. Check out the Digiwaves.

Focus your attention on the mod wheel's response: hold a chord, and bring the wheel in by moving it slowly upwards, until you reach the top (maximum).

The intention behind the programming of this (mod wheel) modulation is to simulate an accelerating Leslie rotor speaker.

The modulation routings do the following jobs:

- Modulation 1 (Cutoff 1) assigns envelope 2 to Filter 1 (the only one used for this patch), and produces a little organ key click with the envelope. The filter is opened slightly (with Keyboard as via) when playing in the higher keyboard range (with the maximum value).
- Modulation 2 and 3 (Pitch 2/Pitch 3) bring in LFO 1 vibrato, and both Oscillators are modulated out of phase.
- Modulation 5 reduces the overall volume—according to personal taste, the organ's level shouldn't increase too drastically when all modulations are moved to their respective maximums.
- Modulations 6 and 7 (Pitch 2/Pitch 3) detune Oscillators 2 and 3 against each other, within symmetrical values (to avoid the sound getting out of tune, overall). Again, both work out of phase with modulations 2 and 3; Oscillator 1 remains at a stable pitch.
- Modulation 8 brings in LFO1 as a modulator for panorama movement—this patch
 changes from mono to stereo. If you would prefer a full stereo sound, with a slowly
 rotating Leslie in its idle position, just set an amount equal to the desired minimum
 value, thereby achieving a permanent, slow rotation. Another modification you may
 wish to try is a higher value, resulting in more extreme channel separation.
- Modulation 9 speeds up LFO 2's modulation frequency.
- Modulation 10: A little Cutoff was added to Filter 1, increasing the intensity of the big twirl.

Please feel free to find your own values. While doing so, keep in mind the fact that there are two modulation couples, which should only be changed symmetrically (Mod. 2 and 3 work as a pair of twins, and also Mod. 6 and 7). So, if you change Pitch 2's maximum to a lower minus value, remember to set Pitch 3's maximum value to the same positive amount (same goes for modulation pair 6 and 7).

You can also bring in LFO 2 to increase the pitch diffusion against LFO 1's pitch and pan movements. Just exchange it for LFO 1 on modulation 2 and 3—but note that there will be no modulation source for the Leslie acceleration—so you'll need to use it in a static way, just fading it in. Alternately, you'll need to sacrifice one of the other modulations in favor of a second twirl.

For another stereo modification of the idle sound, you can use the patch in *Unison* mode with a slight detune (please adjust the analog parameter for this).

Something Horny (Crescendo Brass)

First up—the oscillators are used for the following tasks:

- Oscillator 1 provides the basic brass wave—sawtooth.
- Oscillator 2 provides a—not particularly "brassy"—pulse wave, which brings in the ensemble. It is pulse-width modulated by LFO 1 (Modulation 4).

Note: The following critical point should be taken into account for any modulations. There are four (4) parameters, which behave in an entirely different fashion, once one of them is changed. As such, all four must be changed when making adjustments:

- You may adjust the initial pulse width of Oscillator 2's wave parameter. A "fat" position, close to the ideal square wave, has been chosen for this patch—in order to program a full, voluminous synth-brass sound.
- Modulation 4 adjusts the modulation intensity, which means: how far does the range differ from fat to narrow, when being pulse width modulated? Set with the *Minimum* parameter.
- The rate of LFO1 directly controls the speed of the movement of the pulse width modulation. For this patch, both LFOs are used, to achieve a stronger diffusion effect at different modulation speeds.
- Tip: You should use LFO1 for all permanent, automatic modulations because you are able to delay its impact with its EG parameter. You may use LFO 2 for all real-time modulations, which you intend to access via modwheel, pressure, or other controls while playing.

A keyboard assignment was set up as the Modulation 4 source. This is because all pitch, or pulse-width, modulations tend to cause a stronger detuning in the lower ranges, while the middle and upper key zones feature the desired diffusion effect. When using this parameter, you should initially adjust the lower ranges until an acceptable amount of detuning (resulting from the modulation) is reached. Once set, check whether or not the modulations in the upper zones work to your satisfaction. Adjust the relationship between intensity (Max) and scaling (Min) values.

Oscillator 3 generates a Digiwave, which is "brassy" enough, within the overall wave mix. As an alternative to the Digiwave, another modulated pulse wave could be used to support the ensemble, or another sawtooth wave—to achieve a "fatter" sound, when detuning it with Oscillator 1's sawtooth wave.

The primary aim, however, is to have a little bit of "growl," achieved through a short wavetable push, as described for the Stratocaster patch, on page 315. This configuration is set up in Modulation 3 (Oscillator 3 Wave moved by Envelope 1's Decay).

Other Controls

Envelope 1 also effects the pitch of Oscillator 2 against Oscillator 3. This results in both pitches clashing with each other, and also with the stable pitch of Oscillator 1 (in the attack phase of the sound).

The filter envelope's design closes with a short stab in the attack phase, then reopens for a slower crescendo phase.

A further real-time crescendo has been assigned to the mod wheel, which also brings in an overall pitch modulation, controlled by LFO 2.

In addition to all of this, a "contrary" real-time modulation (by pressure)—which closes the filters—has been programmed. This allows you to play with an additional decrescendo, remotely controlled by touch. Try to get a feel for the patch's response. You'll find that it offers quite a few controls for expression: velocity, pressure after note on, and pressure in advance. Listen to what happens when pressing with the left hand before hitting a new chord with the right hand, and allowing the swell come in.

MW-Pad-Creator 3

This is an attempt to create a patch which is able to automatically generate new patches.

The Basics

Again, Oscillator 2 is used for a pulse width modulation—which creates a strong ensemble component (please refer to "Something Horny (Crescendo Brass)" on page 318, for further information).

Oscillators 1 and 3 are set to an initial start wave combination within their respective Digiwave tables. You can modify these if you wish, and start with a different combination of Digiwaves from the outset.

Modulation 3 "drives" the wavetables of all three Oscillators, via the mod wheel. Put simply: you can simultaneously scroll through the Oscillator 1 and Oscillator 3 wavetables, and change Oscillator 2's pulse width—by moving the mod wheel.

Try a careful, very slow movement of the mod wheel, and you'll hear drastic changes within the wave configuration. Each incremental position of the wheel offers a different digital pad sound. Avoid rapid movements, or this will sound like an AM radio.

Another potential modification procedure is hidden in the modulation intensity of the Oscillator 1, 2, and 3 wave parameters. As mentioned for the Stratocaster patch, the value of this intensity parameter assigns both the step width, and direction, through the wavetables. You may try modifications to the amount, using positive or negative values.

An interesting side-effect of FM assignment to Filter 2 (Modulation 4/Lowpass Filter FM) occurs when the mod wheel is moved to higher positions: the frequency modulation of the filter is increased, causing all cyclical beats (vibrating pitches, detunes, pulse width) to be emphasized. This also adds a rough, "hissing" quality to the overall sound character.

FM offers vast scope for experimentation, and you can decide between:

- An initial FM, using Filter 2's FM parameter, which you can redraw (set a negative modulation amount for Modulation 4's maximum) by moving the mod wheel to its top position.
- Or you may have permanent FM (and another modulation setup, saved for a different assignment). You can also switch off FM, if you consider its effect too dirty sounding.

Real-time control is via pressure for a vibrato (Modulation 10), and also for a slight opening of the Cutoff to emphasize the modulation (Modulation 9).

Another Approach to Crybaby (Wheelsyncer)

Never obsolete—and undergoing a renaissance in new popular electronic music: Sync Sounds

The technical aspects of forcing an Oscillator to sync are described in "Sync" on page 259. Here's the practical side of the playground.

Wheelsyncer is a single-oscillator lead sound, all others are switched off.

Although Oscillator 2 is the only one actively making any sound, it is directly dependent on Oscillator 1.

If you change Oscillator 1's pitch or tuning, the overall pitch of the sound will go out of tune, or will be transposed.

The pitch of Oscillator 2 provides the tone-color (or the harmonics) for the sync sound. Pitch changes are controlled by modulation 7—Oscillator 2 pitch is assigned to the mod wheel.

If you move the wheel, you can scroll through the spectrum of harmonics that have been programmed—for real-time changes. Any modification here starts with the pitch of Oscillator 2 itself, which is set to three semitones below the overall pitch. Feel free to start with a different pitch for Oscillator 2; it won't effect the tuning of the patch.

The next modification may be modulation 7's intensity (or the interval). The maximum value has been chosen—if this is too extreme for your needs, feel free to reduce it.

Another modification lies in the tone color of the lead sound itself. Oscillator 1 is switched off, as the patch is OK as it is. If you switch it on, all Oscillator 1 waveforms; from Digiwaves, standard waveforms, or a sine wave (which can be further modulated by FM) are available for use.

All real-time controls are via the mod wheel: It is used for opening the filter on modulation 6, a panning movement on modulation 8, and acceleration of panning movement on modulation 9. If you have deeper modulation ambitions, please refer to "The Big Twirl, Basically (Wheelrocker)" on page 317, where a similar setup is used for a Leslie speaker simulation.

EVB3 22

The EVB3 software instrument mimics the sound and features of the Hammond B3 and Leslie sound cabinet.

The EVB3 simulates an organ with two manuals (keyboards) and a pedalboard—each of which can have its own registration (sound settings). The sound generation process fully simulates the tone wheel generators of an electro-mechanical Hammond organ, down to the smallest detail. This includes certain charming flaws, such as the Hammond's enormous level of crosstalk and the scratchiness of the key contacts. You may adjust the intensity of these peculiarities to meet your tastes. This flexibility allows you to generate flawlessly clean through dirty and raunchy sounds—and everything in-between.

The EVB3 simulates different types of Leslie sound cabinets with rotating speakers, with and without deflectors. Beyond this the EVB3 also incorporates three tube overdrives with different tonal characteristics, an equalizer, a wah wah and a reverberation effect. You may freely define the signal flow of these effects as you wish. You can also set the stereo intensity of the microphone position as desired.

If you're familiar with the original B3, you'll remember the inverted (black) keys of the lowest octave on each manual. These inverted keys are switches that recall preset registrations (a preset of your drawbar settings). This feature is emulated by the EVB3 as well, however you won't need a screwdriver to change the registration settings of your presets. A morphing feature allows seamless crossfades between two registrations, without the need for an external fader box. The EVB3 can be played with two manuals and a MIDI pedalboard, if you wish (and own these devices). It also offers functions which allow you to play all registers (Upper, Lower, and Pedal) with a single-manual master keyboard.

MIDI Setup

If you want to fully exploit all features of the EVB3, you will need a MIDI (bass) Pedal unit, and two 73-key MIDI keyboards. As the EVB3 also emulates the B3's preset keys, the lowest octave of attached MIDI keyboards can switch the EVB3 registrations, just like the original B3. Please read the following section for more information.

The EVB3 can, of course, be played with single-manual keyboards with the standard 61 keys (5 octaves C to C). Please read "Keyboard Split" on page 326 for more information.

Playing Both Manuals and the Pedals Live

The EVB3 receives the notes for the Upper and Lower manuals, and the Pedalboard, on three subsequent MIDI channels.

Note: The MIDI channel (MIDI Cha) of the instrument must be set to All, and Keyboard Mode must be set to Multi.

The default Upper-manual receive channel is MIDI channel 1, Lower manual is MIDI channel 2, and Pedal registers receive on MIDI channel 3. If your master keyboard sends MIDI notes on channel 1, it will play the Upper manual, if it sends on channel 2, it will play the Lower manual, and if it sends on channel 3, it will play the Pedal register.

You may use any of your MIDI interface inputs for your master keyboard(s) and/or Pedalboard. You can also use a single-manual master keyboard—with different keyboard zones or a keyboard split feature—that sends data on different MIDI channels. Regardless of the input device(s) used, the only relevant factor is the MIDI send channel!

Logic Pro records the channel information of incoming notes. With most other MIDI and software instruments, this information is not used at all. This is due to the MIDI channel setting (in the Inspector's track parameters), which has priority, and overrides the original channel information. This can be circumvented by setting MIDI Cha to All, which is recommended for the EVB3. This will force the original channel information to be used. This enables you to make use of the two manuals, and Pedal register of the organ directly and simultaneously—when playing live.

Note: Please read the users manual of your master keyboard, to learn how to set its MIDI transmission channel, often abbreviated as TX Channel.

Changing MIDI Channels

You also can set the EVB3 to receive on MIDI channels other than its default configuration. This is done with the Basic MIDI Ch parameter in the Controls view's General section. This parameter assigns a MIDI channel to the Upper manual. The receive channel number for Lower is always one channel number higher than the channel assigned to the Upper manual. The Pedal register receive channel is always two channel numbers higher than the channel assigned to the Upper manual. In other words, if you set the Basic MIDI Ch parameter to 8, the Upper manual will receive on channel 8, the Lower manual will receive on channel 9, and the Pedal register will receive on channel 10.

To adjust the basic MIDI channel parameter:

- 1 Choose View > Controls in the plug-in window to access Controls view.
- 2 Drag the slider or click the arrow buttons to change the MIDI channel number.

Important: Basic MIDI Ch only works if Keyboard Mode is set to Multi.

When the Basic MIDI Ch is set to 16, the Lower manual receives on channel 1, and the Pedal register on channel 2. When the base MIDI Ch is set to 15, Lower receives on channel 16, and the Pedal register receives on channel 1.

Note: This parameter becomes especially useful in live performance situations, particularly if you need to change the MIDI transmission channel of your master keyboard in order to play other sound generators.

Keyboard Ranges of the Upper and Lower Manual

The lowest playable MIDI note is 36 (C1). The range of the preset keys is note # 24 to 35 (C0 to B0).

Note: 128 notes are defined in the MIDI specification, but even the largest master keyboards are only equipped with 88 keys, just like a concert grand piano.

For example: If your master keyboard ranges from C to c (5 octaves—61 keys), and the Region and Track parameters of Logic Pro are set to zero (Transpose. = 0), you can play the entire keyboard range—every possible note of the EVB3. The preset (registration) keys are positioned one octave lower. If you set Transpose. to -12, you can use the lowest octave to switch between presets.

Keyboard Split

The EVB3 can also be played perfectly with a single MIDI keyboard (one manual) that is only capable of transmitting on one MIDI send channel. You can split the keyboard in order to play Upper, Lower, and Pedal sounds on different keyboard zones.

In the parameter field in the bottom center of the interface, set Keyboard Mode to Split.



Set the keyboard zones with the UL Split and LP Split parameters, in conjunction with the Set buttons. The abbreviations are for: Upper/Lower and Lower/Pedal.

To set the keyboard zone:

- 1 Click on the appropriate Set button (it will turn orange).
- 2 Press the desired note on your MIDI keyboard.

If you select the same value for both split points, the Lower manual is not heard/active. If the LP Split is set to a value above the UL Split, the other splitpoint is moved (and vice versa).

Note: In order to use preset switching in conjunction with Keyboard Mode Split, you might need to adjust the transpositions for the Upper Manual (Trans UM), Lower Manual (Trans LM) and Pedal (Trans Ped).

Transposition (Octave Range)

You can individually transpose the Upper (Trans UM), Lower (Trans LM) and Pedal (Trans Ped) registers up/down one or two octaves, independent of the global transpose functions. Transposition is useful for setting the various registers to a particular range to match your needs. This facility is particularly important when using split mode.

Note: These manual-specific transpositions have no impact on the preset keys. Also see "Preset Keys and Morphing" on page 333.

MIDI Mode

This parameter allows you to define how the drawbar settings will respond to remote MIDI control change messages. Normally, you won't need to change anything here. If you own a MIDI drawbar organ, however, you'll probably want to use its hardware drawbars to control the EVB3. Most hardware drawbar organs utilize an independent control change number for every drawbar. Some models allow you to freely define these control change numbers. Lists of control change number assignments for the following MIDI Modes can be found in "MIDI Controller Assignments" on page 348.

If you set the MIDI Mode menu to RK, every drawbar responds to a specific MIDI control change number, commencing with CC #70. (Non-drawbar parameters may be set using control change messages up to CC #118).

If you choose HS in the MIDI Mode menu, all of the EVB3 drawbars will be controlled by just a few control change numbers—CC #80–82. Its values are intelligently mapped to all drawbars. The resolution of this technique is not particularly high (much like the original B3), but it works well. The Hammond-Suzuki XB-2 uses this controller assignment method, allowing you to remotely control the EVB3's drawbars. In addition to the drawbar support, you can use the following controls of the Hammond Suzuki XB-1, XK-2, and XK-3 to remotely control your EVB3:

- Leslie: Brake/On/Fast
- Vibrato and Chorus
- Percussion (2nd/3rd/soft/fast)

A lists of all control change number assignments can be found on page 351.

If you set MIDI Mode to NI, the Native Instruments B4 settings will be replicated, allowing the use of the Native Instruments B4D Drawbar controller.

If you set MIDI Mode to NE, the EVB3 will be able to respond to control change messages from Clavia's Nord Electro II keyboard.

If you set MIDI Mode to off, the EVB3 will not respond to MIDI drawbar messages.

Note: Before you commence a recording session with drawbar movements, performed on a hardware organ and the EVB3 set to MIDI Mode HS, you should disable MIDI Data Reduction in Logic Pro. The parameter is located in File > Project Settings > Recording. The function thins out the incoming data stream, which is hardly ever noticed with normal controller assignments. The controller mapping of the Hammond XB-2's drawbars, however, exploits the controllers to their limits. Therefore, MIDI Data Reduction can lead to incorrect values. Switch the parameter on after the drawbar recording session, in order to regain memory and processing resources.

The EVB3 Parameters

You can open and close the EVB 3's wooden lid by clicking the button underneath the Volume control.



Keep it open while reading this section of the manual, because it will cover every parameter in detail.



Drawbars

The principles of additive synthesis with sine drawbars is further explained in "Additive Synthesis With Drawbars" on page 355. You can intuitively pick up the basic principles by playing a little with the drawbars. The further down you drag the drawbars, the louder the selected sine choir(s) will be—the drawbars behave like reversed mixer faders. MIDI control of the drawbars is also reversed, when using a standard MIDI fader box.



Drawbars of the Upper and Lower manual, plus Pedal drawbars.

Pedal Drawbars

The EVB3 features two drawbars for the bass Pedals. The waveform of the bass is not a pure sine wave, but a mixed waveform, that realistically simulates the Hammond B3 bass. The two registers differ in pitch, and in the following ways:

- The Lower 16' register contains more octaves
- The 8' register has a more prominent fifth portion

Volume

The EVB3 offers several parameters that provide control over volume.

Relative Volumes—Upper/Lower/Pedal

In the parameters section, you can set the volume of the Lower manual, and the Pedal, relative to the Upper manual. The parameters are called Lower Volume and Pedal Volume.



Volume Control and Expression Pedal

The overall volume of the EVB3 is not only controlled by the software instrument channel's volume fader and control change #7, but also with the Volume control in the EVB3 graphic user interface.



Warning: The Volume must be lowered whenever crackling or other digital distortion occurs in the instrument channel. Volume levels over 0 dB can occur if you maximize all registers, play numerous notes, and make use of the Distortion effect.

You can control the volume in real time with an expression pedal. Extensive, and often rhythmic, use of the expression pedal forms part of the style of many organ players. The expression control also emulates the tonal changes of the B3 preamplifier. Bass and treble are not attenuated as strongly as the mids, much like a Hi-Fi amplifier that features a loudness control.

Normally, you would connect an expression pedal to the quarter inch Expression jack of your master keyboard. Your master keyboard should transmit MIDI control change #11 when the pedal is moved.

Tune

The simulated tone wheel generator can be tuned in cents (percentages of a semitone). 0 cis equal to A = 440 Hz.



Scanner Vibrato

The vibrato of the organ itself must not be confused with the Leslie effect, which is based on rotating speaker horns. The EVB3 simulates both.



The Scanner Vibrato is based on an analog delay line, consisting of several lowpass filters. The delay line is scanned by a multipole capacitor, which has a rotating pickup. It is a unique effect that cannot be simulated with simple LFOs.

Like the Hammond B3, the EVB3 features three types of vibrato with different intensities (V1, V2, V3). Vibrato speed is set with the Rate parameter. In the V1, V2, and V3 positions of the Type parameter, only the signal of delay line is heard.

The C1, C2, and C3 Chorus positions of the Type parameter mix the signal of the delay line with the original signal. Mixing a vibrato signal with an original, statically-pitched signal, results in a chorus effect. The Chorus parameter allows you to freely mix the dry signal with the vibrato signal. The Chorus parameter is only active if one of the chorus settings is engaged.

If the C0 setting is active, chorus and vibrato are disabled. Note that the treble portion of the organ is boosted slightly if any vibrato setting is used. This treble boost is maintained in the C0 setting.

The Upper and Lower buttons allow you to switch the scanner vibrato (and its associated treble boost) on or off, individually, for the Upper and Lower manual. As the B3 mixes the bass register (Pedal) with the Lower manual, the Pedal register is affected by the Lower manual's scanner vibrato settings. This side-effect reflects the technical limitations of the original B3.

Note: Check out the chorus and vibrato effects, and compare them with the sound of the rotor cabinet simulation! The organ's chorus sounds different to modern chorus effects (such as the Chorus effect of Logic Studio). Many organ players rarely use the Scanner Vibrato, preferring to work with a Leslie, in isolation. Others, like B3 virtuoso Brian Auger, prefer the integrated organ vibrato over the Leslie.

Percussion

Percussion is only available for the Upper manual—same as the original B3. The percussion of an electro-mechanical organ is polyphonic, but is only (re)triggered after all keys have been released. If you release all keys, new notes or chords will sound with percussion. If you play legato, or sustain other notes on the Upper manual, no percussion will be audible.

On the original B3, percussion is only available if the "B" preset key is selected (see "Preset Keys and Morphing" on page 333). If you want this restriction to be simulated, set the Perc parameter to Only B. You find this parameter in the Organ section. If you always want percussion to be available, select Always.



Activate percussion by clicking the On button in the Percussion section, to the top-right of the interface. The percussion is heard in the 4' register or in the 2 2/3' register, if the 2nd/3rd button is set. Set the percussion decay time with Time, and set its level with Vol. This improves on the B3, where Time and Vol could only be switched on/off, whereas these parameters can be varied.

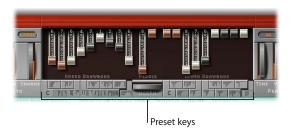
Note: Time has a maximum setting called Paradise. In this position, the percussion doesn't decay at all. The name is derived from a famous Jimmy Smith recording "Groovin' at Small's Paradise," where Jimmy used a B3 that had a defect in the percussion trigger. The cool thing about this technical flaw is that one harmonic sounds without Chorus-Vibrato, while the drawbar harmonics feature Chorus-Vibrato. While this is very specialized, it is a nice inclusion, especially for jazz-oriented players.

The EVB3 percussion register can be played with velocity sensitivity, unlike the B3. Set the percussion velocity sensitivity with Vel.

If you engage percussion on a B3, the volume of the normal, non-percussive registers is reduced slightly. The Up Level parameter simulates this behavior, allowing you to define the volume of the Upper manual, with percussion engaged.

Preset Keys and Morphing

The Hammond B3 is equipped with 12 switches, located below the lowest octave of both keyboard manuals. These are the *preset keys*, and are laid out as an inverted keyboard octave (black keys, white sharps). They are used to recall drawbar registrations. These presets could only be altered with a screwdriver on the original B3.



MIDI notes 24 to 35, the octave below the lowest octave of a (non-transposed) 5-octave keyboard, are used as the preset keys. See "Keyboard Ranges of the Upper and Lower Manual" on page 325.

You can click these preset keys directly in the graphic user interface of the EVB3. They are located to the left (Upper) and right (Lower) of the *Morph* wheel. The current drawbar registrations are indicated by small vertical lines on each key.

The presets only relate to the registration of a single manual. The presets do not store vibrato or other parameter settings. The preset keys work in real-time, thus overall settings (including effects) can be stored and recalled via the *Settings* menu.

Note: On keys C# to A#, the percussion only works in *Percussion Mode Always* (see "Percussion" on page 332).

Disabling MIDI Preset Switching

You can disable the switching of presets with MIDI notes 24 to 35, thus avoiding any problems that may arise from transpositions. To do so, set the MIDI to Presetkey parameter to "off."

Cancel Key, Registering While Playing

The lowest preset key (C) is the cancel key. If you depress it, all drawbars are moved to their minimum setting. The other 11 keys, from C# to B, recall registrations. You can edit recalled presets immediately. The preset memorizes these alterations instantaneously, with no further action required. This means that if you recall a new preset, the former preset memorizes the drawbar settings at the time the new preset was recalled.

If you hold the cancel key (C) on your master keyboard with the small finger of your left hand, and sustain a chord with your right hand, you can trigger the chord with different registrations, by clicking the preset keys with the other fingers of your left hand. This results in an organ-specific gate-type effect, which wouldn't be possible with the right hand alone.

Morphing

You can switch between the presets of the Upper manual with any MIDI controller—such as the modulation wheel. Choose the desired MIDI controller with the MIDI CC parameter.

The step or linear Mode parameter options determine whether an abrupt switch, or a seamless morph (cross-fade), occurs between presets.

Note: Switching the registers with this morphing function is only available for the Upper manual.

Range

After deciding on a controller to use for switching or morphing the Upper manual registers, you can determine the number of preset keys that are affected.

The morphing (or switching) always begins with the top preset key, the B. Range defines the end preset key. If Range = A#, you will switch or morph between two presets. If Range = G#, you will switch or morph between four presets (B, A#, A, and G).

Save To

In Linear Mode (morphing instead of switching), the seamless crossfades result in a variety of new drawbar registrations you might want to save. Before saving, you may wish to alter some drawbar positions manually. Click the words *Save To*, and select the destination preset key. You do not need to do anything else, in order to save the setting to another preset key.

Note: As soon as you morph, the Morph text turns orange, indicating changes—in other words, you're performing a morph. The morphing result(s) can be further modified with the drawbars, but are lost if you don't save. The Morph text will start to flash if any drawbar modifications are made, indicating changes to the Morph values.

Organ Parameters

The Organ parameters adjust the overall behavior of your EVB3.



Organ parameters

The Lower Volume and Pedal Volume parameters are discussed in "Relative Volumes—Upper/Lower/Pedal" on page 329. The Perc parameter is discussed in "Percussion" on page 332.

Max Wheels

Calculating (emulating) all tone wheel generators consumes considerable CPU processing power. A reduction of this parameter value reduces the EVB3's hunger for processing resources.

Note: This will diminish some overtones, so you should not reduce this value if you're after an ultra-realistic simulation.

Tonal Balance

Changes the mix relationship of the higher/lower sounding tone wheels. Positive values result in a lighter and brighter sound. Experiment with different Tonal Balance and Equalizer settings. See the "Equalizer" on page 341, for further information.

Shape

While the Hammond's tone generators produce pure sine waves (despite technical artifacts), other organs deliver distorted waveforms. You can produce sounds resembling those of Farfisa, Solina, or Yamaha organs with the *Shape* parameters.

You can subtly alter the waveform of the sounds emanating from the tonewheel generator with the Shape parameter. Moving the parameter to the right will make the tone brighter (and louder), and moving it to left will make it duller (and softer).

Note: The Shape parameter is placed after the filters which follow the sine generators.

Bass Filter

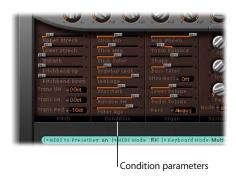
The sound of the Pedal drawbars often appears to be somewhat brilliant, within the overall musical context. To circumvent this issue, and to suppress the treble of the bass register, please make use of the Bass Filter. When active, you will only hear a solid bass organ fundamental in the bass register.

Ultra Bass

If you switch on Ultra Bass, another low octave will be added to the playable range of the Upper and Lower manuals. These additional low octaves, and the ability to independently transpose both manuals (see "Transposition (Octave Range)" on page 326), are not available on the original B3.

Condition Parameters

Technical limitations of electro-mechanical drawbar organs with tonewheels cause some strange tonal artifacts, such as crosstalk. These quirks form an integral part of the B3's charm. You can adjust the following parameters to define the age of your EVB3.



Note: Read more about the Click parameters in "Click Parameters" on page 337.

Drawbar Leak

Even if all drawbars are at their minimum position, the tonewheel generators of the B3 aren't completely quiet. This is due to leakage of the tone wheels, cross-talking with the output. Drawbar Leak allows you to set the minimum output level of the drawbars, when set to minimum. At a value of 0, you can completely eliminate this leakage. At the maximum setting, drawbar leakage is most definitely audible.

Leakage

Leakage adds a sound resulting from the crosstalk of all tone wheels—including those of the notes that you don't play. This provides breath to the organ sound.

Crosstalk

The Hammond's tone wheels are divided into compartments of four—with the same key, but in different octaves. There are two tonewheels, four octaves apart, on each rotating shaft. The signal of the Lower wheel contains a small amount of signal, induced by the higher wheel, and vice versa. This crosstalk can be adjusted with the Crosstalk slider. Note that crosstalk is only audible on certain wheels, avoiding rumble when chords are played.

Random FM

If the tonewheel generator is clean, all frequencies are straight—meaning the frequencies are even/in tune. The three-fold decoupling of the tonewheels—via springs, flexible couplings and flywheels—is very effective, but can't compensate for irregularities that come with dirt and grease in the driving gear. This gradual build-up of grime in the mechanism makes the revolution of the tonewheel assembly irregular on its axis. This is transmitted to the tonewheels. The Random FM slider allows you to simulate this effect.

Note: The effect only becomes audible in the higher frequency ranges.

Filter Age

The high frequency output signals of the B3's tone wheel generators are filtered by bandpass filters. The center frequency of these filters varies as the capacitors age. Filter Age allows you to alter the center frequencies of the filters. This colors the sound of the jitter applied by Random FM and the background noise resulting from leakage. (See "Leakage" on page 336 and "Random FM" on page 337.) This parameter also influences the intonation of the organ, if you use the pitch bender.

Click Parameters

The key contacts of electro-mechanical tonewheel organs tend to saw a little on the busbar, introducing a short click sound. If any corrosion occurs to the key contacts or busbar, this will increase the length, and level, of this click. This aspect of the B3's design causes irregular scratching noises (commonly referred to as keyclick), when striking and releasing keys. Hammond fans like these clicking noises, as they introduce a transient, percussive quality to the note. The EVB3 allows you to adjust the volume and sound of the key click. The click sound is altered randomly, and independently, from the click on and click off (release) volume settings.



Click On/Click Off

These two knobs independently control the click volume for the beginning (Click On), and release, of the note (Click Off). The click off is quieter, even if both controls are set to the same position.

Click Min/Click Max

Not only are the tone color and volume of clicks altered randomly, but also their duration. Click duration can vary between a short tick and a longer scratch. Minimum duration is defined by Click Min, and maximum duration with Click Max. The duration is displayed in milliseconds, as you move the sliders.

Note: Even if both parameters have identical values, there is still a random variation in sound. This variation makes some clicks seem shorter than the value set with *Click Min*.

Click Color

Set the tone color of the click here. Despite its random variation, you can define its treble portion globally.

Pitch Parameters



Compared to the original B3, the EVB3 offers several parameters to change its pitch behavior.

Note: The Trans UM, Trans LM and Trans Ped parameters are explained in "Transposition (Octave Range)" on page 326.

Stretch

The EVB3 is tuned to an equal-tempered scale. As a deviation from this standard tuning, you can stretch the tuning in the bass and treble ranges, much like acoustic pianos (especially upright pianos). If you select a value higher than 0 for Upper Stretch, the pitch of the higher notes will be raised. If you select a value higher than 0 for Lower Stretch, the pitch of the Lower notes will be lowered.

Note: The tones of clavinets, harpsichords, and pianos have inharmonicities in their harmonic structure. The frequencies of these overtones (harmonics) are not exact, whole-number multiples of the base frequency. They are only approximate and are, in fact, a little higher. This means that the overtones of lower (tuned) notes are more closely related to the main frequencies of the upper notes. Due to the lack of strings, this inharmonic relationship is not true of organs. The stretch feature was included for situations where you may wish to use the EVB3 in an arrangement alongside an acoustic or digital piano (EVP88 and EVD6 Clavinet).

Lower Stretch controls the amount of deviation from the equal-tempered scale in the bass frequencies. The higher the value, the further down the low notes are tuned. At a setting of 0, the EVB3 is tuned to an equal-tempered scale, with each octave below exactly halving the frequency. Upper Stretch controls the amount of deviation from the equal-tempered scale in the treble end of the sound. The higher the value, the further up the high notes are tuned. At a setting of 0, the EVB3 is tuned to an equal-tempered scale, with each octave up exactly doubling the frequency.

Warmth

Warmth controls the amount of random deviation from an equal-tempered scale. High values add life to organ sounds, but tend to sound a little out of tune.

When applying Warmth and Stretch, you should consider that these parameters may result in a detuned sound, which is similar to a heavy chorus effect. Set Warmth to 0 if you're after a pure sound.

Pitch Bender, Brake Effect

The Hammond organ has no pitch bender. As such, use of the pitch bender is not suitable for realistic simulations, but it does provide a number of creative options.

Pitch up/down bender sensitivity can be set independently, in semitone steps, with the Pitchbend Up and Pitchbend Down parameters. The maximum sensitivity for upward bends is one octave.

You can set Pitch Bend Down to Brake, which gradually slows the movement of the tone wheels down to a total stop, at the pitch bender's minimum position.

Note: The Brake setting recreates an effect that is audible at the end of "Knife Edge" by Emerson, Lake and Palmer. Keith Emerson's virtuoso Hammond work was recorded on a reel-to-reel tape recorder. At the end of the song, you can hear the tape recorder being gently slowed to a total stop.

Sustain

Synthesizer players call the time the note takes to fade out after the release of the key the release time. The EVB3 allows you to control this parameter as well; it's called Sustain in the organ lexicon. The three controls allow for individual settings in the Upper (Up), Lower (Low) and Pedal (Ped) registers.



If you select Smart Mode, playing new notes will cut the sustain (release) phase of released notes. Normal Mode allows polyphonic sustain phases—all released notes will continue to sustain. Smart Mode allows long sustain times, even in the bass register, which would normally cause rumbling dissonances.

Effects

The EVB3 features a three-band equalizer, a reverberation effect, a pedal-controllable wah effect, and a distortion processor that simulates the sound of an overdriven tube amplifier. Finally, the signal can be processed by the rotor effect.

Effect Chain

The EVB3's signal flow is as follows: the organ's signal runs through the equalizer, wah and distortion effects. You can choose between four different signal flow routings for the equalizer, wah wah, and distortion effects in the Effect Chain menu. This treated signal is then fed into Reverberation and finally passed to the Rotor effect.

A classic B3 patch would be: an EQed organ, plugged into a wah wah pedal, amplified by an overdriven Leslie. Select EQ-Wah-Dist.

The sound of the overdrive changes if the input signal is being filtered—be it by the EQ, or the wah. If you patch the EQ before the overdrive, the sound of the overdrive becomes much more flexible. The output signal of the distortion effect always contains high frequency content. If you want to suppress these frequencies, the wah wah must be the final effect in the chain—EO-Dist-Wah.

If you wish to create a screaming sound (achieved by distorting the wah wah output), you can minimize any harshness by choosing the Wah-Dist-EQ routing.

You can suppress the brutal overtones of extreme distortions with two filters: Select Dist-EQ-Wah.

Effect Bypass

The Distortion, Wah, and EQ effects can be bypassed separately for the Pedal register. Set Effect Bypass to Pedal to do so. This avoids the entire bass portion of your organ being suppressed by the wah wah. It also avoids undesirable intermodulation artifacts, when utilizing the overdrive effect.

If you choose None, the entire output of the organ is processed, as if you had plugged the B3's mono output into a Leslie cabinet.

Equalizer



- EQ Low: Adjust the level of the low frequency range.
- EQ Mid: Adjusts the level of the mid frequency range.
- EQ High: Adjusts the level of the high frequency range.
- EQ Level: Adjust the overall EQ level.

Reverb

The EVB3 offers six reverb algorithms: Box, Small, Medium, Large, Big, and Spring.



Reverb level is defined by the Reverb parameter. A Reverb =0 value conserves processing resources.

You can also choose Bypass in the Reverb Mode menu, if you want to disable the reverb without changing its level.

The reverb is always patched *after* the EQ, wah wah and distortion effects, but *before* the rotor effect. This means that the reverb always sounds as if it is played back through the rotor speaker. To hear the reverb after the rotor, switch off the organ's reverb, and use an aux send to apply reverb to the instrument channel.

Wah

The name Wah Wah comes from the sound it produces. It has been a popular effect with electric guitarists since the days of Jimi Hendrix. The pedal controls the cutoff frequency of a bandpass, lowpass, or—less commonly—highpass filter. The wah wah pedal has also been used extensively with the Hammond organ.



MIDI Pedal Control

For the most dynamic and musical performance of the wah wah effect, attach an Expression Pedal to your MIDI master keyboard. Your master keyboard should transmit MIDI control change #11 (Expression real-time volume), if you attach an Expression Pedal to the Expression jack and move it. This would normally be used to control the volume while playing.

If you program an EVB3 setting, set the *Expression* parameter to *0*, and then define a wah wah effect (with controller 11 controlling the wah wah's cutoff frequency), you can control the wah wah with the pedal—without having to program anything on the master keyboard.

Read more on this in the "MIDI Setup" on page 324. You should also consult the users manual of your keyboard.

Wah Wah Control With Other MIDI-Controllers or Aftertouch

You can use any MIDI control change message to control the wah wah effect. You can select any controller number and Channel Aftertouch (*Touch*) in the *CC* field.

Mode

Mode allows you to enable/disable the wah wah effect. If you select Mode off, the effect is disabled. There are six different filter types available:

- ResoLP (Resonating Lowpass Filter): In this mode, the wah wah will work as a resonance-capable lowpass filter. At the minimum pedal position, only low frequencies can pass.
- ResoHP (Resonating Highpass Filter): In this mode, the wah wah will work as a resonance-capable highpass filter. At the maximum pedal position, only high frequencies can pass.

- Peak: In this mode, the wah wah will work as a peak (bell) filter. Frequencies inside
 the center frequency, which is controlled by the selected MIDI controller, will be
 emphasized.
- CryB: This setting mimics the sound of the popular Cry Baby wah.
- *Morley 1:* This setting mimics the sound of a popular wah pedal, manufactured by Morley. It features a slight peak characteristic.
- Morley 2: This setting mimics the sound of the Morley distortion wah pedal. It has a constant O.

Range

Controls the sensitivity of the wah to controller movements. If you only intend to make slight alterations to the cutoff frequency, choose a small value.

Bite

The resonance parameter of the wah filter. You'll know the parameter's meaning, if you're a synthesizer player. Anyway, check it out: The cutoff frequency is boosted. High values make the wah sound more aggressive.

Distortion

The distortion effect simulates an overdriven two-stage tube amplifier. Its primary role is to simulate the Leslie amplifier, or whatever amp might be used to feed the Leslie speaker cabinet.



Type offers the choice between three different tube amp types: Growl, Bity, and Nasty. Growl simulates a two-stage tube amplifier, resembling the Leslie 122 Model, the classic partner for the Hammond B3 organ. Bity is reminiscent of a bluesy guitar amp. Nasty delivers hard distortions, and is well suited for very aggressive sounds.

The Tone control only affects the distorted portion of the sound, while the dry signal portion remains unaffected. This allows for very warm overdriven sounds that won't become scratchy if you try to get more treble out of the instrument.

Drive controls the amount of overdrive distortion. The output level is automatically compensated for, so there's no need for another master volume control adjustment facility.

Rotor Cabinet

The Hammond story can't be fully told without a chapter on the rotor sound cabinets, manufactured by Leslie. In fact, playing the B3 organ without a rotor cabinet is viewed as a special effect these days. The EVB3's rotor cabinet section simulates not only the speaker cabinet itself, but also the microphones which pick up the sound.



Rotor Speed Buttons

These buttons switch the rotor speed as follows:

Chorale: Slow movementTremolo: Fast movement.Brake: Stops the rotor.

Cabinet Menu

You can choose between the following settings in the Cabinet menu:

- Off: Choose to switch off the rotor. There's an alternative to switching the rotor effect off: in the Brake mode, the speakers don't rotate, but are still picked up by the simulated microphones, in a random position (see paragraph "Rotor Speed" below). You can also directly disable the Leslie simulation with the On/Off button beside the Speed Control menu.
- *Wood:* Mimics a Leslie with a wooden enclosure, and sounds like the Leslie 122 or 147 models.
- Proline: Mimics a Leslie with a more open enclosure, similar to a Leslie 760 model.
- *Single*: Simulates the sound of a Leslie with a single, full-range, rotor. The sound resembles the Leslie 825 model.
- *Split:* The bass rotor's signal is routed more to the left side, and the treble rotor's signal is routed more to the right side.
- Wood & Horn IR: This setting uses an impulse response of a Leslie with a wooden enclosure.
- *Proline & Horn IR*: This setting uses an impulse response of a Leslie with a more open enclosure.
- Split & Horn IR: This setting uses an impulse response of a Leslie with the bass rotor signal routed more to the left side, and the treble rotor signal routed more to the right side.

Speed Control

The Speed Control menu allows you to define controllers to switch the Rotor Speed buttons remotely.

The ModWheel setting uses the modulation wheel to switch between all three speed settings. Brake is selected around the modulation wheel's center position, while Choral is selected in the lower, and Tremolo in the upper third of the modulation wheel's travel.

All other entries in the Speed Control menu switch between Tremolo and the speed set with the Rotor Speed buttons, that means either between Chorale and Tremolo, or between Brake and Tremolo. If Rotor Speed is set to Tremolo, you will switch between Tremolo and Choral. The difference between the entries is how the switch action takes place:

- ModWhl Toggle: Switches as soon as the modulation wheel exceeds its center value
 on its way from its low position to its high position. If the modulation wheel passes
 its center value on its way from its high to its low position, there will be no toggle
 This has been implemented for Roland keyboards with combined Pitch Bend- and
 Modulation levers.
- ModWhl Temp: Switches as soon as the modulation wheel passes its center value, regardless if you move the mod wheel from high to low or from low to high. This has been implemented for Roland keyboards with combined Pitch Bend- and Modulation levers.
- *Touch*: Switches as soon as you press aftertouch. If you release aftertouch, there will be no toggle.
- *Touch Temp:* Switches as soon as you press Aftertouch. If you release aftertouch, there will be another toggle.
- SusPdl Toggle: Toggles as soon as you press the sustain pedal. If you release the sustain pedal, there will be no toggle.
- SusPdl Temp: Toggles as soon as you press the sustain pedal. If you release the sustain pedal, there will be another toggle.
- CC #18 and CC #19 Touch: Toggles as soon as you press controller 18 or 19. If you release the controller, there will be no toggle.
- CC #18 and CC #1 Temp: Toggles as soon as you press controller 18 or 19. If you release controller 18 or 19, there will be another toggle.

Rotor Fast Rate

Rotor Fast Rate defines the maximum possible rotor speed (Tremolo). While moving the slider, the Tremolo rotation speed is displayed in Hertz.

Acc/Dec Scale

The Leslie motors need to physically accelerate and decelerate the speaker horns in the cabinets, and their power to do so is limited. Acc/Dec Scale determines the speed at which the motors can accelerate the rotors (time it takes to get the rotors up to a determined speed), and the length of time it takes for them to slow down. If the slider is set to its far left position, you can switch to the preset speed immediately. If you drag the slider to the right, speed changes take more time to occur. In the default position of 1 the behavior is Leslie-like.

Horn Deflector

If you look inside a Leslie cabinet, you'll see a double horn, with a deflector at the horn mouth. This deflector makes the Leslie sound. Some people, however, removed it to alter the Leslie sound, as removal of the deflector increases amplitude modulation, and decreases frequency modulation. This parameter allows you to switch the deflectors on and off, without needing to order spare deflectors.

Mic Angle and Mic Distance

The Mic Angle slider defines the stereo image, by changing the angle of the simulated microphones. An angle of 0° results in a mono sound, while an angle of 180° causes phase cancellations. Experienced sound engineers tend to avoid wide spreads.

Mic Distance makes the sound darker, and less defined, when set to higher values. This is typical of microphones, when positioned further from the sound source.

Motor Control

In the Motor Control menu, you can set different speeds for the bass and treble rotors:

- Normal: Both rotors use the speed set by the rotor speed buttons.
- Inv (inverse mode): In Tremolo mode, the bass compartment rotates with fast speed, while the horn compartment rotates with slow speed, and vice versa in Chorale mode. In Brake mode, both rotors will stop.
- 910: The 910, or Memphis mode, stops the bass drum rotation at slow speed, while the speed of the horn compartment can be switched. This may be desirable, if you're after a solid bass sound, but still want treble movement.
- *Sync*: The acceleration and deceleration of the horn and bass drum are about the same. This sounds as if the two were locked, but is only clearly audible during acceleration or deceleration.

Note: If you choose Single Cabinet in the Cabinet Type menu, the Motor Ctrl setting is irrelevant, because there are no separate bass and treble rotors in a Single Cabinet.

Extended Parameters

A number of additional parameters are accessible in the extended parameter view of the EVB3 window.

Dry Level

Adjusts the level of the dry signal, which can also be useful if the "Switches to dry sound" option is selected in the Brake menu, found below.

Brake Menu

The Brake menu offers two options that allow you to modify the EVB3's Brake mode:

- *Stops rotor:* In this mode, the movement of the rotor is gradually slowed down to a total stop.
- Switches to dry sound: In this mode, the rotor cabinet is bypassed when stopped,
 with a delay time of one second. This is useful if using the modulation wheel to
 switch between the Tremolo (fast rotor speed) and Chorale (slow movement) modes.
 If you then switch to Brake mode, the rotors are slowed down during the transition
 to the dry sound.

Horn Brake and Drum Brake Pos

The Horn Brake Pos and Drum Brak Pos sliders allow you to set an exact stop position for the Leslie horn or bass rotator, respectively. This is something that the original Leslie could not do, sometimes resulting in a horn that was aimed at the back of the cabinet when it came to a halt, which has a less than desirable sound.

Reverb Menu

The Reverb menu allows you to patch the reverb effect before (Pre) or after (Post) the rotor effect.

Velo to Click

The Velo to Click slider allows you to set the velocity sensitivity of the *Click* parameters (see "Click Parameters" on page 368).

Bass Menu

The Bass menu allows you to imitate the non-foldback bass behavior of the first Hammond organ ever made, the model A. This model had no foldback for the 16' drawbar in the lowest octave, with the bottom twelve tone generator outputs available on the first drawbar of the manuals' bottom octave. On other console organs, the lowest twelve tone generator outputs are only available on the pedals. To simulate the behavior of the Hammond Model A, choose the "All the way down" option in the Bass menu. Without foldback, you'll find that the sound is more strident, and similar to the pedal sound, especially if you're not using a real Leslie cabinet when playing the EVB3.

When using a two drawbar hardware controller, the "Drawbar affects" menu offers an additional mode that allows Hammond-like switching between two registrations. If you use the default setting ("Drawbar affects current preset key"), the drawbars will always change the registration of the currently active preset registration key. This works differently in a real" Hammond organ, where the drawbars only affect the Bb (upper manual) and B (lower manual) preset registrations. This allows the player to prepare a new registration with the drawbars while playing, and then switch to the new registration as desired.

To simulate this behavior with the EVB3:

- 1 Choose the "Only B & Bb key" option in the Drawbar mode menu. This option enables the upper manual drawbars to change the registration of the Bb preset key, and the drawbars of the lower manual affect the B preset key.
- 2 Change the drawbars of the Bb preset key as desired. You can freely play the keyboard while doing so, without changing the currently chosen registration.
- 3 Switch to the prepared registration with the Bb preset key.

MIDI Controller Assignments

MIDI controller assignments are only updated if you load the default setting or a setting that was saved with a project. All parameters that allow you to select a MIDI controller feature a Learn entry. If this option is selected, the parameter will automatically be assigned to the first appropriate incoming MIDI data message. The Learn mode features a 20 second time-out facility: If the EVB3 does not receive a MIDI message within 20 seconds, the parameter will revert to its original MIDI controller assignment.

Note: As the new entry is added at the top of the list, existing automation data needs to be increased by one.

In most circumstances, the track automation of Logic Pro is your best option for the recording of manual movements in the plug-in window.

MIDI automation is the better option when you wish to make use of your hardware MIDI drawbar organ as a remote control for the EVB3. The following tables outline the MIDI controller assignments of the EVB3. The assignments are optimized for use with several popular MIDI drawbar organ models. In order to match the default settings of all popular organs, some parameters are assigned twice. You need not concern yourself about this.

MIDI Mode: RK

This table describes the MIDI Control Change Message number assignment when MIDI Mode is set to RK. This is the correct setting if you use a Roland VK series or Korg CX-3 drawbar organ as a remote controller for the EVB3.

Controller Number	MIDI Mode RK: Parameter Name
70	Drawbar 16'
71	Drawbar 5 1/3'
72	Drawbar 8'
73	Drawbar 4'
74	Drawbar 2 2/3'
75	Drawbar 2'
76	Drawbar 1 3/5'
77	Drawbar 1 1/3'
78	Drawbar 1'
Rotor Cabinet	
80, 92	Chorale/Brake/Tremolo
81	Chorale/Brake
Reverb	
82	Reverb Level
Vibrato	
85	Upper Vibrato on/off
86	Lower Vibrato on/off
87	Chorus Vibrato Type
Percussion	
94	on/off
95	2nd/3rd
102	Percussion Volume
103	Percussion Time
Equalizer	
104	EQ Low
105	EQ Mid
106	EQ Hi
107	EQ Level
Wah	
108	Wah Mode
109	Wah Bite

Controller Number	MIDI Mode RK: Parameter Name
Distortion	
110	Distortion Type
111	Distortion Drive
112	Distortion Tone
Click Levels	
113	Click On Level
114	Click Off Level
Balance	
115	Main Volume
116	Lower Volume
117	Pedal Volume
Rotor Fast Rate	
118	Rotor Fast Rate

MIDI Mode: HS

This tables describes the MIDI controller assignments when MIDI Mode is set to HS. This setting matches the controller mapping of Hammond XB-series organs. Switch MIDI Data Reduction off while recording data from an XB-Series organ (File > Project Settings > Recording).

Controller Number	MIDI Mode HS: Parameter Name
80	All Upper Drawbars
81	All Lower Drawbars
82	Pedal Drawbars, Scanner Vibrato, Bass Filter
Rotor Cabinet	
Leslie On	Rotor Cabinet on/off
Leslie Fast	Chorale/Brake
Leslie Brake	Controls Brake functionality of Rotor Cabinet
Vibrato	
Vibrato On	Upper Vibrato on/off (on XK-3 only)
Vibrato Mode	Vibrato Type (V1–C3, XK-3 only)
87	Chorus Vibrato Type
Drive	Distortion Drive
Reverb Level	Reverb Level
Perc 2nd and Perc 3rd	Percussion Harmonic, 3rd harmonic has priority over 2nd. Translation from XK buttons to EVB3 is as follows: • 2nd off, 3rd off → EVB3: Percussion off • 2nd on, 3rd off → EVB3: 2nd Harmonic • 2nd off, 3rd on → EVB3: 3rd Harmonic • 2nd on, 3rd on → EVB3: 3rd Harmonic
Perc Fast	Selects a preset decay time for fast or slow decay.
Perc Soft	Selects a preset level for either soft or normal percussion.
Vibrato Mode	Selects either Vibrato Off, V1/V2/V3 or C1/C2/C3 (XK-2 only)
Vibrato VC	Switches between Vibrato or Chorus Vibrato (XK-2 only)

MIDI Mode: NI

This table describes the MIDI controller assignments when MIDI Mode is set to NI. This setting matches the controller mapping of the Native Instruments B4D controller.

Controller Number	MIDI Mode NI: Parameter Name
12	Upper Drawbar 16'
13	Upper Drawbar 5 1/3'
14	Upper Drawbar 8'
15	Upper Drawbar 4'
16	Upper Drawbar 2 2/3'
17	Upper Drawbar 2'
18	Upper Drawbar 1 3/5'
19	Upper Drawbar 1 1/3'
20	Upper Drawbar 1'
21	Lower Drawbar 16'
22	Lower Drawbar 5 1/3'
23	Lower Drawbar 8'
24	Lower Drawbar 4'
25	Lower Drawbar 2 2/3'
26	Lower Drawbar 2'
27	Lower Drawbar 1 3/5'
28	Lower Drawbar 1 1/3'
29	Lower Drawbar 1'
Vibrato	
31	Upper Vibrato on/off
30	Lower Vibrato on/off
Brightness	Vibrato
Attack Time	Chorus Intensity
Percussion	
Sostenuto	Percussion on/off
Release Time	Percussion Harmonic (2nd/3rd)
Sound Variation	Percussion Volume
Harmonic Content	Percussion Time
Equalizer	
90	EQ Low
70	EQ Mid
5	EQ High

Controller Number	MIDI Mode NI: Parameter Name
Distortion/Click	
76	Distortion Drive
78	Distortion Tone
75	Click On Level
Leslie	
Pan MSB	Microphone Angle
3	Microphone Distance
GP 8	Leslie Accelerate/Decelerate
GP 7	Leslie Fast
ModWheel MSB	Leslie Speed
68	Controls Brake functionality: If Value = 0.0, switch Leslie to Brake. All other values switch Leslie to previous speed.

MIDI Mode: NE

This table describes the MIDI Control Change Message number assignment when MIDI Mode is set to NE. This setting matches the controller mapping of the Clavia Nord Electro 2.

Controller Number	MIDI Mode NE: Parameter Name
16	Upper Drawbar 16'
17	Upper Drawbar 5 1/3'
18	Upper Drawbar 8'
19	Upper Drawbar 4'
20	Upper Drawbar 2 2/3'
21	Upper Drawbar 2'
22	Upper Drawbar 1 3/5'
23	Upper Drawbar 1 1/3'
24	Upper Drawbar 1'
70	Lower Drawbar 16'
71	Lower Drawbar 5 1/3'
72	Lower Drawbar 8'
73	Lower Drawbar 4'
74	Lower Drawbar 2 2/3'
75	Lower Drawbar 2'
76	Lower Drawbar 1 3/5'
77	Lower Drawbar 1 1/3'
78	Lower Drawbar 1'

Controller Number	MIDI Mode NE: Parameter Name
Chorus/Vibrato	
85	Upper Vibrato on/off
86	Lower Vibrato on/off
84	Vibrato mode (selection goes from V1 to C3, C0 is excluded)
Percussion	
87	Percussion on/off
88	Percussion Volume (soft/normal) and Time (short/long)
95	Percussion Harmonic (2nd/3rd)
Equalizer	
113	EQ High
114	EQ Low
Distortion/Click	
111	Distortion Drive
Leslie	
GP 6	On/off
GP 7	Leslie Speed
GP 8	Controls Brake functionality

Additive Synthesis With Drawbars

The Hammond B3 is *the* classic drawbar organ. As with an acoustic pipe organ, the registers (drawbars, or stops on a pipe organ) can be pulled out, in order to engage them. But in contrast to a pipe organ, the B3 allows seamless mixing of any drawbar registers. The more you drag the drawbars down, the louder they will become.

Despite characteristics such as key clicks, intonation undulations, distortions, and crosstalk (all of which the EVB3 emulates), playing a single note, with a single register, results in a pure sine tone. Mixing harmonic sine tones results in more complex spectra; this is known as additive synthesis. Organs—even acoustic pipe organs—can be regarded as additive synthesizers. There are, however, several limitations that need to be considered before viewing the instrument in this way. These limitations, on the other hand, constitute the character of any real musical instrument, loaded with charm.

The naming of the drawbars derives from the length of organ pipes, measured in feet ("). This naming convention is still used with electronic musical instruments. Halving the length of a pipe doubles its frequency. Doubling the frequency means nothing other than: one octave up.

The lowest register, 16' (far left, brown drawbar), and the higher octaves 8', 4', 2', and 1' (white drawbars) can be freely mixed, in any combination. 16' is commonly described as the sub-octave. With the sub-octave regarded as the fundamental, the octave above 8' is the second partial, 4' the fourth, 2' the eighth and 1' the sixteenth partial.

With the $5 \, 1/3'$ register—the second brown drawbar—you can add the third partial. This is the fifth above the 8'. Basically, the drawbars are arranged by pitch, with one exception. The second drawbar ($5 \, 1/3'$) sounds a fifth higher than the third drawbar. See the "Residual Effect" on page 356, for an explanation.

22/3' gives the sixth, 1 3/5' the tenth and 1 1/3' the twelfth partial. So the electromechanical tone-wheel organ gives you the partials 1 (16'), 2 (8'), 3 (5 1/3'), 4 (4'), 6 (2 2/3'), 8 (2'), 10 (1 3/5'), 12 (1 1/3') and 16 (1'). As you can see, the harmonic spectrum is nowhere near complete. That's the reason why overdrive distortion effects are so popular with electromechanical tone-wheel organs—they enrich the harmonic spectra by generating more partials.

Note: The term partial is basically the same as harmonic, but they are counted in a slightly different way. The fundamental is considered the first partial. Its octave, twice the frequency, is the second partial, but is known as the first harmonic. The fifth partial oscillates at five times the frequency of the fundamental. The fifth partial is known as the fourth harmonic, because with harmonics, the fundamental is not counted (which makes the term harmonic less practical to use).

Note: $2 \frac{2}{3}$ is the fifth over 4'. $1 \frac{3}{5}$, is the major third over 2'. $1 \frac{1}{3}$ is the fifth over 2'. In the bass range, this can lead to inharmonic tones, especially when playing bass lines in a minor key. This is because mixing 2', $1 \frac{3}{5}$ and $1 \frac{1}{3}$ results in a major chord.

Residual Effect

The residual effect is a psychoacoustic phenomenon. Human beings can perceive the pitch of a note, even when the fundamental is completely missing. If human beings didn't hear this way, it would make listening to music with a kitchen radio impossible. Its speaker will never play back the fundamental of the bass line, as this frequency is far below the range that the speaker can transmit. If you pull out all registers of the drawbar organ, except for the fundamental—16', you'll still perceive the same pitch. The sound becomes thinner, with less bass and less warmth, but the pitch remains the same.

Setting drawbar registrations often involves this psychoacoustic phenomenon. In the lower octaves, mixing the 8' and 5 1/3' sine drawbars creates the illusion of a 16' sound, although the frequency is missing. Old pipe organs also make use of the residual effect, by combining two smaller pipes, eliminating the need for long, heavy, air-hungry, and expensive giant pipes. This tradition is continued in modern organs, and is the reason for arranging the 5 1/3' under 8: The 5 1/3' tends to create the illusion of a pitch that is one octave lower than 8'.

A Short Hammond Organ Story

Three inventions inspired Laurens Hammond (1895–1973), a manufacturer of electric clocks, to construct and market a compact electro-mechanical organ with tone wheel sound generation. The Telharmonium by Thaddeus Cahill was the musical inspiration, Henry Ford's mass production methods, and the domestic synchron clock motor were the other factors.

The Telharmonium was the first musical instrument that made use of electromechanical sound generation techniques. In the year 1900, its man-sized tone wheel generators filled a two-story building in New York. For a short period around this time, subscribers could order Telharmonium music over the New York telephone network (the streaming audio system of the time). The only amplification tool was the telephone's mechanical diaphragm, as a proper tube amplifier and acceptable speakers had not yet been invented. The Telharmonium was a commercial flop but its historical status as the predecessor of modern electronic musical instruments is undeniable. The Telharmonium also introduced the principles of electronic additive synthesis (see "Additive Synthesis With Drawbars" on page 355).

Laurens Hammond began producing organs in 1935, in Chicago, Illinois, making use of the same sound generation method. The differences were; much smaller tone generators, and fewer registers. The patent for his model A organ dates from 1934.

Hammond also holds the patent for the electro-mechanical spring reverb, still found in countless guitar amplifiers today!

The Hammond B3 was manufactured between 1955 and 1974. It is the Hammond model preferred by jazz and rock organ players such as: Fats Waller, Wild Bill Davis, Brother Jack McDuff, Jimmy Smith, Keith Emerson, Jon Lord, Brian Auger, Steve Winwood, Joey DeFrancesco, and Barbara Dennerlein. In addition to the B3, there are a number of smaller Hammond instruments, known as the spinet series (M3, M100, L100, T100). Bigger console models, many of which were designed to suit the needs of American (USA) churches or theatres (H100, X66, X77, E100, R100, G-100), were also manufactured.

The production of electro-mechanical organs ceased in 1974. Thereafter, Hammond built fully electronic organs. Today, people at Hammond-Suzuki are more conscious of their glorious tradition and produce fine electronic drawbar organs. In 2002, they even introduced a new digital B3 model which mimics the design and functions of the classic B3 (except for the weight). The new B3 utilizes a real, mechanical, rotor speaker cabinet.

Tonewheel Sound Generation

Tonewheel sound generation resembles that of a siren. Of course, there's no air being blown through the holes of a revolving wheel. Rather, an electro-magnetic pickup, much like a guitar pickup is used.

A notched metal wheel, called a tone wheel, revolves at the end of a magnetized rod. The teeth of the wheel cause variations in the magnetic field, inducing an electrical voltage. This voltage/tone is then filtered, sent through the manuals, amplified, has vibrato and expression applied to it, and is then amplified.

An AC synchronous motor drives a long drive shaft. 24 driving gears with 12 different gear sizes are attached to the shaft. These gears drive the tone wheels. The frequency depends on the gear ratios, and the number of notches in the wheels. The Hammond is tuned to an almost exact well-tempered scale.

As with pipe organs that feature multiplexed registers, the Hammond organ uses certain generators for more than one purpose. Some high frequency wheels serve as the fundamental for high notes, and provide harmonics for lower notes. This has a positive impact on the overall organ sound, avoids detuning and stabilizes levels.

The Leslie

Don Leslie developed his rotor cabinets in 1937, and began marketing them in 1940. Laurens Hammond wasn't keen on the concept of rotating speakers at all! Leslie's approach was to simulate a variety of locations in the pipes, resulting in a new spatial perception for every note. The rotor speaker cabinets could simulate this effect, and the sense of space that they impart is incomparable, when placed side-by-side with any fixed speaker. The periodic undulations in sound and volume, and the vibrato caused by the doppler effect aren't all there is to the Leslie sound—it's the space effect, too!

The first Leslie, the model 30, had no chorale, just tremolo and stop. The Chorale idea (which came much later) was borne of a desire to add a vibrato to the organ. Chorale offers far more than a simple vibrato, and was first introduced to the market with the 122/147 models. At this time, Leslie also added the "Voice of the pipe organ" label to his cabinets.

It wasn't until 1980 that the two companies and brand names came together, six years after the last tonewheel organ was built. Mechanical Leslie rotor cabinets are still being built today, by the Hammond-Suzuki company. Even the newest digital B3 model is combined with a real, mechanical, Leslie cabinet.

EVD6 23

The EVD6 is a virtual emulation of the classic Hohner Clavinet D6.

The sound of the Hohner Clavinet D6 is synonymous with funk, but was also popularized in the rock, pop, and electric jazz of the 1970s, by artists like Stevie Wonder, Herbie Hancock, Keith Emerson, Foreigner, and the Commodores. If you've heard "Superstition" or "Higher Ground" by Stevie Wonder, then you know that the D6 is the funkiest instrument alive!

About the EVD6

The EVD6's synthesis engine emulates the sound of the Hohner D6 Clavinet. It improves on the original in that it can be used in stereo, and unlike the original D6, the EVD6 has no noisy hardware components that compromise the sound of the instrument.

The 60-key range (F to E) of the original instrument has been extended across the full MIDI range (127 notes) while maintaining the dynamics and scaling of the sounds.

The EVD6's engine also simulates the various string buzzes, key clicks, and the tone of the pickups found in the original instrument. It synthesizes the pluck and bite of the attack phase, as well as the sticking of the hammer pads. The sound generator reacts smoothly, musically, and precisely to the 127 steps of velocity sensitivity, as defined in the MIDI specification. You can almost feel the strings beneath your keyboard!

The extensive set of String parameters allow you to radically alter the tone of the EVD6, enabling you to simulate an aging Clavinet, or to create new instruments. The EVD6 is capable of some truly unique sounds, which you'll discover when exploring it, and auditioning some of the included settings.

You will also discover an integrated effects processor incorporated into the EVD6's luxurious front panel, which provides a number of classic effects popularly-used with the original Clavinet. The algorithms featured in the effects processor have been adapted, and optimized, for the EVD6. Included are: a great sounding Wah, Modulation, and Distortion circuit.

The EVD6 Parameters



Most of the slider parameters on the EVD6 interface are mapped to zero-centered ranges—if a slider is in its middle neutral position, it doesn't affect the base sound of the selected EVD6-model. If the slider is moved to the left or right, it will scale the original parameter value positively or negatively by that amount.

The EVD6 front panel can be broken down into five main sections:

- Global section: Contains global parameters
- Excite/Click and String sections: Offers parameters adjust string response
- Pickup display: An adjustable visual representation of pickup position
- Effects section: Parameters for the integrated effects.

Global Parameters

The global parameters are found in the lower-left portion of the EVD6 interface.



Voices

Sets the maximum number of voices that can sound simultaneously. Lowering the value of this parameter limits the polyphony and also the processing requirements of the EVD6.

There are two monophonic settings: mono and legato. Each setting provides only one voice for playing the EVD6. In the mono setting, the EVD6 voice is triggered each time a key is pressed. In the legato mode, the EVD6 sound shaping processes is not triggered if the notes are played legato—only the pitch changes. If the notes are played staccato, an EVD6 voice with all sound shaping processes is triggered. Minimal CPU power is used when the instrument is operated monophonically.

The maximum setting is 24, allowing for sustained glissandi. A setting of 24 will, of course, be more processor-intensive. *10* is the default. Click-hold, and use your mouse as a slider to adjust.

Tune

Adjusts tuning in one cent intervals. A value of 0 equals concert-pitch A 440 Hz. The range is ± 50 cents or, in more music-related terminology, plus or minus half a semitone. For transpositions in semitone or octave steps, please use the Instrument Parameter box in the Arrange window, as per any standard MIDI instrument.

Bender

Determines the bend range, in semitone steps. Click-hold, and use your mouse as a slider to adjust.

Warmth

Sets the amount of random deviation from an equal-tempered scale. High values add life to sounds. It can be useful for simulating an instrument which has not been tuned for a while, or for slightly thickening a sound. When playing chords, the *Warmth* parameter creates the warm detuning or beating effect between the chord's notes. Click-hold, and use your mouse as a slider to adjust.

Stretch

The EVD6 is tuned to an equal-tempered scale. As a deviation from this standard tuning, you can stretch the tuning in the bass and treble ends of the sound. This simulates the way stringed keyboard instruments such as pianos are tuned, attempting to find a more constant tuning balance between high and low notes. The *stretch* feature was included in case you wish to use the EVD6 alongside acoustic pianos. When arranged in conjunction with an orchestra or synthesizers, the stretch tuning facility should *not* be used. Click-hold, and use your mouse as a slider to adjust.

Note: The tones of clavinets, harpsichords, and pianos have inharmonicities in their harmonic structure. The frequencies of these overtones (harmonics) are not exact, whole-number multiples of the base frequency. They are only approximate and are, in fact, a little higher. Because of this, the overtones of lower (tuned) notes are more closely related to the main frequencies of the upper notes. Also see "Stiffness/Inharmonicity" on page 370.

Note: When applying *Warmth* and *Stretch*, you should consider that these parameters may result in a detuned sound, which is similar to a heavy chorus effect.

Pressure

On an original D6, applying pressure (aftertouch) to a depressed key raises the pitch slightly. The Pressure parameter allows you to do this, or alternately lower the pitch by pressure. Click-hold, and use your mouse as a slider to adjust. Range: -1.00 to +1.00

Filter Switches

The four filter switches emulate the original switches on the D6, with one exception. When all switches are set to off, you'll hear the unfiltered sound, rather than the original D6's humming silence. Simply click anywhere on each switch to toggle between its on/ off position. Active switches are indicated by pale green lettering, and by being depressed towards the bottom of the plug-in window. You may use the filter switches in any combination of on/off positions.



- Brilliant: Makes the sound nasal/cuts bass.
- *Treble*: Makes the sound sharper/cuts bass more gently.
- *Medium:* Makes the sound thinner/slight bass reduction.
- Soft: Makes the sound softer/more muted.

Pickup Switches

As with the original D6, the two pickups can be used in different modes. The AB and CD switches are used to change modes. The internal wiring of the two pickups changes in accordance with the different switch positions, and with it, the sound at the combined pickup output. The EVD6 features an additional menu that displays the current pickup mode above the pickup switches. More information on the use of these parameters/the pickups is found in "Pickup Parameters" on page 371.



C/D Switch	A/B Switch	What It Does
Down	Down	Neck pickup—warm sound
Down	Up	Bridge pickup—bright sound
Up	Up	Both pickups—full sound
Up	Down	Both pickups out of phase—thin sound

Stereo Spread

The Stereo Spread control offers two sections: Pickup and Key. You can use both spread types at the same time. They will automatically be mixed. The area around the circular Stereo Spread button graphically displays the effect of both stereo spread parameters.

Pickup

While the original D6 only has a mono output, the EVD6 has stereo capabilities. When both pickups are active (upper+lower and upper-lower modes), the two pickup signals can be spread across the stereo spectrum. To adjust the position of the stereo spread, click-hold on the up/down arrows in the lower half of the circular button—in the Pickup section.



Turning the Pickup Spread control up will move the signals of both pickups away from the center position—one to the right, and the other to the left. Set Spread to its maximum value for extreme left/right panning. Range: 0.00 (center, no effect) to 1.00 (full left/right stereo).

More information on the use of the pickups is found in "Pickup Parameters" on page 371.

Key

Sets a keyscale modulation of the panning position. The keyboard position determines the panning. The center position is MIDI note 60 (Yamaha C3). To adjust the *Key*(board) position, click-hold on the up/down arrows in the in the Key section (top half) of the circular button.

When fully turned up, the extreme left/right position will be reached at MIDI note 60 \pm 30 semitones. Range: (center, no effect) to 1.00 (full left/right stereo).

Model

The Model parameter allows you to select a basic type of tone, or model. Each model has its own unique tonal characteristic designed to create very different sounds. Each model is an instrument in its own right, and can immediately be played, without any further editing. Each model is explained below.



To select a model, click-hold in the area between the Stereo Spread and Level controls, and make your choice from the pop-up list. Release the mouse button once your selection is made. All EVD6 parameters are available to further shape the tonal character of the model.

In some respects, you can view choosing between Model parameters similarly to selecting an oscillator waveform in a synthesizer. As with raw synthesizer waveforms, the editing parameters can affect the model quite differently. As an example, particular *Excite* settings may make one model more nasal sounding, and another model more noisy. These behavioral differences are a result of the unique harmonic structures used by each model.

The Models

This section describes the characteristics of the various clavinet models being emulated.

Class(ic) D6

An almost exact emulation of the original D6. It includes string noises on long decays, and realistic release behavior following the release of the key(s). Each D6 was unique in its way, so feel free to adjust the many sound shaping controls, in order to match the sound of D6 clavinets that you have heard.

Old D6

This model emulates a well-worn D6. Hammers and strings are a bit aged, and worn. The sound of the sticky hammer heads is emulated, as well as the typically richer sound in the bass range.

Sharp D6

Very sharp with a lot of bite—nice with wah wah and phaser.

Mello(w) D6

As the name suggests, a mellow fellow.

Basic

Basic, simple clavinet.

Domin(ation)

A powerful model with a strong and punchy attack—reacts more aggressively to velocity than other models.

GuruFnk (Guru Funk)

In the lower bass-octave ranges, the string oscillations become increasingly resonant over time, until they finally collapse (after 20 to 30 seconds). Higher notes have a much shorter decay, which also applies to their resonating behavior. This model invites heavy, funk-style bass playing in the lower octaves. It's nice with a little phaser, and sustained chords, when playing low bass notes. Adding a Logic Studio Delay plug-in is also a great option!

Harpsi(chord)

Harpsichord-like model.

Pluck

Plucked string. Changing the pick-up positions modifies the plucked string, making the sound more guitar-like. Harp style sounds are also possible, by positioning the lower pick-up around the mid position. To get a harp sound, increase String Decay, Release, and Excite Shape and decrease Excite Brilliance.

(Tuned) Wood

Somewhat wooden, thin, and with some inharmonic overtones. Can sound slightly detuned in some contexts.

Ltl (Little) India

Sitar-like sound, rich in resonance.

S(tring) Bells

A bell model with strong inharmonic overtones (inharmonicities).

Dulcimer

Dulcimer-like model.

Picked

This model emulates a picked nylon string.

Special Notes About the Models

You may note some spots on the keyboard where the sound changes significantly between adjacent keys. This is intentional, and reflects the behavior of some of the real clavinet models emulated by the EVD6. The original D6 has some strong key-to-key timbral differences, with the most obvious one being between the highest, wound string, and the lowest, non-wound string.

If you're a player who likes the original's sound, but not the original's mechanical timbre jumps, the EVD6 offers a smoothed model—MelloD6.

When using a setting with both pickups quite close to the upper end of the strings and Brilliant + Treble filter switches active, the fundamental tone is quite weak in the output signal. As such, you will mostly hear the overtones that are not exactly in tune for inharmonic models (Wood, for example). Try moving the pickups to the center, and deactivate all filter switches to circumvent this detuned effect.

Level

Sets the post -effect level, in dB (decibels). Click-hold, and drag, to adjust. If the MIDI controller used for Expression scales the output level unless assigned to Wah or Damper.



Damper Wheel and Damper Ctrl

The original D6 features a damper slider on the right-hand side of the keyboard that allows the player to create muted string sounds.

There are two ways to adjust the Damper Wheel:

- Click-hold and drag the Damper Wheel.
- Use a MIDI controller, such as your keyboard's mod wheel.

The Damper Wheel position is saved with the sound.



The Damper Ctrl (number) parameter allows you to select the MIDI controller that moves the Damper Wheel. Click and choose a controller number/name selection from the menu. Release the mouse button, once your selection is made.

Note: You can use MIDI Velocity to control the Damper Wheel. Just select the Velocity parameter from the menu.

When you move the software-wheel via MIDI, it moves on-screen as well. MIDI control can be disabled by selecting the off option, found in the menu.

Note: The Wah Ctrl parameter is discussed on page 374.

Velocity Curve

The EVD6 offers nine preset velocity curves. These allow you to set up a curve which is suitable for your playing style, or the sound.

The nine curves available are: fix25%, fix50%, fix75%, fix100%, convex1, convex2, linear (the default), concave1, and concave2.

Excite Parameters

Excite describes the string excitation. In other words, it sets how much physical power stimulates the string to oscillate.



Shape

Adjusts the attack shape, allowing you to simulate the hardness of the rubber hammers in the original D6. As the instrument aged, the hammers would become worn, split, and so on, which had an impact on the overall brightness/tone of the D6. Negative values (to the left) provide a softer attack, while positive values result in a harder attack. Range: -1.00 to +1.00

Brilliance

Controls the harmonic content of string excitation. Positive values (to the right) result in a sharper sound. Negative values result in a more muted sound.

Click Parameters

The rubber hammers of the original D6 age and decay, just like piano hammer felts. Well-loved (worn out) D6's produce a distinctive click when a key is released. This is due to the string sticking to the rubber hammer, before being released. The characteristics of this release click are part of each model, and can be finely adjusted with the following parameters.



Intensity

Positive values increase the level of the release click above the original model setting. Negative values reduce the level—a value of -1.00 equals no release click. If you'd like to simulate an old D6, increase the value, by moving the slider to the right. Range: -1.00 to +1.00

Random

Controls the amount of random click level variations across the keyboard. This slider simulates the wearing of some hammers, but not all of them, emulating the real-world wear and tear of the original. The further to the right the slider is moved, the greater the variation between key clicks on some keys. If all the way to the left, all keys have the same level of key click. Range: 0.00 to +1.00

Velocity

The Velocity parameter controls the level of release click modulation by velocity—Note On velocity or Note Off/release velocity. Range: 0.00 to 1.00. The selection of note on/off information as the modulation source is determined by the KeyOn/Key Off buttons.

KeyOn/KeyOff Button

Press the appropriate button to select the type of velocity information that should be used for release click level modulation—click the KeyOn button, if you wish to use your attack velocity (how hard you hit the keyboard) as the value for the key click. If you wish to use your release velocity (how quickly you release the keys on your keyboard) to determine the value of the key click, click the KeyOff button.

Note: KeyOff requires a keyboard with release velocity facilities.

Note: The Velocity parameter must be set to a reasonable level in order for the KeyOn/ KeyOff modulation to be effective.

String Parameters

The selected model determines the basic behavior of the strings, but the following parameters allow you to modify several string characteristics relative to the model setting. See "Model" on page 364, for further information on model selection.



Decay

Positive values provide a longer Decay time after attacking a note. Negative values reduce the decay time. Range: -1.00 to +1.00

Release

Positive values provide a longer Release time following the physical release of a key. Range: -1.00 to +1.00

Damping

Lets you modify the damping of strings. Damping is essentially a faster decay for the higher partials/harmonics in a sound, and is a property of the string material used (high damping for catgut strings, medium damping for nylon strings, low damping for steel strings). Sonically, damping results in a more mellow and rounded, or woody sound, dependent on the Model in use. A positive value will make the sound more mellow, and a negative value will allow more of the higher partials through, making the sound brighter. Range: -1.00 to +1.00

Tension Mod

Lets you add a nonlinear effect to the strings, which usually results in the pitch being slightly higher, immediately after being plucked/struck/strummed. It is common to all stringed instruments, like the D6, guitars, and so on. Tension modulation characteristic is built into each model, but can be further modified with the Tension Mod parameter. The range of this parameter is quite large, and can be used to obtain weird sound effects from the EVD6. It can also be used to simulate an out-of-tune Clavinet, or as a quick and dirty sitar sound, for those Norwegian Wood covers. Range: -1.00 to +1.00

Stiffness/Inharmonicity

Allows you to intensify/reduce the strength of inharmonicity in the sound. When combined at different levels, these parameters can create metallic, bell-like sounds, or DX-like electric piano style sounds. They can also be useful for wood bass sounds. Experiment with both parameters, on each Model.

The higher the level of the Inharmonicity parameter, the lower its threshold to incoming frequencies. In other words, the Inharmonicity parameter determines the lowest harmonic, above which inharmonic spectral spreading becomes relevant. Range: -1.00 to +1.00

Stiffness controls the intensity of this stretching/spectral spreading.

The keynote is not affected by these parameters.

Pitch Fall

Due to the physical construction of the original D6, the pitch of each note falls immediately after releasing the key. The intensity of this effect, which varies with each model, can be modified with this parameter. To completely deactivate the pitch fall, regardless of the selected model, set this parameter to the leftmost position (-1.00). Range: -1.00 to +1.00

Pickup Parameters

The original D6 is equipped with two electromagnetic pickups, much like those found in electric guitars: one below the strings (lower) and one above (upper).



Pickup Position

In contrast to the fixed pickups of the original instrument, the EVD6 pickups can be set to arbitrary positions *and* angles. To do so, simply click-hold on one end of the desired pickup (Upper or Lower) and drag the end to another position. Release the mouse button when done. Both values can be moved simultaneously. To do so click and drag the point in the middle of the pickup to a new position.

The numerical *upper* and *lower* panels, to the top-left of the window, indicate the current position of each pickup with respect to the string. A value of 50 (percent) means that the specific pickup end is positioned above/below the centre of the string, resulting in a full-bodied tone. When the pickup approaches either end of the string (values near 0 or 99), the tone becomes thinner.

In the graphical Pickup display the strings are aligned from left to right in respect to pitch—low strings to the left, high strings to the right.

Them moving pickup positions, repeatedly strike a note in order to hear the effect that the pickup position has on the overall tone of your sound.

Interesting, phaser-like effects can be achieved by automating the pickup positions.

Important: It is possible to cross-over the pickups in the Pickup Position window. This may lead to a hole (non or very soft sounding notes) within your keyboard range. This is due to a phase-cancellation between the pickups. If you encounter such cancellations, adjust one (or both) of the pickups until the required notes are playable.

Pickup Mode

Clicking the AB and CD switches changes the virtual wiring of the two pickups. The current wiring, the EVD6 calls it Pickup Mode, is displayed in the Pickup Mode panel. You can also click directly on the Pickup Mode panel, and choose the desired mode from a menu.

- C + A = Lower
- C + B = Upper
- D + A = Lower-Upper
- D + B = Lower + Upper

Also see "Stereo Spread" on page 363, and "Pickup Switches" on page 362.

Effects Parameters

No Clavinet simulation would be complete without a selection of effects processors. The EVD6 doesn't disappoint in this regard, incorporating three footpedal effects that have formed an integral part of classic Clavinet sounds over the decades. Each effect was painstakingly modelled on effects pedals that were available in the heyday of the Clavinet—the 1970s—ensuring that vintage sound in your performances.

Needless to say, you can also take advantage of Logic Studio's extensive range of effect plug-ins, to further tailor your sound.

Distortion

The integrated distortion effect offers adjustable intensity and tone. Range: Tone – 2000 Hz to 20,000 Hz, Gain –0 dB to 20 dB. Using low Tone and Gain settings allows the Distortion unit to create warm overdrive effects. Bright and screaming distortion effects are produced with high Tone and Gain settings.



Compressor

The compressor always precedes the Distortion effect. This allows you to increase/decrease the perceived gain, to provide the desired input level to the Distortion circuit.

The compressor allows you to create really crunchy distortions, coupled with wah, or phaser. It can also be useful for enhancing the keyclick sound, and emphasizing harmonics in the various models.

Compression Ratio

The Compression Ratio panel allows you to adjust the slope of the compression applied. To adjust, simply click-hold on the panel, and use the mouse as a slider.

The Compressor is tied to the Distortion effect, and always precedes it. As such, the Effects Order parameter is very important for placement of this compressor in the effects chain. Please see "FX Order" on page 376 for further information.

Note: If the Compressor/Distortion is the last effect in the chain and its gain is turned down but the Compression Ratio is high, you will effectively compress the output signal of the EVD6.

Wah

The typical Wah effect is generated by a dynamically moving filter. The EVD6 offers simulations of several classic wah effects, as well as some basic filter types. Possible values are: off, ResoLP, ResoHP, Peak, CryB, Morl1, Morl2. The abbreviations are for Resonant Low and Highpass filters, Peaking filter, CryBaby, Morley 1, and Morley 2. The latter three model famous effects pedals that continue to be manufactured.



Note: The combination of wah, followed by distortion, delivers those sought-after funky fuzz-wah results.

Wah Mode

Simply click-hold on the (Wah) Mode panel, and choose the desired (pedal effect) model from the pop-up menu.

Range

Determines the cutoff frequency of the filter set with Wah Mode. With Range set to the left, the cutoff will only move in a narrow range. To provide a wider control range, turn the Range knob to the right.

Envelope (Depth)

Sets the sensitivity of the envelope in respect to your performance and thus the resulting filter modulation depth. The envelope shape follows the dynamics of your performance. An auto wah effect is produced by using an envelope follower to control the filter cutoff automatically. Turn Envelope to the right to increase the modulation depth.

Wah Ctrl

Lets you define the MIDI Controller (number/name) used as a manual wah effect control. For example, MIDI foot controllers are popular wah effect controllers, and you can use this parameter to configure the EVD6 to use your MIDI foot controller to control the wah effect. You can also use MIDI Velocity to control the wah effect. Just click into the respective parameter field and select velocity from the ensuing pop-up menu. MIDI-control/coupling can be disabled by selecting off.



Note: Both envelope and a manual controller can control the wah simultaneously. In this situation, the effect of the envelope and manual controls are mixed.

As with the Damper, there is a slider named Wah Pedal Position that always represents the current pedal position. This ensures that the most recent pedal position is saved with the sound. The Wah Pedal Position Slider is only available in the Controls view. Choose View > Controls to switch from the Controls view to the Editor view. The Pedal Position can be automated by either recording the MIDI controller messages, or by using the Track Automation system.

Modulation

The EVD6 features a Modulation unit with three switchable modulation effect types.



Modulation unit

Mode

The Mode panel allows you to select either a Phaser, Flanger, or Chorus as the modulation effect. Click-hold, and make your choice from the pop-up menu.

Phaser

The Rate parameter adjusts the speed of phasing, and the Intensity parameter adjusts the depth of phasing. Ranges: Rate 0.10 Hz to 10 Hz, Intensity 0 to 100

High values lead to very deep, self-oscillating phase shifts, for those cutting sounds (which can also damage ear and speakers, so be careful!).

Chorus

Adjusts the speed of the Chorus effect, and the Intensity parameter adjusts its depth. High Intensity values lead to ensemble-type effects.

Ranges: Rate –0.10 Hz to 10 Hz, Intensity –0 to 100

Flanger

The Rate parameter adjusts the speed of flanging, and the Intensity parameter adjusts the depth of flanging. Ranges: Rate –0.10 Hz to 10 Hz, Intensity –0 to 100.

FX Order

The order of the serial effects combination can be selected here. The four choices are:



- WDM: Wah > Distortion > Modulation
- DWM: Distortion > Wah > Modulation
- MDW: Modulation > Distortion > Wah
- WMD: Wah > Modulation > Distortion

Just like the foot pedals which could be freely connected to each other in series, the EVD6 effect section invites you to experiment.

The freely assignable effect routing is especially useful for selecting whether a distorted signal shall be wah-filtered (for funkier sounds), or if the wah-filtered sound shall be distorted (for screaming sounds), as one example.

Controlling the EVD6 via MIDI

It is possible to control and automate the parameters of the EVD6 and other plug-ins using the MIDI controls provided by many master keyboards or MIDI fader boxes.

MIDI controller assignments (Wah Ctrl, Damper Ctrl) are only updated if the default setting is loaded or a setting that was saved with a project is loaded, with one exception: If Velocity is chosen in a setting, the assignment is updated. When selecting a setting that has any controller other than velocity assigned, the default value or the last manually selected value (other than velocity) is used.

All parameters that allow you to select a MIDI controller offer a Learn entry. If this option is selected, the parameter will automatically be assigned to the first appropriate incoming MIDI data message. The Learn mode features a 20 second time-out facility: If the EVD6 does not receive a MIDI message within 20 seconds, the parameter will revert to its original MIDI controller assignment.

Note: As the new entry is added at the top of the list, existing automation data needs to be incremented by one.

A Brief History of the Clavinet

German Company, Hohner, manufacturer of the Clavinet, were known mainly for their reed instruments (harmonicas, accordions, melodicas, and so on), but had made several classic keyboards, prior to the first incarnation of the Clavinet, known as the Cembalet.

Musician and inventor, Ernst Zacharias, designed the Cembalet in the 1950s. This was intended to be a portable, amplifiable version of the Cembalo, or Harpsichord. Its mechanism worked by plucking the end of a flat reed with the key, which was then picked up and amplified in much the same way as an electric guitar.

A year or two after the Cembalet's release, two Pianet models appeared. Both the CH and N models used flat reeds for tone generation, but employed a very different plucking/striking action. When a key was depressed, it engaged a sticky pad with a foam backing, which actually stuck to the reed. When the key was released, the weight of the key caused the pad adhesive to free itself from the reed. This made the reed vibrate, and this vibration was then amplified.

The model T Pianet was released several years later, and utilized a soft rubber suction pad on the reeds, rather than the adhesive of the CH and N models. This method still had several drawbacks, however, as the dynamics available from the keyboard were limited. As a further shortcoming, all reeds were damped on release, thus negating the possibility of obtaining sustain via a foot pedal. Despite these glaring problems, the sound of the model T Pianet was popularized by bands such as the Zombies and Small Faces, in the 1960s.

In the years between the releases of the Pianet N and T models, Zacharias invented what was to become Hohner's most successful, and certainly funkiest keyboard—the Clavinet. The Clavinet was designed to replicate the sound of a Clavichord, but with an altogether fuller sound. (The Clavichord was notoriously thin sounding)

The early models—Clavinet I with built-in amp, Clavinet II with tonal filters, Clavinet L with its bizarre triangular shape, all led to the Clavinet model C. This, in turn, was refined into the D6—a portable, amplifiable keyboard. The D6 used a hammer striking a string against a metal surface to produce its tone. It had a fully dynamic keyboard—since the striker is directly underneath the key, the harder you hit, the louder and more vibrant the tone.

Mention the Clavinet today and most people will automatically think of Stevie Wonder's "Superstition"—a recording that owes as much to the D6 as it does to the artist that wrote and performed it. The D6 was later superseded by the E7 and the Clavinet/Pianet Duo. These were basically the same as the D6, but more roadworthy, quieter, and better protected against proximity hums than previous models.

How the D6 Clavinet Works

Each D6 keyboard key forms a single arm lever. When a key is depressed, a plunger underneath touches the string and presses it onto an anvil. The string impinges on the anvil with a strength determined by key velocity. This affects the dynamics of the sounding string.

These mechanical vibrations are converted into electrical frequencies through magnetic pick-ups which are amplified and reproduced through the loudspeaker.

As the key is released, contact between plunger and anvil is immediately broken, leaving the wool-wound part of the string free, so that the string vibration is immediately muted.

Double-Triggered Notes

When experimenting with the EVD6, or auditioning some of the included settings, you may encounter sounds which seem to be triggered on both the note on *and* the note off.

This is actually a feature, which emulates the original D6. The real D6 has the "problem" of the strings sticking to the hammers if they are worn out, producing a second trigger when the key is released. You can adjust the intensity of this key-off click, with the Intensity slider in the Click section (see the "Click Parameters" on page 368). Move the slider to the left, and the second key off trigger will no longer be audible!

EVP88 **24**

The EVP88 virtual electric piano instrument simulates the sound of different Rhodes and Wurlitzer pianos, as well as the sound of the Hohner Electra piano.

The sounds of various Fender Rhodes pianos are among the most popular keyboard instrument sounds used in the second half of the twentieth century. The various Rhodes models have been popularized in a wide range of musical styles, ranging from pop, rock, jazz, and soul, as well as more recent genres like house and hip hop. Nearly as popular was the Wurlitzer piano, which enjoyed most of its success in the seventies. The Rhodes, Hammond organ and subtractive analog synthesizers were considered the fundamental instruments in the keyboard rigs of rock musicians between 1965 and 1985, and they appeared to be incomparable and unbeatable ... until now.

About the EVP88

The EVP88's piano synthesis engine is designed solely for the ultra-realistic simulation of electric pianos. It offers silky smooth dynamics and scaling of the sounds over the entire 88-key range, and it does not suffer from the abrupt changes in sound that typify sampled instruments. There are no audible loops, and you'll never hear any lowpass filters closing while the sound of a note is decaying.

The EVP88's engine also simulates the physical movement of the various electric piano reeds, tines, and tone bars in the electric and magnetic fields of the pickups found in the original instruments. It synthesizes the ringing, smacking, and bell-like transients of the attack phase, as well as the hammer action and damper noises. The sound generator reacts smoothly, musically, and precisely to the 127 steps of velocity sensitivity as defined in the MIDI specification.

Incorporated into the EVP88's front panel, you will discover an integrated effects processor which provides a number of classic effects popularly used on electric piano sounds. The algorithms featured in the effects processor have been specifically designed, adapted, and optimized for the EVP88. Included are: a great sounding equalizer, an overdrive, a stereo phaser, a stereo tremolo and stereo chorus.

The FVP88 Parameters

This section outlines the various controls available on the EVP88 front panel.



Global Parameters

Global parameters affect the entire EVP88 instrument, rather than specific electric piano models.



Model

The large Model dial allows you to choose the electric piano model. When selecting a new model, all currently sounding voices are muted, and all parameters are reset to standard values. Because of this, you will usually want to select the model before attempting to edit the effect and parameter settings. There are several Rhodes models available, such as the Mark I, Mark II and the suitcase piano plus the Wurlitzer and Hohner Electra Models. You can read more about the simulated instruments in "A Brief History of the Clavinet" on page 377.

The names of these instruments are registered trademarks and are protected by law.

Voices

The Voices parameter allows you to set the maximum number of voices that can sound simultaneously. Lowering the value of this parameter limits the polyphony and processing requirements of the EVP88. When the parameter is set to 1, the instrument is monophonic, and uses minimal CPU power. The maximum setting is 88, allowing for glissandi over the entire keyboard range with the sustain pedal depressed. A setting of 88 will, of course, be more processor-intensive.

Tune

The global Tune setting lets you tune the EVP88 in one cent increments. A value of 0 equals concert-pitch A 440 Hz. The range is ± 50 cents or, in more music-related terms, plus/minus half a semitone. For transpositions in semitone or octave steps, please use the Region Parameter box in the Arrange window, as per any standard MIDI instrument.

Modeling Parameters

The modeling parameters specifically affect the currently selected model.



Decay

Decay time of the piano sound. The lower the value, the less the sound sustains, and the higher the level of damping applied to the vibration of the tines. When short values are used for this parameter, the main tone is more pronounced, and sounds longer than the transient harmonics. The effect is somewhat reminiscent of an electric guitar string being damped with the palm of the picking hand. Electric pianos can be modified in a similar way. Longer settings result in more sustain and a less dynamic feel.

Note: Check out Logic Studio's compressor plug-ins and experiment with different settings for decay.

Release

The release parameter determines the amount of damper applied after the keys are released. Extremely long settings allow you to play the piano like a vibraphone.

Bell

Bell determines the level of the inharmonic treble portion of the tone. It is useful for emulating a number of classic and typical electric piano sounds.

Damper

This parameter sets the level of the damper noise caused by the damping felt hitting the vibrating tine.

Stereo

When set to high values, bass notes sound from the left, and treble notes from the right channel. The effect is nice and spacey, but it is not typical for vintage electric piano sounds.

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Note: Using Logic Studio's plug-ins, you can process the upper notes differently to the lower ones. With appropriate signal processing routings, you can, for example, add some bass via an EQ in the left bass channel and apply a little echo to the higher notes. Be creative!

Stretch and Warmth



The EVP88 is tuned to an equal-tempered scale. However, you can deviate from this standard tuning, and stretch the tuning in the bass and treble ranges, much like acoustic pianos (especially upright pianos). You can also modulate the tuning of each note randomly.

Note: The tones of upright pianos, and to a lesser extent (due to their longer strings) grand pianos have inharmonicities in their harmonic structure. The frequencies of the harmonics are not exact, even multiples of the base frequency. They are only approximate and are, in fact, a little higher. As a result of this, the harmonics of lower (tuned) notes are more closely related to the main frequencies of the upper notes. Due to the lack of strings, this inharmonic relationship is not true of electric pianos, nor the EVP88. The stretch feature was included for situations where you may wish to use the EVP88 in an arrangement alongside an acoustic piano.

Lower Stretch

Sets the deviation from the equal-tempered scale in the bass end of the sound. The higher the value, the further *down* the low notes are tuned. At a setting of 0, the EVP88 is tuned to an equal-tempered scale, with each octave down exactly halving the frequency.

Upper Stretch

Sets the deviation from the equal-tempered scale in the treble end of the sound. The higher the value, the further *up* the high notes are tuned. At a setting of 0, the EVP88 is tuned to an equal-tempered scale, with each octave up exactly doubling the frequency.

Warmth

Amount of random deviation from an equal-tempered scale. High values add life to sounds.

Note: When applying Warmth and Stretch, you should consider that these parameters may result in a detuned sound similar to a heavy chorus effect.

Effects

The EVP88 features built-in Equalizer, (Over)Drive, Phaser, Tremolo, and Chorus effects.

Equalizer

The Equalizer allows you to boost or cut the high and low frequency ranges of the EVP88 sound.



Treble

This is a conventional filter for the high frequency range. Depending on the model selected, shelving or peak type filters are utilized, with optimized frequency ranges for each model pre-selected.

Bass

This is a conventional filter for the low frequency range. Depending on the model selected, shelving or peak type filters are utilized, with optimized frequency ranges for each model pre-selected.

Note: By defeating the treble and bass frequency ranges, you can achieve a very direct and aggressive sound with a more dominant mid range. If you require more precise equalization, remember that you can insert any of Logic Studio's equalizer plug-ins into the instrument channel strip. The Tone control in the overdrive effect can also be used to further contour the sound.

Drive

Electric pianos sound best when playing through tube amplifiers. Tube amplifiers offer a wide range of tones, ranging from the subtle warmth of crunchy guitar amplifiers through to psychedelic, screaming rock distortion. The EVP88 features an overdrive effect, which simulates the saturation characteristics of a tube amplifier stage. The overdrive process is the first signal processing circuit in the effects chain available in the EVP88.



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Tone

The Tone control equalizes the sound before the virtual tube amplifier circuit amplifies or distorts it. You can choose a more mellow tonal color here, and still boost the treble with the equalizer after the overdrive circuit. If you prefer harsh distortion characteristics typical of overdriven transistor stages, use higher tone parameter values. If the sound becomes too hard, you can defeat the treble via the *Treble* control, post the overdrive process.

Gain

The Gain control determines the amount of harmonic distortion.

Phaser



The phaser pedals popularized by electric guitarists are also popular among electric pianists—especially in the electric jazz, jazz-rock and pop styles of the seventies. Classic four-stage phasing effects are based on phase shifting using modulated all-pass filters. Mixing the phase-delayed signal with the original signal results in characteristic notches in the frequency response curve, also called comb-filtering. The frequencies of the notches in the frequency range are not harmonic and these notches are shifted up and down through the sonic spectrum by LFO (low frequency oscillator) modulation.

Note: Logic Studio offers more parameters in its Phaser and other modulation plug-ins. You can use these effects alternately to, or in conjunction with, the EVP88's Phaser. The parameters found in the EVP88 Phaser have much in common with the best analog phasers of the 60s and 70s, including subtle analog-style distortion. It offers the same 32 Bit internal processing and sound quality of the Logic Studio Phaser plug-in.

Rate

Sets the speed of the phasing effect. When set to 0, the Phaser is switched off.

Color

Sets the intensity of sound coloration introduced by the Phaser, caused by feeding the Phaser output signal back into its input.

Stereophase

Relative phase shift between the left and right channels, ranging from 0° to 180°. With 0° selected, the effect is most intense, but not stereophonic. With 180° selected, the effect symmetrically rises in the left channel while simultaneously falling in the right channel, and vice versa.

Tremolo



A periodic modulation of the amplitude (level) of the sound is known as tremolo. The modulation is controlled via an LFO. The Fender Rhodes suitcase piano features a stereo tremolo and many other electric pianos have a simple, but quite obtrusive, mono tremolo that can introduce a strange kind of polyrhythmic feel to performances.

Note: The original Wurlitzer piano has a mono tremolo with a fixed modulation rate of 5.5 Hz. For an authentic Wurlitzer sound, choose 0°. For Rhodes sounds, select 180°. The settings in-between result in nice spacey effects, especially at low LFO rates.

Rate

Sets the speed of the tremolo effect (LFO frequency).

Intensity

Adjusts the amount of amplitude modulation. With 0 selected, the tremolo effect is switched off.

Stereophase

At a setting of 0°, the level undulates in phase on both channels. With 180° selected, the modulation is perfectly out of phase, resulting in a stereo tremolo effect that is also known as auto panning. The effect is similar to manually turning the pan pot from side to side.

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Chorus

The well-known chorus effect is based on a delay circuit, the delay time of which is permanently modulated by an LFO, while the delayed effect signal is mixed with the original dry signal. It is the most popularly used effect on electric piano sounds. The single parameter regulates the intensity (the amount of delay time deviation), while the LFO rate is fixed at 0.7 Hz. Pay close attention when using high values as this may result in the piano sounding detuned.



Extended Parameters

The EVP88 features a number of extended parameters that are accessible via the disclosure triangle at the bottom of the EVP88 window.



- *Volume slider and field:* Sets the overall output level of the EVP88 (Range: –20 to +20 dB).
- Bend Range Down and Bend Range Up sliders and fields: Determines the pitch bend range in semitone steps.
- Chorus Rate slider and fields: Sets the speed of the Chorus effect, in Hz.
- Delay PP and FF sliders and fields: Determines the delay time (in milliseconds) when the keys are struck pianissimo (PP—soft) or forte (FF—hard).

Emulated Flectric Piano Models

This section provides some background information on the instruments emulated by the EVP88.

Rhodes

Harold Rhodes (born 1910) constructed arguably the most commonly known and widely used electric piano. Designed in 1946 as a piano surrogate for practice, education, and army entertainment, guitar manufacturer Fender successfully marketed the Rhodes piano beginning in 1956. The Fender Rhodes has become one of the most popular musical instruments in jazz, especially electric jazz. Once CBS took over production of the Rhodes in 1965, its popularity in pop and rock music grew. Despite further changes in ownership throughout Rhodes's history, the instrument remains ingrained in the mind of the public as the Fender Rhodes. There are also a number of Rhodes synthesizers (which were developed by the now-defunct synthesizer manufacturer ARP). Japanese synth and music technology manufacturer Roland owned the Rhodes name for a while, and released several digital pianos under the Rhodes moniker. From 1997, until his death in december 2000, Harold Rhodes again owned the name.

The Rhodes piano bases its method of sound generation on metal reeds which function much like a tuning fork. These are hit by a hammer action that works in a similar fashion to that of a grand piano action. The asymmetrically designed tuning fork consists of a thin tine and a massive tone bar, which are bolted together. Due to construction considerations, some of the tone bars are rotated by 90 degrees. The piano is kept in tune by the mass of a spring which can be moved along the tine. The tine oscillates in front of an electric pickup, similar to that of an electric guitar. This functions along inductive principles, with permanent magnets placed around the tine having a damping effect on its movement, thereby affecting the sound.

Like the output signal of an electric guitar, the Rhodes output signal is rather weak and needs quite a bit of pre-amplification. The Rhodes sound is not harmonically-rich. This is why so many performers used a treble boost or an overdrive effect when playing the Rhodes. As mentioned earlier, the Rhodes sounds its best when played through tube amplifiers.

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The Rhodes piano was also made available as a suitcase piano (with pre-amp and two-channel combo amplifier) and as a stage piano, without amplifier. Both of these 73-key portable versions have a vinyl-covered wooden frame and a plastic top. In 1973, an 88 key model was introduced. Smaller Celeste and bass versions were less popular. The Mk II (1978) had a flat top instead of a rounded one. This allowed keyboardists to place extra keyboards on top of the Rhodes. In 1984 the Mark V was introduced, and even sported a MIDI output. Around this time, Rhodes production decreased as most keyboard players invested in the more flexible (and lighter) digital synthesizers available. These keyboards could emulate the sound of older pianos, like the Rhodes, and also had the bonus of a range of great new piano sounds.

The individual characteristic sound of each Rhodes piano depends more on the adjustment and maintenance of the instrument than on the model. Early models had hammers covered with felt, resulting in a smoother sound than the newer models, which had neoprene-covered hammers. The suitcase piano featured a pre amplifier which could create a sound with a very dominant mid range. But appropriate pre amplifiers and equalizers can make a stage piano sound the same. The stage piano has no power cord—just like an electric guitar.

The MkII has no resonance clamps in the treble range, unlike former models. This is why it has a little less sustain in the treble range. The most significant differences in terms of sound depend on how close to the pickup the tine is adjusted. When the tine closer to the pickup the bell characteristic becomes more prominent. In the eighties, many Rhodes pianos were adjusted so that they had more bell.

Rhodes Models

- Suitcase Mkl
- Suitcase V2
- Bright Suitcase
- · Stage Piano Mkl
- Stage Piano MkII
- · Bright Stage MkII
- Hard Stage MkII
- MarkIV
- Metal Piano
- Attack Piano

The Metal Piano and Attack Piano models feature idealized sound qualities that can be aimed at with the original Rhodes instruments, but not achieved to the extent of these models. These models may not sound realistic, but they represent ideals that the Rhodes technicians might have had in mind when preparing their keyboards.

Wurlitzer Piano

This well-known manufacturer of music boxes and organs also built electric pianos which helped write pop and rock music history. The 200 series Wurlitzer pianos are smaller and lighter than the Rhodes pianos, with a keyboard range of 64 keys from A to C and an integrated amplifier and speakers.

The action resembles that of a conventional acoustic piano. It can be played with velocity sensitivity, just like the Rhodes. Its sound generation system is based on spring steel reeds which can be tuned with a solder weight. The Wurlitzer has electrostatic pickups: The reeds are supplied with a 0 volt current and move between the teeth of a comb, connected to a 150 volt current. The tone of the Wurlitzer, which was first manufactured in the early sixties, features many odd harmonics.

The Wurlitzer is best known as the signature piano sound of the band "Supertramp." You will know it from their "Crime of the Century" album. It can be heard on "Bloody Well Right," "Dreamer," "Hide in Your Shell" and also in "The Logical Song." You might also recognize the Wurlitzer sound when listening to Pink Floyd's "The Dark Side of the Moon" or "Wish You Were Here" ("Have a Cigar," "Money," "Time") and "I am the Walrus" by the Beatles.

Wurlitzer Models

- Wurlitzer 200 A
- Wurlitzer 240 V
- Soft Wurlitzer
- Funk Piano

The model Funk Piano offers special synthetic piano engine sound with exaggerated bass that was not part of any real Wurlitzers, but can be very useful and musical.

Hohner Electra Piano

Not to be confused with the all-electronic RMI Electrapiano, the extremely rare Hohner Electra Piano offers striking hammers like those of the Rhodes, but a stiffer keyboard action. It was designed to resemble the look of a conventional acoustic upright piano. Led Zeppelin's John Paul Jones played it on "Stairway to Heaven," "Misty Mountain Hop" and "No Quarter."

Hohner Electra Model

· Electra Piano

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EVP88 and MIDI

This section outlines some of the special MIDI control features available to the EVP88.

Adaptation of Your MIDI Keyboards Velocity Sensitivity

The EVP88 responds with extreme sensitivity to the velocity information transmitted with MIDI note messages. It's advisable to set the velocity and dynamic track parameters in Logic Pro with care. You can try the following tip to fine-tune the velocity curve if you find that you're not getting the right feel with your MIDI keyboard.

- Create a transformer object in the Environment, and cable it between the Physical Input and Sequencer Input objects on the Click and Ports layer.
- Set the transformer parameters so that all MIDI events with the condition note are set to Use Map under Vel in the lower operation line.

You can then draw your own individual keyboard velocity curve. For more detailed information, please refer to the Environment chapter of the Logic Pro 8 User Manual.

MIDI Controller List

Controller Number	Parameter Name	
1	Tremolo parameters: Intensity knob	
12	Model knob	
13	Model parameters: Decay knob	
14	Model parameters: Release knob	
15	Model parameters: Bell knob	
16	Model parameters: Damper knob	
17	Model parameters: Stereo Intensity knob	
18	EQ parameters: Treble knob	
19	EQ parameters: Bass knob	
20	Drive parameters: Gain knob	
21	Drive parameters: Tone knob	
22	Phaser parameters: Rate knob	
23	Phaser parameters: Color knob	
24	Phaser parameters: Stereophase knob	
25	Tremolo parameters: Rate knob	
26	Tremolo parameters: Intensity knob	
27	Tremolo parameters: Stereophase knob	
28	Chorus parameters: Intensity knob	

The EXS24 mkll is a *software sampler*. This means that instead of having a built-in sound set, it plays back audio files (called *samples*) that you load into it.

These samples are combined into tuned, organized collections of samples called *sampler instruments*. You can use the EXS24 mkll to play, edit, and create sampler instruments. You can assign the samples (in sampler instruments) to particular key and velocity ranges, and process them with the EXS24 mkll filters and modulators. As sampler instruments are based on audio recordings, they are ideally suited to emulating real instruments.

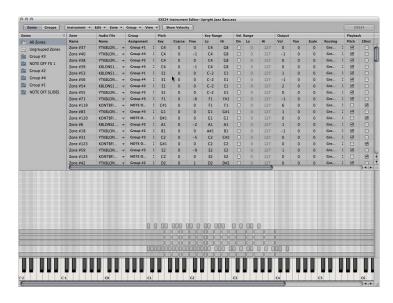
The EXS24 mkll ships with a library of sampler instruments in its native file format, known as the EXS format. It also allows you to import sampler instruments in the AKAI S1000 and S3000, SampleCell, Gigasampler, DLS, and SoundFont2 sample file formats.

The EXS24 mkll interface consists of two windows:

• *Parameters window:* Offers numerous sample processing and synthesis options, enabling you to tailor EXS instrument sounds to meet your needs.



• Instrument Editor: Used to create and edit sampler instruments.



Using the EXS24 mkll typically involves the following steps:

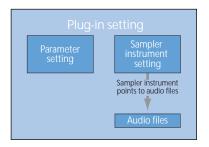
- 1 Load or import a sampler instrument.
- 2 Change the overall sound of the sampler instrument in the EXS24 mkll Parameters window by turning the knobs, pressing switches, and moving sliders. You can also automate these controls, enabling dynamic changes over time.
- 3 Edit specific samples in the Instrument Editor. Advanced users may also create an instrument from scratch, in which case they will start with this step, then move on to step 2 above.

Learning About Sampler Instruments

A sampler instrument is the file type that is loaded into the EXS24 mkll for playing. A sampler instrument instructs the EXS24 mkll which samples (audio files) to load, and how to organize them into zones and groups. The EXS24 mkll allows you to play and record the loaded sampler instrument like you would any other software instrument.

You load sampler instruments with the Sampler Instruments menu of the EXS24 mkll. When you select a sampler instrument, the associated audio files are automatically located on the hard disk (or disks), and are loaded into your computer's RAM.

Sampler instruments are distinct from plug-in settings, which are loaded and saved in the plug-in header. Plug-in settings reside above the sampler instruments in the file 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.



Plug-in settings store all parameter adjustments made in the Parameters window. These settings are *not* part of the sampler instrument being loaded.

Tip: This separation allows you to use sampler instruments like waveforms in a synthesizer. Create a plug-in setting and set its envelope, modulation, and other parameters as desired. Then use the Sampler Instruments menu to load various "waveforms" to create new "synthesizer" sounds.

You can, however, choose to store the current Parameters window settings into a sampler instrument (see "Working With Sampler Instrument Settings" on page 396 for further information). This overrides the settings currently saved in the sampler instrument.

The EXS24 mkll is compatible with all audio file formats supported by Logic Pro: AIFF, WAV, SDII, CAF. Each audio file is loaded into the EXS24 mkll as a separate sample. Each audio file is then automatically assigned a *zone* in the EXS24 mkll Instrument Editor. These zones can then be edited and organized into sampler instruments. See "Editing Zones and Groups" for more information on using audio files in zones.

A point to note is that the actual audio files themselves are not contained within a sampler instrument. The sampler instrument only stores information about an audio file's name, its parameter settings, and its location on the hard disk. If you delete or rename an audio file, any sampler instrument that makes use of this file will not be able to find it, so take care when handling audio files. You can, however, move audio files to another location on your system. The EXS24 mkll will still be able to find such files when sampler instruments are loaded.

Loading Sampler Instruments

The EXS24 mkll ships with a ready-to-play sampler instrument library.

To load an instrument:

1 Click the Sampler instrument field directly above the Cutoff knob in the EXS24 mkll Parameters window. This opens the Sampler Instruments menu.



2 Browse to, and select, the desired sampler instrument.

In order to be visible within the EXS24 mkll Sampler Instruments menu, instruments must be stored in the Sampler Instruments sub-folder of any of the following folders:

- ~/Library/Application Support/Logic: User-defined or edited instruments are stored here.
- /Library/Application Support/Logic: Factory-supplied EXS instruments are installed here
- /Applications/Logic 6 Series: EXS instruments of the Logic 6 Series are stored here.
- .../ProjectName: Logic Pro also searches for EXS Instruments in the project folder.

Note: You can store your sampler instruments in any folder on any of your computer's hard drives. If you create an alias pointing to this folder within the Sampler Instruments folder (located in the folders listed above), the instruments are displayed in the Sampler Instruments menu.

To browse to the next or previous instrument of your sampler instrument library, do one of the following:

Click the plus or minus button to the left and right of the Sampler Instruments menu.



 Choose Next Instrument or Previous Instrument in the Sampler Instruments menu (or use the Next EXS Instrument and Previous EXS Instrument key commands).

If the EXS24 mkll is the topped window, you can also use the following two key commands:

- Next Channel Strip or Plug-In Setting or EXS Instrument
- Previous Channel Strip or Plug-In Setting or EXS Instrument
- Tip: You can also browse through your sampler instruments using your MIDI keyboard. The Sampler Preferences window offers two preferences for Previous Instrument and Next Instrument. These let you choose a MIDI event, such as a MIDI note, control change, program change, and so on, to select the previous or next sampler instrument in the sampler instrument list. See "Setting Sampler Preferences" for more information.

You can manually load sampler instruments that are not shown in the Sampler Instruments menu, by using the Instrument menu in the Instrument Editor.

To load sampler instruments from other locations:

1 Open the Instrument Editor by clicking the Edit button in the upper right corner of the Parameters window.



2 Choose Instrument > Open, then locate the desired instrument in the ensuing file selector box.

It is strongly recommend that you copy any EXS sampler instruments, along with their associated audio files, to your hard drives. This way, you always have direct, immediate access to your sampler instruments without searching for and inserting CD-ROMs or DVDs. This also allows you to organize your sampler instruments in accordance with your needs.

To copy sampler instruments to your hard drives:

- 1 Copy the sampler instrument file into the ~/Library/Application Support/Logic/Sampler Instruments folder.
- 2 Copy the associated samples into a folder named Samples in the same directory as the Sampler Instruments folder.

Working With Sampler Instrument Settings

Do not mistake plug-in settings, loaded and saved in the plug-in header, with sampler instruments. The plug-in settings that can be stored and recalled in the EXS24 mkll Parameters window are *not* part of the sampler instrument being loaded. Those settings need to be saved and loaded like standard plug-in settings in the plug-in header.

The EXS24 mkll does, however, offer a few commands in the Options menu of the Parameters window which let you set and recall plug-ins settings as part of your sampler instrument. These are:

- Recall default EXS24 settings: Recalls a neutral setting for all parameters in the Parameters window. This gives you a "clean slate" when adjusting the parameters of your sampler instrument.
- Recall settings from instrument: Recalls the original parameter settings of the loaded sampler instrument. This parameter is extremely useful if you've been over zealous with your tweaking, and want to return to the original parameter settings of the sampler instrument.
- Save settings to instrument: Stores the current settings of the Parameters window into the sampler instrument file. When the instrument is reloaded, these settings are restored.
- Delete settings from instrument: Removes the stored settings from the instrument.

Managing Sampler Instruments

As your sample library grows, the list of sampler instruments will also expand. To aid you in keeping the list of sampler instruments manageable, the EXS24 mkll features a simple, but sophisticated file management method.

To organize your sampler instruments into a preferred hierarchy:

- 1 Create a folder in the Finder—Basses for example—and drag it into the desired Sampler Instruments folder.
- 2 Drag the desired EXS24 mkll sampler instruments into this newly created folder.



Their menu structure will be reflected when clicking on the EXS24 Sampler Instruments menu.

Note: You will need to choose the Refresh Menu command in the Sampler Instruments menu 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 (that contain EXS instrument files) outside the Sampler Instruments folders 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 another location.

Backing Up Instruments

You can use the "Backup audio files of all used and active instruments of current project" key command to copy the audio and sampler instrument files of all currently active sampler instruments in the project to a file location of your choice. Folders for the audio files associated with these sampler instruments are created in the target location.

This makes it easy to keep all of your sampler instruments and audio samples in one place, and ensures that every project folder contains all the necessary sampler instruments and audio files you will need, even if you don't currently have access to your sampler instrument library.

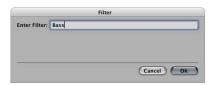
Tip: You can also do this by setting up your project to copy EXS24 sampler instruments and samples into the project folder. For more information, see the Logic Pro 8 User Manual.

Searching for Sampler Instruments

In order to minimize the number of sampler instruments displayed in the Sampler Instruments menu, you can make use of the Find function. This will limit the Sampler Instruments menu to only display sampler instrument names that contain the search word.

To search for sampler instruments:

- 1 Click the Sampler Instrument field directly above the Cutoff knob in the EXS24 mkll Parameters window, then choose Find in the Sampler Instruments menu.
- 2 In the Filter window, type in the character string (search term) to search for.



To disable the Search Filter:

• Choose Clear Find in the Sampler Instruments menu.

The full Sampler Instruments menu is displayed, but the actual search term typed into the Filter window is not cleared. You can return to the limited menu by choosing the Enable Find setting in the Sampler Instruments menu. This allows you to switch between the two without re-typing the search term.

If you wish to use a different character string, choose the Find command a second time and type in the desired search term.

Importing Sampler Instruments

The EXS24 mkll is compatible with the AKAI S1000 and S3000, SampleCell, ReCycle, Gigasampler, DLS, and SoundFont2 sample formats, as well as the Vienna Library.

Importing SoundFont2, SampleCell, DLS, and Gigasampler Files

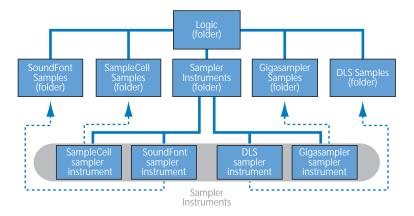
The EXS24 mkll automatically recognizes SoundFont2, SampleCell, DLS, and Gigasampler files placed inside the Sampler Instruments folder and converts them into sampler instruments.

To import SoundFont2, SampleCell, DLS, or Gigasampler files into the EXS24 mkll:

- 1 Copy or move your SoundFont2, SampleCell, DLS, or Gigasampler files into the ~/ Library/Application Support/Logic/Sampler Instruments folder.
- 2 Choose the SoundFont2, SampleCell, DLS, or Gigasampler file in the EXS24 mkll Sampler Instruments menu.

The EXS24 mkll automatically converts the SoundFont2, SampleCell, DLS, or Gigasampler file into an EXS sampler instrument:

- An EXS instrument file will be created in the Sampler Instruments folder which contains the file in its original format.
- The raw samples associated with the sampler instrument will be placed in one of the following folders, depending on the format being converted:
 - ~/Library/Application Support/Logic/SoundFont Samples
 - ~/Library/Application Support/Logic/SampleCell Samples
 - ~/Library/Application Support/Logic/Gigasampler Samples
 - ~/Library/Application Support/Logic/DLS Samples

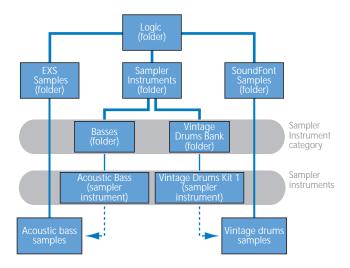


The procedure outlined above can also be used to import SoundFont2 and SampleCell *Bank* files, which contain multiple sounds, in addition to single instrument files. If you load a SoundFont2 or SampleCell Bank into the EXS24 mkll, it creates a Bank and a Samples folder, named after the SoundFont2/SampleCell Bank file. The word Bank or Samples is appended to each folder name.

All sounds contained in the bank will automatically have an EXS sampler instrument file created, and placed into the new *Bank* folder. The EXS24 mkll Sampler Instruments menu automatically updates to reflect the new folder hierarchy. All samples associated with the Bank automatically have a Samples folder created inside the SoundFont/ SampleCell Samples folder.

For example, if you load a SoundFont2 bank file named Vintage Drums, which contains over 50 individual drum kits from several different vintage drum machines into the EXS24 mkll:

- A new folder named Vintage Drums Bank will be created in the Sampler Instruments folder inside the ~/Library/Application Support/Logic folder.
- A second folder named Vintage Drums Samples will be created in the SoundFont Samples folder inside the ~/Library/Application Support/Logic folder.
- The Sampler Instruments 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, SampleCell, or Gigasampler 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 make sure that these instruments are displayed in the Sampler Instruments menu, you must create an alias pointing to this folder within the ~/Library/Application Support/Logic/Sampler Instruments folder.

Converting ReCycle Files to EXS Instruments

ReCycle, a sample editing program from Propellerhead software, separates sample material into small segments (called *slices*) based on the waveform peaks, called transients, in the audio file. This way, ReCycle is able to split an audio file into musically relevant slices. ReCycle can generate a number of file types which can be read by Logic Pro and the EXS24 mkll.

The EXS24 mkll supports the following ReCycle file types:

- Old ReCycle file: These files have the suffix .rcy. The abbreviation for this file type is RCSO. These files are not commonly used these days.
- Old ReCycle export file: These files have the suffix .rex. The abbreviation for this file type is REX. Older sample libraries that support REX files are often in this format.
- ReCycle 2.0 file: These files have the suffix .rx2. The abbreviation for this file type is REX2. These files are used extensively by Propellerhead Reason, and many popular sample libraries include REX2 format files.

Generating a Zone for Each Slice

The Extract MIDI Region and Make New Instrument command creates a new EXS24 instrument from a ReCycle file, and generates an independent zone for each slice.

To create a new EXS instrument and assign each slice to a zone:

- 1 Choose Instrument > ReCycle Convert > Extract MIDI Region and Make New Instrument in the Instrument Editor.
- 2 Browse to, and select the desired ReCycle file in the file selector, then click Open.
- 3 Enter a velocity factor in the Create MIDI Region window.



The velocity factor determines how the loudness of each slice of the imported ReCycle file affects the velocity values of the MIDI note generated to trigger it.

- If a positive value (up to 100) is entered, louder slices will generate MIDI notes with higher velocity values.
- The use of negative values on louder slices will result in the generation of lower MIDI note velocities.

4 Click OK.

The EXS24 mkll generates a zone for each slice of the imported ReCycle file, and assigns these zones to one group (see "Editing Zones and Groups" for more information on zones and groups). The new EXS instrument will be assigned the name of the ReCycle loop. Should an EXS instrument of that name already exist, a # sign and a number will be appended. In other words, if you import a ReCycle file named Tricky Backbeat but Tricky Backbeat already exists as a sampler instrument, the imported instrument would be named Tricky Backbeat#2, ensuring that the filename is unique within the Sampler Instruments folder.

In addition, a MIDI region is generated on the currently selected track, at the current project position (rounded to whole bars). This MIDI region is used to trigger the imported slices at the timing defined by the ReCycle file. You can generate new MIDI regions at any time from the imported EXS instrument (see "Generating a MIDI Region From a ReCycle Instrument" on page 403), so feel free to modify or delete it.

The Extract MIDI Region and Add Samples to Current Instrument command allows you to add the slices of a ReCycle loop to any EXS instrument currently opened in the Instrument Editor. This allows you to use several different ReCycle loops in a single sampler instrument.

Assigning the Complete ReCycle Loop to a Zone

The Instrument > ReCycle Convert > Slice Loop and Make New Instrument command creates an EXS instrument from a ReCycle loop, in which each zone plays back the ReCycle loop to its very end (at the current project tempo), starting with the slice points originally assigned to the respective zones. In other words, the lowest zone will play back the entire loop, and the highest zone will only play the last slice of the loop.

This kind of loop trigger techniques allow for old school drum'n'bass style triggering, in which the sample loop start point is determined by playing the respective notes on the keyboard.

The Instrument > ReCycle Convert > Slice Loop and Add Samples to the Current Instrument command adds the zones from the sliced loop to the currently active sampler instrument.

Pasting Loops From the Clipboard

The Edit > Paste ReCycle Loop as New Instrument command creates an EXS instrument from a ReCycle loop that has been copied into the Clipboard via ReCycle's Copy Loop facility.

The instrument creation is identical to the Extract MIDI Region and Make New Instrument command.

The Edit > Paste ReCycle Loop to Current Instrument command adds the zones to the currently active sampler instrument.

Generating a MIDI Region From a ReCycle Instrument

You can generate a MIDI region from imported ReCycle files in EXS instruments, triggering the imported slices at the timing defined by the ReCycle files.

To generate a new MIDI region from a ReCycle instrument:

Choose Instrument > ReCycle Convert > Extract Region(s) from ReCycle Instrument.

The MIDI regions are created on the currently selected track, at the current project position (rounded to bars). A single MIDI region is generated for each imported ReCycle loop in the currently open instrument. This function will also ask for a velocity factor (see above).

Converting AKAI Files

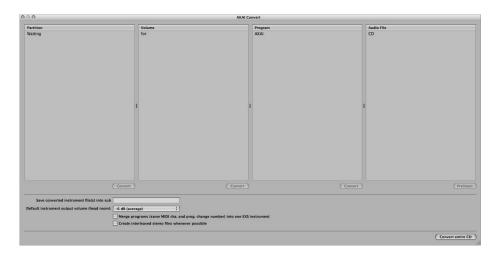
The EXS24 mkll can import samples in the AKAI S1000 and S3000 sample formats. The AKAI Convert function can be used to import:

- An entire AKAI format CD ROM
- An AKAI Partition
- · An AKAI Volume
- An AKAI Program
- An individual audio file (sample)

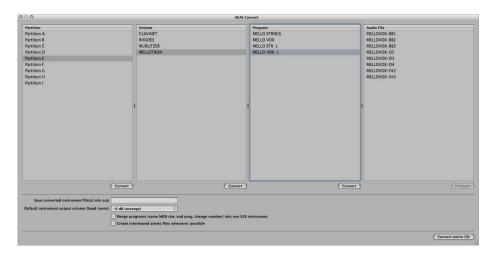
To convert AKAI files:

1 Click the Options button in the Parameters window, then choose AKAI Convert from the menu.

This will launch the AKAI Convert window, with the text Waiting for AKAI CD spread across the four columns.



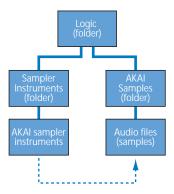
- Insert an AKAI format sample disc into your CD-ROM drive.
 The display updates to show the contents of the CD-ROM. The Partition column will display information, with Partition A, Partition B (and so on) entries listed.
- 3 To view the contents of the Partitions, click once on the appropriate entry 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).



You can use the Prelisten button below the Audio File column to individually audition AKAI audio files before deciding whether or not to import them.

- 5 If desired, set any of the additional AKAI Convert parameters at the bottom of the window (see "Additional AKAI Convert Parameters" below).
- 6 Once you have made your selection of Partition, Volume, or Program, click on the Convert button beneath the appropriate column. 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 Convert window.

The selected Partition, Volume, or Program will be imported, along with all associated audio files.



- Any audio files imported will be stored within a folder which matches the name of the Volume. This folder is created within the ~/Library/Application Support/Logic/ AKAI Samples folder.
- The sampler instruments created by the import procedure match the Program names. They are placed inside the ~/Library/Application Support/Logic/Sampler Instruments folder, or the sub-folder determined by the "Save converted instrument file(s) into sub folder" parameter. 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.

The Sampler Instruments menu displays the converted instruments as follows:



Additional AKAI Convert Parameters

Within the AKAI Convert window, you will find the additional parameters listed below.

Save converted instrument file(s) into sub folder

You can use this when importing an entire CD. This creates a folder name which reflects the CD-ROM's name. Alternately, you may want save your converted instruments into a sub-folder based on a category, such as Strings. This way, if your AKAI CD consists of string samples, any imported Programs or Volumes will be added to the Strings sub-folder.

To enter a name for a sub-folder in this parameter field, click once with the mouse and type in the desired name, followed by pressing Return. All imported Volumes and Programs will automatically be added to this folder.

Note: If an existing name is used, the imported sampler instrument will be *added* to the folder. It will *not* create a new folder of that name.

Default instrument output volume (head room)

Sustained pad sounds and polyphonic instruments in AKAI format often tend to have a higher output than a drum groove, for example. This can result in the output levels of some converted AKAI instruments being much higher than the rest of your EXS24 mkll sampler instrument library—occasionally, converted programs may be so loud that they clip. Set this parameter to the desired amount, which will limit the headroom (output level) of the EXS24 mkll parameter settings for each converted AKAI program.

The value you should choose for a given AKAI CD is a matter of trial and error, but here are some suggestions:

- For drum CDs, start with either no change (0 dB) or a headroom value of –3 dB.
- For piano, string, or pad CDs, you might try a headroom value of –9 dB.
- For very loud programs, such as layered analog synthesizer instruments, you might even want to try –12 dB.
- In cases where you're not sure of which headroom value to select, start with the average setting of –6 dB.

Merge programs (same MIDI cha. and prog. change number) into one EXS instrument

Many CD-ROMs created for AKAI samplers feature programs that contain single velocity layers for an instrument. AKAI samplers require that an entire volume, or all necessary single programs are loaded, in order to play all velocity layers. All of these single programs are automatically assigned to the same MIDI channel, and also respond to the same MIDI program change number.

The EXS24 mkll AKAI conversion intelligently checks for these settings, and builds a single EXS sampler instrument out of multiple single programs. In general, when importing samples of this type, this option should be switched 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 switched off.

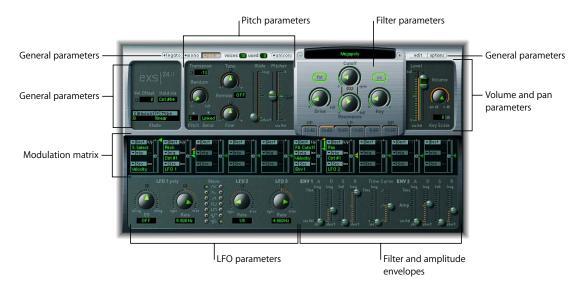
Create interleaved stereo files whenever possible

This option should always be left enabled, as interleaved files offer better performance within the EXS24 mkll. When converting AKAI format samples, 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.

Parameters Window

The EXS24 mkll Parameters window contains settings that determine how the EXS24 mkll processes the entire loaded sampler instrument.



It contains the following parameter groups:

- General parameters: Includes parameters for selecting and configuring sampler instruments, how many voices the EXS24 mkll will play at a time, crossfades, and configuring samples as modulation destinations.
- Pitch parameters: Let you adjust the tuning, transposition, pitch bend, and the like.
- Filter parameters: Allows you to activate and configure parameters controlling the resonance, slope, drive, and envelope of the EXS24 mkll filter.
- *Volume and Pan parameters:* Adjust the volume, volume envelope, and panning of the loaded sampler instrument.
- LFO parameters: Adjust any of the three available LFOs, which you can use as modulators.
- Modulation matrix: Lets you set up to ten complex modulation routings which allow you to select a modulation source parameter to modulate a destination parameter; you can even modulate the modulation source using the via parameter.

General Parameters

The following section describes the general parameters of the EXS24 mkll.

Legato/Mono/Poly Buttons



These buttons determine the number of voices used by the EXS24 mkll (in other words, how many notes can be played simultaneously):

- When Poly is selected, the maximum number of voices is set via the numeric field alongside the Poly button. To change the value, drag up or down to increase or decrease polyphony. In general, source instruments that let you play chords, such as a piano or guitar, are polyphonic, and lend themselves to this mode.
- When Mono or Legato is selected, the EXS24 mkll is monophonic, and uses only one
 voice. In general, instrument sources that only play one note at a time, such as a flute
 or Moog monophonic synthesizer, lend themselves to Mono or Legato mode.
 - In Legato mode, Glide is only active on tied notes. Envelopes are not retriggered when tied notes are played (in other words, playing a series of tied notes results in only a single envelope trigger). For more information on the Glide function, see "Glide" on page 415
 - In Mono mode, Glide is always active and the envelopes are retriggered by every note played.

Voices

This parameter determines the number of voices (polyphony) that the EXS24 mkll 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

In Unison mode, multiple EXS24 mkll voices are played when a 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.

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 Instruments Menu

You can click in the Sampler Instruments menu to load a sampler instrument into the EXS24 mkll. See "Loading Sampler Instruments" for complete information on the Sampler Instruments menu.

Edit Button

This button to the right of the Sampler Instruments menu opens the currently loaded sampler instrument in the EXS24 mkll Instrument Editor. If none is loaded, the Instrument Editor will open, allowing the creation of a new sampler instrument. See "The Instrument Editor" for more information on the Instrument Editor.



Options Button

Clicking the Options button opens a menu that offers the following options:

- Recall default EXS24 settings: Recalls a neutral setting for all parameters in the Parameters window. This gives you a clean slate for adjusting parameters for your sampler instrument.
- Recall settings from instrument: Recalls the original parameter settings of the loaded sampler instrument. This parameter is extremely useful if you've been over zealous with your tweaking and want to return to the original parameter settings of the sampler instrument.
- Save settings to instrument: Stores the current settings of the parameters window into the instrument file. When the instrument is reloaded, these settings are restored in the parameters window.
- Delete settings from instrument: Removes the stored settings from the instrument.
- Rename instrument: Opens a file dialog box which allows you to rename the current instrument. This will overwrite the existing instrument name.
- Save instrument as: Allows you to store the currently opened sampler instrument under a different name. When invoked, a file dialog box will open.
- *Delete instrument:* Deletes the opened sampler instrument.

- (Recall default EXS24 mkl settings): For sampler instruments that were created using the older EXS24. This will recall the parameter settings for the former version of the selected instrument, especially the former modulation paths (see "EXS24 mkl Modulation Paths" on page 423). For sampler instruments that were created in the EXS24 mkll, this parameter is of little use.
- 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 dimmed.
- AKAI Convert: Launches the AKAI Convert window (see "Converting AKAI Files" on page 403).
- SoundFont Convert, SampleCell Convert, DLS Convert, Giga Convert: Each of these commands will launch a dialog with instructions on performing these conversions.
- *Preferences*: Opens the EXS24 mkll preferences (see "Setting Sampler Preferences" on page 446).
- Virtual Memory: Opens a settings window, used to configure the EXS24 mkll's virtual memory functions. Virtual memory allows samples of almost unlimited length to be played back, using streams (of audio data) that are fed directly from the hard disk in real time. See "Configuring Virtual Memory" for more information.

Vel Offset

The Vel(ocity) Offset parameter—found towards the top left of the EXS24 mkll—offsets the incoming MIDI note velocity value by ± 127 . This limits or expands the dynamic response of the EXS24 mkll to incoming note events.

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 MIDI controller number 64 (the MIDI standard controller number for hold functions). 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) parameters

If you are familiar with the concept of layering sample zones by Velocity Range, the Xfade parameters allow you to crossfade between layered sample zones with adjacent Velocity Range settings. If you are not familiar with this idea, here's a short explanation:

When assigning a sample to a zone, you can set the lowest and highest velocity that will trigger that zone. The range in between these values is the zone's Velocity Range. You can layer your zones by having different zones triggered by the same key, but different velocities.

For example, if you have a sample of a snare drum hit lightly and a little off center assigned to Zone #1, and a sample of a snare drum hit hard in the center of the drum heading Zone #2. If Zone #1 has a Velocity Range of 24 to 90, and Zone #2 has a Velocity Range of 91 to 127, if both zones are assigned to the MIDI note A#2, the zones can be said to be layered on A#2.

In the example above, the high Velocity Range parameter of Zone #1 and the low Velocity Range parameter of Zone #2 are adjacent, and at this point, there will be a rather abrupt change from the audio sample used in Zone #1 to the audio sample used in Zone #2. To make this less abrupt, the crossfade parameters lets you smoothly fade between both samples. When you have distinctly different audio samples in adjacent zones, you will find crossfading very helpful in creating realistic sampler instruments.



Crossfades are controlled by two parameters: Amount and Type.

Amount refers to the range of velocity values in which the crossfade takes place. In other words, the crossfade is applied symmetrically around each layered zone, with the crossfade amount determining the overlap of the two zones. The Velocity Range setting of all zones will be expanded by this value, with the crossfade taking place in the expanded area. When the Amount parameter is set to 0, the EXS24 mkll will not fade smoothly between zones but simply switch from one zone to another.

Also, as you will read in the "Modulation Matrix" section later, you can set other modulation sources, for example the modulation wheel of your MIDI controller, to modulate the Amount parameter. In this case, the Amount parameter still functions the same way, but the crossfade will not be triggered by velocity, but by the chosen modulator.

In the Type menu, you can choose between three different choices of fade curves for the velocity crossfade:

- dB lin (dB linear): A logarithmic curve, so that the crossfade sounds even on both sides.
- *linear (gain linear):* A convex crossfade curve, so that the crossfade sounds as if nothing is happening at first, but then with a rapid volume fade towards the end.
- Eq. Pow (equal power): A nonlinear curve in which the level rises faster in the beginning but rises back to normal more slowly. This is useful if your crossfade seems to drop in volume in the middle.

Pitch Parameters

These parameters let you adjust the tuning and transposition of the loaded sampler instrument.



Tune

Use this knob to raise or lower the pitch of the loaded samples in semitone increments. The middle position of the knob (which can be set by clicking the small 0 button) leaves the pitch unaltered.

Transpose

This parameter also allows you to transpose the EXS2 mkll in semitone increments. In contrast to the Tune parameter, however, 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.

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

This parameter lets you tune the loaded sampler instrument in cent increments. You can use this to correct for samples that are slightly out of tune, or to create a thick chorus-like effect.

Pitch Bend Up

This parameter determines the upper limit of pitch bending (in semitones) that can be introduced by moving the pitch bend wheel to its maximum position using this parameter. Pitch Bend Up has a range of 0 semitones (the pitch bend wheel in its maximum position will not increase the pitch at all) to 12 semitones (the pitch bend wheel in its maximum position will increase the pitch a full octave).

Pitch Bend Down

This determines the lower limit of pitch bending (in semitones) that can be introduced by moving the pitch bend wheel to its minimum position. Pitch Bend Down has a range of 0 semitones (the pitch bend wheel in its minimum position will not lower the pitch at all) to 36 semitones (the pitch bend wheel in its minimum position will lower the pitch three full octaves). When Linked is selected, the Pitch Bend Up value is used.

Remote

The Remote parameter allows you to easily pitch complete EXS24 mkll 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. Once set, all keys in a range of ± 1 octave around this key will now pitch the entire instrument instead of triggering the 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 will no longer 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 (called *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

These parameters control the EXS24 mkll filter section. You can configure the type of filter, its resonance, cutoff frequency, drive, and amount of key following, and adjust its ADSR (Attack Decay Sustain Release) envelope.



Filter On/Off Button

This button switches the filter section on or off. Please note that the knobs and buttons in the silver panel area and the filter envelope are only active when the filter is turned on. When the filter section is turned off, the EXS24 mkll is far less CPU-intensive.

Lowpass (LP)

Click any of the four buttons under the LP label to engage the desired *lowpass* slope. The orange line above the button indicates the selected lowpass slope. You can select among four different settings for the lowpass filter's 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 ultradeep 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.

Fat (Fatness)

Click the Fat button to engage the fatness feature. The fatness feature 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 highpass or bandpass filters.

Highpass (HP)

Click the button under the HP label to engage the highpass filter. 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)

Click the button under the BP label to engage the bandpass filter. 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, which introduces additional harmonics.

Cutoff

Adjust this knob to set the cutoff frequency of the 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



Click and drag on the chain symbol located between the Cutoff and the Resonance knobs to control both parameters simultaneously: vertical mouse movements alter Cutoff, and horizontal mouse movements affect Resonance values. This way you can experiment with different filter cutoff and resonances while maintaining the relationship between the two parameters.

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.

Filter Envelope

In the lower section of the EXS24 mkll interface, ENV1 is the ADSR envelope generator for the filter. This envelope allows you to control the filter over time. It offers Attack, Decay, Sustain, and Release parameters.



Drag the A, D, S, and R sliders to configure the appropriate parameter. 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 inbetween the two slider segments, both can be moved simultaneously.

Time Curve

The sliders apply to both the filter (ENV1) and volume (ENV2) envelopes; the left slider, known as time via key, can be used to scale (lengthen or shorten) the time intervals of both envelopes. Note position C3 is the center point; time intervals for all zones assigned to keys above C3 can be reduced in length with this slider. All time intervals for zones assigned to keys below C3 can be lengthened. The (Attack) Curve slider determines the shape of the envelope attack time.

Volume and Pan Parameters

The following section describes the Volume and Pan parameters of the EXS24 mkll.

Level via Vel

This slider 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 Volume knob is the main volume parameter for the EXS24. Move this knob to find the right balance between avoiding distortion and getting the best (highest) resolution in the channel fader and the Level via Vel slider.



Key Scale

This parameter modulates the sound's level by note number (position on the keyboard). Negative values increase the level of lower notes. Positive values increase the level of higher notes.

Amp Envelope (ENV 2)

This is an ADSR envelope generator for controlling the sound's level over time. It offers Attack, Decay, Sustain, and Release parameters.



Drag the A, D, S, and R sliders to configure the appropriate parameter. 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 inbetween the two slider segments, both can be moved simultaneously.

LFO Parameters

The EXS24 mkll includes three LFOs (low frequency oscillators) which can be used as modulation sources. This section explains their parameters.



LFO 1 EG

This knob allows LFO 1 to be faded out (when the knob is pointing inside the *Decay* area) or faded in (when the knob is pointing in the *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).

Waveform for LFO 1 and LFO 2

These radio buttons allow you to select the waveform type used by LFO 1 and LFO 2, respectively. A selection of Triangle, falling and rising Sawtooth, Square up and Square down, a random stepped waveform, and a smoothed random waveform is available for each LFO.

LFO 1 is a polyphonic LFO with key synchronization. This means that when LFO 1 is used, each voice of the EXS24 has its own discrete LFO. When a note is played, the LFO corresponding to that voice starts its cycle. This scheme means that the LFO cycles of each voice played are not synchronous, and operate independently of each other. This opens up a range of modulation possibilities. As an example—the LFO of one voice could generate the maximum modulation value, while the LFO assigned to another voice could output its minimum value. This extremely flexible approach can result in some very lively modulations.

In contrast, LFO 2 is a monophonic LFO without key synchronization. This means that LFO 2 runs continuously, and is not restarted by the triggering of a new note. All voices are modulated by the sole LFO, so the degree of modulation at any given time is the same for all voices. This results in a rather synthetic-sounding modulation.

Use these different characteristics to tailor the sound to your needs.

LFO 2 Rate

The frequency rate of LFO 2 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).

LFO 3 Rate

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

Modulation Matrix



The modulation matrix is the dark horizontal band that spans the center of the EXS24 mkll interface. It consists of ten modulation paths, each linking a modulation *source* (the element that will do the modulating) 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 (also known as modulation intensity) via another modulation source (labeled as *via*).

To create a new modulation path:

1 Choose the modulation destination in the Dest(ination) menu.



2 Choose the desired modulation source in the Src (Source) menu.

3 Set the modulation depth with the green triangular fader on the right side of each modulation path.



In the example above, 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. The via modulation source doesn't directly modulate the destination, but instead it modulates the source—in essence, modulating the modulator.

When you select a modulator for the via slot, 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 the example below, the key number of the MIDI keyboard (Key) determines how strongly channel pressure controls the Speed of LFO1. Experienced users can 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.

The following example shows an inverted 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 the example above, both modulation sources—Pressure and Key—are disconnected from the LFO1 Speed modulation destination. Clicking the b/p button a second time reconnects the modulation path, restoring the old modulation depth settings.

Second Order Modulations

The EXS24 also allows the use of second order modulation destinations:

- The same source can be used as often as desired to control different destinations.
- The same *destination* can be controlled by different *sources*. The different input values are accumulated.

EXS24 mkl Modulation Paths

Many of the hard-wired modulation paths that were available as sliders on the original EXS24 (mkl) are part of the modulation matrix. In order to reconstitute the modulation slider configuration of the mkl version, click on the "options" button in the upper-right corner and choose (Recall default EXS24 mkl settings) from the pop-up menu. This will load the mkl modulation paths into the modulation matrix, as follows:

- Velocity to Sample Select
- LFO 1 to Pitch via ModWheel (= Ctrl#1)
- Velocity to Sample Start (inv)

- LEO 2 to Filter Cutoff via ModWheel
- Velocity to Filter Cutoff
- Envelope 1 to Filter Cutoff via Velocity
- LFO 2 to Pan via ModWheel

You can, of course, freely alter the settings of these modulation paths. To exchange modulation sources with sources that were not available in EXS24 mkl, for example (see the complete list of sources and destinations below).

Note: For technical reasons, the settings of the modulation matrix can not translate backwards to the EXS24 mkl.

Available Modulation Sources and Destinations

The following modulation sources and destinations are available in the EXS24 mkll.

Modulation Sources:

- Side Chain (level)
- Maximum
- Fnv 1
- Env 2 (Amp)
- LFO 1, LFO2, and LFO 3
- Release Velocity
- Pressure
- · Pitch Bend
- Key
- Velocity
- Control Nr. 1 to Control Nr. 120

Modulation Destinations:

- Sample Select
- Sample Start
- Glide Time
- Pitch
- Filter Drive, Filter Cutoff, and Filter Resonance
- Volume
- Pan
- Relative Volume
- LFO 1 Dcy./Dly (LFO 1 Decay/Delay)
- LFO 1 Speed, LFO 2 Speed, LFO 3 Speed
- Env 1 Attack, Env 1 Decay, Env 1 Release
- Env 2 Attack, Env 2 Decay, Env 2 Release

- Time
- Hold

Note: Controllers 7 and 10 are marked as (not available). Logic Pro uses these controllers for volume and pan automation for the audio channel strips. 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.

Sample Select

This modulation destination deserves a more thorough explanation. By default, Sample Select is controlled by velocity through the default Velocity to Sample Select modulation path. This means that the received note velocity value determines which of the layered zones with different Velocity Range settings is heard.

However, you aren't limited to only using velocity to choose what sample is played. By assigning modulation sources other than velocity, for example the modulation wheel, to the Sample Select modulation destination, you can determine which sample plays using something other than velocity. You can even use multiple sources, such as both velocity and the modulation wheel, using the modulation matrix.

When using multiple modulation sources (or any source other than velocity or key), however, please keep in mind that these can 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. This means that CPU usage will increase due to all velocity layers being active in memory, even if you don't hear them.

If you choose a continuous controller such as the modulation wheel to modulate the Sample Select destination, you can step through the velocity layers during playback. In this case, the crossfade (XFade) parameters are very important for creating smooth transitions between velocity split points.

The Instrument Editor

You can use the Instrument Editor of the EXS24 mkll to create and edit sampler instruments.

A sampler instrument consists of zones and groups:

- A zone is a location into which a single sample (or audio file, if you prefer this term) can be loaded from hard disk or CD ROM.
- Zones can be assigned to *groups*, which offer several parameters for simultaneous control over all assigned zones. You can define as many groups as desired. You can edit the group parameters in Group view.

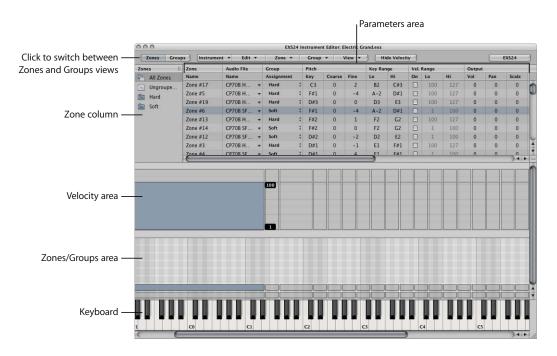
To open the Instrument Editor:

 Click the Edit button in the upper right corner of the EXS24 mkll Parameters window (or use the Open EXS24 Instrument Editor key command).



The Instrument Editor features two views: Zones view and Groups view. Zones view displays zones and their parameters, in the parameter area. In Zones view, the area above the keyboard becomes the Zones area, and displays bars for the created zones. Groups view shows groups and their parameters. In Groups view, the area above the keyboard becomes the Groups area, and displays bars for the created groups. The general items—menus, buttons, and so on—are displayed in both views.

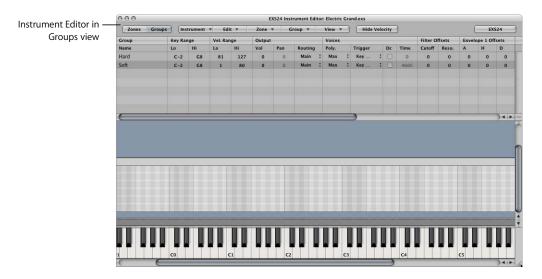
The screenshot below shows the Instrument Editor in Zones view.



• Zones column: Displays all zone groups of the instrument. The All Zones and Ungrouped Zones groups exist by default in every instrument. Click the desired group to display the associated zones in the Parameters area. You can also select and view multiple zone groups at once (but only if the All Zones group is not selected).

- *Parameters area:* Displays the parameters of the zone group chosen in the Zones column.
- *Velocity area*: Shows the velocity range of the selected zone.
- Zones/Groups area: Shows the zones or groups graphically placed above the keyboard.
- *Keyboard:* Use to trigger notes for the EXS24 mkll in the currently selected track. The keyboard also serves as a visual reference for the placement of the zones or groups in the Zones or Groups area.

Click the Groups button in the upper left corner to switch to Groups view. Click the Zones button in the upper left corner to switch to Zones view. You can also use the Toggle Zones/Groups View key command to switch between views.



The EXS24 button in the upper right of the Instrument Editor reopens a closed EXS24 mkll Parameters window. The button does not bring the Parameters window to the foreground if covered by other floating windows.

Creating Instruments, Zones, and Groups

You can add zones and groups to loaded instruments, or create a new instrument and fill it with zones and groups.

To create a new instrument:

Choose Instrument > New from the Instrument Editor menu.

Please see "Loading Sampler Instruments" on page 394 for information on loading existing sampler instruments.

Zones

A zone is a location into which a single sample (or audio file, if you prefer this term) can be loaded. The sample loaded into the zone is memory resident—it uses the RAM of your computer. A zone offers various parameters for controlling the playback of the sample. Each zone allows you to determine the range of notes over which the sample should be heard (Key Range), and the root key—the note at which the sample sounds at its original pitch. In addition, sample start, end, and loop points plus volume, and several other parameters can be adjusted within the zone. You can define as many zones as you wish. Each zone requires at least one EXS24 mkll voice when played.

To create a zone and assign a sample to it:

1 Choose Zone > New Zone (or use the New Zone key command).

A new zone entry appears in the Instrument Editor.

- 2 Do one of the following:
 - Click the arrow in the Audio File column, then choose Load Audio Sample from the pop-up menu (or you can use the Load Audio Sample key command).



- Double-click the empty area in the audio file column.
- 3 Browse to, and select the desired audio file in the file selection dialog.

If you activate the "Hide used Audio Files" option, files used in the currently loaded EXS instrument are dimmed.

- 4 Use the Play button to preview the audio file:
 - Click the Play button to loop playback of the currently selected sample file.
 - Clicking the button a second time stops playback.
 - You can audition multiple files by pressing Play once, and then stepping through the files.

The "Preview audio file in EXS instrument" option temporarily replaces the sample files in the currently selected zone. The zone is not directly triggered by activating this option, but it can be triggered via MIDI notes while the file selection dialog is open and different files are being selected. The selected sample can be heard as part of the zone, inclusive of all synthesizer processing (filters, modulation, and so on).

5 Once you find a sample you'd like to use, click Open to load it.

When loaded, the sample's name is displayed in the gray field.

Creating Zones via Drag and Drop

You can also create a new zone (and a new instrument, if none is currently displayed in the editor) by dragging a file onto one of the keys of the onscreen keyboard. The start key, end key, and root key are all set to the note that the file was dropped on. This drag and drop functionality works for audio files from the following sources: Browser, Audio Bin, and the Finder.

You can also create a new zone (and a new instrument, if none is currently displayed in the editor) by dragging a file directly in the zone area.



The root key for the zone is the key at which the sample is played at its recorded pitch. This information is written in the sample header. If there is no root key defined in the sample header, the C3 key is used by default.

Note: If you drag an audio file onto an existing zone, the file referenced by that zone is changed to the new (dropped) file. The mouse cursor changes to a replace arrow.



Dragging a single sample into the empty area below Ungrouped Zones creates a default zone and a default group, with the new default zone placed in the default group.

You can also load multiple samples in one operation. The Instrument Editor automatically creates new zones, and places the loaded samples into them.

To load multiple samples in one operation:

- 1 Choose Zone > Load Multiple Samples in the Instrument Editor (or use the Load Multiple Samples key command).
- 2 Browse to the desired location, then use the Add, Add All, Remove, or Remove All buttons to select the desired samples.
- 3 Click the Done button when you are finished.
- 4 Choose one of the three automatic mapping modes in the Load Multiple Samples dialog:



- "Auto map" by reading the root key from audio file: Uses the root key stored with the audio files, and places the samples (as zones) across the keyboard range. The number of keys that constitute a zone is intelligently determined by the placement of neighboring zones.
- "Drums" zone without range, root key from audio file: Uses the root key stored with the audio files. Each zone contains a single key on the keyboard determined by the root key information.
- Contiguous zones: Ignores all root key information of the audio files and places the samples on the keyboard in chromatic order. The "Zone width" field allows you to specify the width of the newly-generated zones. The Start Note field defines the start note of newly-generated zones.

You can also load multiple samples by dragging them into the Instrument Editor. Dragging multiple samples into a group folder assigns the samples to the respective group. Dragging multiple samples below the Ungrouped Zones area assigns the audio files to a new default group.

If you drag multiple files onto one of the keyboard keys, the Load Multiple Samples dialog does not include the Start Note field, as the start key, end key, and root key are all set to the note that the file was dropped on.

Groups

Imagine a drum kit has been created, with a number of different samples being used in several zones, mapped across the keyboard. In many musical circumstances, you might want to adjust the sound editing parameters of each of the samples independently—to alter the decay of the snare, or to use a different cutoff setting for the hi-hat samples, for example.

This scenario is where the EXS24 mkll's *groups* feature comes in. Groups allow for very flexible organization of samples. You can define an unlimited number of groups, and can assign each zone to one of these groups. In a drum set, for example, you could assign all kick drums to Group 1, all snares to Group 2, all hi-hats to Group 3, and so on.

Why might you want to do this?

A group makes it possible to define a velocity range for all assigned zones, allowing you to specify a velocity window in which the grouped zones should sound, as one example. Each group also features offset parameters for the amplitude envelope and filter settings made in the Parameters window.

It's also possible to play all zones without defining and assigning even a single group—in this case, the parameter settings affect all samples in all zones equally.

To create a new group:

 Choose Group > New Group in the Instrument Editor (or use the New Group key command).

A new group appears in the Zones column on the left-hand side of the Instrument Editor.



To assign a zone to a group, do one of the following:

- Select the group in the zone's Group menu.
- Select a zone in the EXS Instrument Editor, Finder, Audio Bin, or Browser and drag it into a group displayed in the Zones column.
- Drag a zone (or multiple selection of zones) out of one group into another group to change the group assignment to a new group.

- Drag a zone (or multiple selection of zones) out of one group to Ungrouped Zones to change the group assignment to unassigned.
- Drag an ungrouped zone (or multiple selection of zones) into the empty area below
 Ungrouped Zones to create a new group containing the dragged zone (or zones).
- Drag a zone (or multiple selection of zones) out of one group folder to the empty area below Ungrouped Zones to create a new group containing the dragged zone (or zones).
- **Tip:** Pressing Option while dragging zones to another group will copy the selected zones.

The Group parameters will now affect the sample in the zone.

To delete all groups that do not have a zone assignment:

• Choose Group > Delete Unused Groups in the Instrument Editor.

Editing Zones and Groups

Zones and groups have a number of parameters which you can edit to customize your sampler instrument. You edit the pitch, velocity range, panorama, looping parameters, and other functions of zones. You can also edit group parameters for velocity, output, and offset envelopes and filters, among other parameters. While some of the editing techniques, such as editing graphically, selection menu options, clicking on values to change them, and so on are similar, zones and groups have different parameters, and will be discussed separately.

General Editing Techniques

The Edit menu handles all basic sampler instrument editing operations, such as copying zones, undoing edit operations, and so on.

- *Undo:* Allows the most recent change to the sampler instrument to be undone.
- Redo: Undoes the last Undo command.
- Cut, Copy, Paste: The standard commands for cutting, copying, and pasting values. In
 addition to values you may also cut, copy, and paste selected zones and groups.
 When you copy groups in the Zones view, the selected groups and their associated
 zones are copied. The group assignments of the zones are retained. When you copy
 groups in Groups view, only the groups themselves are copied, not the associated
 zones.
- *Delete*: Deletes the currently selected zone or group.

Selecting Zones and Groups

There are a number of ways that you can select zones and groups for editing.

First of all, the Edit menu also contains commands specifically relating to the selection of EXS24 mkll zones and groups:

• Select All: Selects all zones and groups of the loaded sampler instrument.

• *Toggle Selection (also available as key command):* Toggles the selection between the currently selected zones or groups and all currently unselected zones or groups.

You can also click on zones and groups in the Parameters area:

- Clicking on the parameters of a single zone or group selects that zone or group.
- Shift-clicking multiple zones in the Zones view selects all the zones between the two clicked zones.
- Command-clicking multiple zones selects each clicked zone.

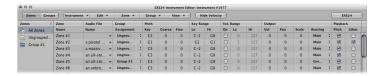
You can also use the Up Arrow key and Down Arrow key to select the previous and next zone or group.

Select Zone of Last Played Key

If you enable the Select Zone of Last Played Key menu item in the Zone menu, you can switch between zones by pressing a key on a connected MIDI keyboard. You can still select zones by clicking on them in the editor even when this feature is enabled.

Setting Zone Parameters

The zone parameters provide extensive control over each zone (sample) in your sampler instrument.



Zone Name

New zones are numbered consecutively. Double-click a zone's number to enter a name instead.

Audio File

This column displays a zone's audio file. Mousing over an audio file reveals a help tag with additional information about the audio file, such as format, bit depth, sample rate, and so on. If you press Command before the help tag displays, the help tag will also include the full file path of the audio file. Clicking the arrow button opens a shortcut menu that offers the following commands:

- Load Audio Sample: Opens a file selection box in which you can select an audio file. See "Zones" on page 428 for more information.
- Open in Sample Editor: This opens the selected sample in the Logic Pro Sample Editor (or the sample editor chosen in the Open External Sample Editor preference) for further editing.
- Reveal in Finder: Shows the (loaded) audio file in the Finder.

▶ Tip: Double-clicking the Audio File column—when an audio file is loaded—opens the audio file in the Sample Editor. When no audio file is loaded, the audio file selector opens.

Group

Shows the group assignment of a zone. See "Groups" for more information.

Pitch Parameters

Key allows you to determine the root note of the sample—in other words, the note at which the sample will sound at its original pitch. The Coarse and Fine fields allow coarse and fine-tuning of the sample in semitone and cent increments.

Key Range

The two key range parameters allow you to define a key range for the zone.

- Lo(w): Sets the lowest note at which the zone will sound
- *Hi(qh):* Sets the highest note at which the zone will sound.

Playing notes outside this range will not trigger the sample assigned to this zone.

Velocity Range

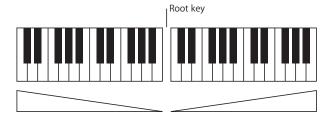
Activate the Velocity Range checkbox to define a velocity range for the zone. L(ow) sets the lowest velocity at which the zone will sound, and Hi(gh) sets the highest velocity at which the zone will sound. Playing notes outside this range will not trigger the sample assigned to this zone.

Output Parameters

Volume adjusts the volume of the zone.

Pan adjusts the pan position of the zone. This parameter only works when the EXS24 mkll is used in stereo.

Negative Scale values make notes lower than the note position defined by the root key sound louder than higher ones; positive values have the opposite effect. Use this parameter for balancing the volume of a sample across the selected key range.



The Routing parameter determines the outputs used by the zone. Choices include the main outputs, and paired channels 3 and 4, 5 and 6, 7 and 8, 9 and 10, or individual outputs 11 through to 16. This allows individual zones to be routed independently to aux channels in a multi-output EXS24 mkll instance.

Playback Parameters

To enable the sample to change pitch when the sample is triggered by different keys, switch on the Pitch checkbox. When disabled, the sample is always played at its original pitch, regardless of the note played.

Activating 1Shot causes the zone to ignore the length of notes used to trigger the sample—the sample is always played to the end. This option is useful for drum samples, where you often don't want the MIDI note length to affect sample playback.

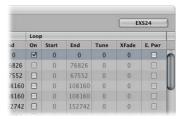
Enable the Rvrs (Reverse) checkbox to play the sample from the end to the beginning.

Sample Start and End

The Sample Start and End parameters set the sample's start and end points, respectively. Control-clicking the two values will open a shortcut menu, allowing you to open the sample in the Sample Editor of Logic Pro. In the Sample Editor, you can set the start and end points graphically. See "Editing Samples in the Sample Editor" below.

Loop Parameters

The EXS24 mkll can loop playback of either all or a portion of an audio sample, when you trigger a sample to play. In other words, the sample will loop when sustained MIDI notes are received.



- Loop On checkbox: Activate this checkbox and the rest of the Loop parameters become available for editing.
- Loop Start, Loop End: You can define discrete loop start and end points in these
 fields, allowing you to loop a portion of the audio file. Clicking on the small E button
 between the two values will open the audio file the Logic Pro Sample Editor, allowing
 you to set the loop start and end points graphically: Loop Start is represented by the
 LS marker and Loop End, by the LE marker. "Editing Samples in the Sample Editor"
 below.
- *Tune*: Allows you to change the tuning of the looped portion of the audio file in cent increments.

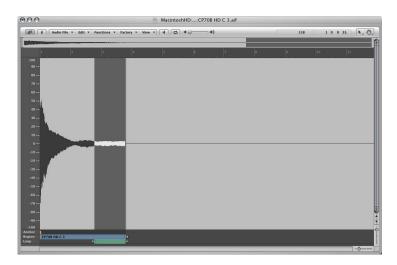
- Xfade (Crossfade): In a crossfaded loop, there is no hard cut between the loop end and loop start points. Rather, the loop end and start points are crossfaded for a smooth transition. This is especially convenient with samples that are hard to loop, and would normally produce clicks at the transition point—the join in the loop. The Crossfade field allows you to determine the time of the crossfade. The higher the value, the longer the crossfade, and the smoother the transition between the loop end and start points.
- E. Pwr (Equal Power) checkbox: Allows you to enable an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This will fade out/fade in the joined portions of a loop at an equal volume level.

Note: The perfect settings for the crossfade parameters depend on the sample material. A loop which cycles reasonably smoothly is the best starting point for a perfect crossfade loop, but a crossfaded loop does not always sound better. Just experiment a little with the parameters, and you'll soon find out how, when, and where they work best.

Editing Samples in the Sample Editor

The most intuitive way to adjust sample and loop start and end points is to have a visual representation of the waveform, which you can use as a guide to graphically move the points to where you want them to be on the waveform. This allows you to get much closer to the desired values much quicker than having to type in numerical values without any visual guide.

The Instrument Editor does not offer a graphic representation of a sample's waveform. Because of this, the only way to set the start and end points or loop points of a sample within the Instrument Editor is numerically. Thankfully Control-clicking the loop start and end point parameters will open a shortcut menu, allowing you to open the selected sample in the Logic Pro Sample Editor (or the external sample editor set in the preferences). This allows you to edit the sample borders and loop points graphically. You can then simply save the sample, and the new values will be written to the audio file header, and used properly by the EXS24 mkll.



Once you save and reopen a sample that you have edited in either the Logic Pro Sample Editor, or a third party sample editor, it is likely that either the start and end points, or loop points—in the Parameters area—will no longer be accurate. The Update Selected Zone(s) Info from Audio File command (in the Zone menu) reads loop settings and start and end points from the audio file, and updates the settings of the zone accordingly.

The Sample Editor window also offers the Edit > Sample Loop \rightarrow Selection, Edit > Selection \rightarrow Sample Loop, and Edit > Write Sample Loop to Audio File functions.

To use the Sample Editor Loop commands:

- 1 Choose either of the selection commands in the Edit menu of the Sample Editor:
 - Sample Loop → Selection: The loop area (defined by the Loop Start and End points) is used to select a portion of the overall audio file.
 - Selection → Sample Loop: The selected area is used to set the Loop Start and End points.
- 2 Once you've selected the desired area using either of the commands above, choose Edit > Write Sample Loop to Audio File.

The new loop values will be written to the audio file header.

Setting Group Parameters

Groups offer several parameters for simultaneous control over all assigned zones.



Key Range

The two key range parameters allow you to define a key range for the group.

- Lo(w): Sets the lowest note at which the group will sound
- *Hi(gh):* Sets the highest note at which the group will sound.

Playing notes outside this range will not trigger the zones assigned to this group.

Velocity Range

The two velocity range parameters allow you to set up a velocity range for the group. Use these parameters for sounds where you wish to mix, or switch between, samples dynamically by playing your MIDI keyboard harder or softer—with layered sounds, or when switching between different percussion samples, for example. Low sets the lowest velocity at which the group will sound, and High sets the highest velocity at which the group will sound. Playing notes outside this range will not trigger the sample assigned to this group.

The settings made here override the settings in the zones, if necessary: When a zone's velocity range is larger than that allowed by the group setting, the zone's velocity range is limited by the group setting.

Output Parameters

- *Volume:* Adjusts the volume of the group, and therefore the volume of all assigned zones, simultaneously. This works much like a sub-group volume slider on a mixing console.
- *Pan:* Adjusts the pan position of the group (stereo balance for stereo samples), and the pan position of all assigned zones simultaneously.
- Routing: Determines the outputs used by the group. Choices include the main outputs, and paired channels 3 and 4, 5 and 6, 7 and 8, 9 and 10, or individual outputs 11 through to 16. This allows individual groups to be routed independently to aux channels in a multi-output EXS24 mkll instance.

Voices Parameters

- *Poly. (Polyphony):* Determines how many voices the group is allowed to use. A practical use of this would be to set up a classic hi-hat mode within a full drum kit, mapped across the keyboard. In this scenario, you could assign both an open and closed hi-hat sample to a group, and set the Voices parameter of the group to 1. In this example, the most recently triggered of the two hi-hat samples will mute the other, as only one voice is allowed for the group. This mirrors the real-world behavior of hi-hats. When samples in zones are assigned to another group, the other sounds of the drum kit can still be played polyphonically. The Max option ensures the group uses the maximum number of voices allowed by the Voice parameter in the Parameters window.
- *Trigger menu:* Determines if zones pointing to this group are triggered on key down (Key Down setting) or on key release (Key Release setting). This is useful for emulating organ key clicks, for example, in which you may want the organ note triggered on key down, but the organ click triggered on key release.
- Dc (Decay) checkbox: Activate to access the Decay Time parameter.
- (Decay) Time: Determines how long it will take the volume of a sample triggered by key release to decay. Keep in mind, the decay parameters will only function when the Trigger is set to Key Release.

Filter Offsets

Allow you to offset the Cutoff and Resonance settings of the Parameters window separately for each group. This can be useful if you want the initial impact of a note to be unfiltered for one group, but not others.

Envelope 1 and Envelope 2 Offsets Parameter

Use these parameters to offset the envelope settings from the Parameters window separately for each group. This is useful if you want the filter or volume envelopes to affect the samples in a group after the initial impact of the triggered sounds.

Note: When the Trigger parameter is set to Key Release, the Decay Time parameter determines the volume decay, not Envelope 2 (the volume envelope). This means that when Trigger is set to Key Release, the Envelope 2 Offsets have no effect.

The H (Hold) parameter determines the time period that the envelope will be held at the maximum attack level, before the decay phase begins.

Select Group By

By using the parameters in this column, you can define a specific selection event to select a group. Whenever the defined selection event is triggered, zones pointing to this group can be played, and other groups (selected using a different selection event) are not played. The defined event is not triggered itself, it simply acts as a remote for selecting the group in question.

This feature works with MIDI events (Notes, Controllers, Bends, MIDI Channel), and groups, once you define which group number the Select By command will start with. Once defined, selecting that group means that only zones pointing to that group will play, and other groups are not played.

For example, if you want the EXS24 mkll to automatically switch between two string sample groups, one for staccato samples and one for legato samples, you could use the Select Group By menu selection for MIDI notes, and assign a different MIDI note to trigger each group. This way, you can use one note that you are not playing to select between the two groups, and automatically switch groups from the keyboard as you are playing. Also see "Remapping of Pitch Bend and Modulation Wheel Events" below for another example of how this feature can be used to automatically switch between groups.

Click the plus icon in the upper right corner of the Select Group By column to refine the group selection conditions. Click the minus icon to remove a Select Group By condition, and broaden the group selection criteria.

Remapping of Pitch Bend and Modulation Wheel Events

In order to create realistic-sounding performances in an easy and intuitive way, the Jam Pack 4 (Symphony Orchestra) instruments use the modulation wheel to switch between articulations (legato, staccato, and so on) and the pitch bend wheel to change expression (crescendo, diminuendo, and so on). Further information about this can be found in the Jam Pack 4 documentation.

This is achieved by internally remapping pitch bend events to MIDI controller 11 and modulation wheel events to MIDI controller 4. To ensure compatibility with the Jam Pack 4 instruments, the EXS24 mkll automatically uses this remapping behavior for Jam Pack 4 instruments.

You can also use this remapping model for other instruments by activating the Map Mod & Pitch Wheel to Ctrl 4 & 11 setting in the Instrument menu.

The EXS24 mkll will remap incoming pitch bend and mod wheel events to controller 11 or controller 4, respectively. The default pitch bend and modulation wheel functionality can not be used.

Editing Zones and Groups Graphically

You are not limited to editing zones and groups in the Parameters area. You can also edit some zone and group parameters graphically in the Zones Display or Groups Display area above the Instrument Editor keyboard.



To move a zone or group:

1 Move the mouse cursor over an existing zone or group. It will change to a two-headed arrow.



2 Click-hold and drag the zone or group to the desired position.

Note: You can also change the root key of a zone, by pressing Option and Command simultaneously, while dragging the zone.

You can move multiple zones or groups at the same time by rubber-band, Shift-click, or Command-click selecting them.



To change the start or end note of a zone or group:

1 Move the mouse cursor to the beginning or end of a zone or group.

The cursor will change to a two header arrow.



2 Click-hold and drag the start or end point of the zone or group to the desired position.

You can also shift a zone to the left or right by using one of the following two key commands:

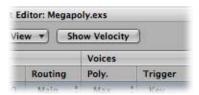
- Shift Selected Zone(s)/Group(s) Left: Option-Left Arrow key)
- Shift Selected Zone(s)/Group(s) Right: Option-Right Arrow key)

If you want to shift the root note of the zone along with the zone position, use one of the following two key commands:

- Shift Selected Zone(s)/Group(s) Left (Zones incl. Root Key): Shift-Option-Left Arrow key
- Shift Selected Zone(s) Right (Zones incl. Root Key): Shift-Option-Right Arrow key

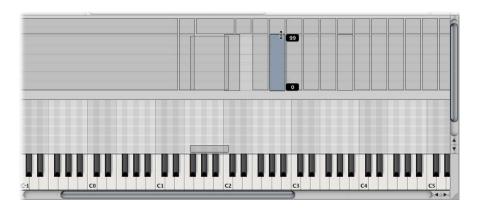
To edit the velocity range of a zone or group:

1 Click the Show Velocity button at the top right of the Instrument Editor (or use the Show/Hide Velocity key command).



The Velocity Display area will open above the Zones or Views Display area.

- 2 Click on one or more zones or groups in the Display area.
 The velocity bars of the selected zones will be highlighted in the Velocity Display area.
- 3 Click-hold on either the High or Low value of the velocity bar of the zone or group you wish to edit.
- 4 Drag the parameter up to raise the value or down to lower the value.



Sorting Zones and Groups

You can easily sort zones and groups in the EXS24 mkll Instrument Editor by clicking the sub-column heading that you wish to sort by. For example, if you want to sort your zones by name, click the Name sub-column heading under the Name column, and your zones will be sorted alphabetically by name.



If you wanted to sort groups by lowest to highest starting velocity, for example, you would click the Low sub-column heading in the Velocity Range column, and your groups would be sorted with the group that had the lowest starting velocity range first.

Showing and Hiding Zone and Group Parameters

Using the View menu, you can determine which zone and group parameters you want to view in the Parameters area of the Instrument Editor:

- *View All:* Choose to show all available columns and sub-columns (corresponding key command: View: View All).
- Individual Group and Zone display settings: Enable the columns and sub-columns you want to be displayed. The zone entries are available in Zone view. The groups entries are available in Groups view.
- **Tip:** Pressing Option and choosing a disabled zone or group column will display only the chosen column.
 - Restore to Default: Choose to revert to the default view (corresponding key command: View: Restore to Defaults).
 - Save as Default: Saves the current view of zone and group parameters as the default view whenever you open the EXS24 mkll Instrument Editor.

Saving, Deleting, Renaming, and Exporting Instruments

You can access all basic sampler instrument operations in the Instrument Editor's Instrument menu.

Save

Saves the currently loaded sampler instrument. When you create a new instrument and save it for the first time, you will be asked to give it a name. If you have edited an existing sampler instrument and save it via this command, the existing file name is used and the old instrument is overwritten. You can also use the Save Instrument key command.

Save As

This command also saves the currently loaded sampler instrument. When Save As is used, you will be prompted to give the file a name. Use this command when you want to save a copy or multiple versions of an edited sampler instrument, rather than overwriting the original version.

Rename

This command allows you to rename the loaded sampler instrument. The renamed version replaces the previous version on the hard disk.

Export Sampler Instrument and Sample Files

Copies the selected sampler instrument and its audio files into another folder location of your selection. Choosing this command opens a standard operating system file navigation/browse dialog. You may browse to an existing folder or create a new folder with a new name, as desired. You can also use the Export Sampler Instrument and Sample Files key command (default: Control-C).

Setting Sampler Preferences

The EXS24 mkll offers a separate Sampler Preferences window, allowing you to configure various operational and sample related preferences, such as sample rate conversion quality, velocity responsiveness, sample storage, search-related parameters, and so on.

To open the Sampler Preferences window, do one of the following:

- Click the Options button in the Parameters window, then choose Preferences in the pop-up menu.
- Choose Edit > Preferences in the Instrument Editor.



Sample Rate Conversion

Choose the interpolation quality used by the EXS24 mkll. When Sample Rate Conversion is set to Best, the highest possible sound quality is maintained when transposing samples.

Sample Storage

Choose the format in which the EXS24 mkll handles the loaded sample data. When set to Original, the samples are loaded into RAM at their original bit depth, and are converted to the internal 32 bit floating point format of Logic Pro on playback. When 32 Bit Float is chosen, the samples are stored and loaded in this format. This eliminates the need for any real time conversion, meaning that the EXS24 mkll 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.

Velocity Curve

Determines how responsive the EXS24 mkll is to velocity values received from your MIDI keyboard. Negative values increase the responsiveness to soft key strikes, and positive values decrease it.

Search Samples On Menu

Determines the location that instruments samples should be searched in. You may either choose the drives normally used by the operating system or external SCSI, FireWire, or USB drives, accessible directly or over a network. Drives can be selected individually, or grouped as follows:

- Local Volumes internal storage media (hard disks and CD ROM mechanisms) attached to or installed in the computer directly.
- External Volumes storage media accessible over a network.
- All Volumes both internal and network media are scanned for appropriate data.

Note: Selecting External or All Volumes may result in a dramatic increase in the time required by the EXS24 mkll to find and load sampler instruments and files.

Read Root Key From Menu

Allows you to determine how the EXS24 mkll determines the root key of loaded audio files. You can choose between the following options:

- File/filename: Initially reads information about the root key from the file itself (in the header of the AIFF or WAVE file) when loading an audio file into a zone. If no information of this type exists in the file header, a smart analysis of the filename may detect a root key. If this second method doesn't provide any useful results, C3 will be used as the default root key in the zone.
- Filename/file: As above, but vice versa, with the filename read first, and the header read second.
- Filename only: Reads from the filename only. If no root key information exists, C3 will automatically be assigned to the zone as the root key.
- File only: Reads from the file header only. If no root key information exists, C3 will automatically be assigned to the zone as the root key.

Root Key at File Name Position

Normally, the EXS24 mkll intelligently determines the root key from the file header of the loaded audio file. However, sometimes you may want manual control over this parameter, if you feel that the root key is not being properly determined. For those occasions, use the Root Key at File Name Position parameter. Possible options in the pop-up menu are Auto, or numerical values from 1 to 30.

Auto is the recommended value. It provides a smart analysis of numbers and keys from the file name. A number in the file name can be recognized, regardless of its format—60 or 060 are both valid. Other valid numbers can range between 21 and 127. Numerical values outside of these are generally just version numbers. A key number is also a valid possibility for this use—C3, C 3, C_3, A-1, A -1 or #C3, C#3, for example. The possible range is C-2 up to G8.

Note: There may be cases where a sound designer has used multiple numbers in a filename, which is common with loops, with one value being used to indicate tempo—"loop60-100.wav", for example. In this situation, it isn't clear which, if either of the numbers, indicates a root key or something else: 60 or 100 could indicate the file number in a collection, tempo, root key, and so on. You can set a value of 8 to read the root key at position (letter/character) eight of the filename—namely the 100 (E6). Alternately, setting a value of 5 will select the 60 (C3) as the root key position.

Previous Instrument and Next Instrument

You can use the Previous Instrument and Next Instrument preferences to determine which MIDI event type (and data value) will be used for selection of the previous or next instrument.

Choose the desired MIDI event type in the Previous Instrument and Next Instrument menus. Choices include Note, Poly Pressure, Control Change, Program Change, Channel Pressure, and Pitch Bend.

In the field next to the menus, you can enter either the note number or the value of the first data byte. When Control Change 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 EXS24 mkll.

Ignore release velocity

This option also refers to the release trigger function of the Gigasampler 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 projects

Determines whether or not the samples commonly used by two open project files are reloaded when switching between projects.

Configuring Virtual Memory

These days, many sample libraries contain many gigabytes of audio samples in order to create the most accurate sampler instruments possible. Often, these gigantic sample libraries are too large to fit into your computer hardware's random access memory (RAM) all at once. To let you use these huge sample libraries, the EXS24 mkll can use a portion of your hard drive as virtual memory. When you turn on the EXS24 mkll's virtual memory, only the initial attacks of audio samples are loaded into the computer's RAM; the rest of the sample is streamed in real time from the hard drive.

You can configure the EXS24 mkll's virtual memory feature in the Virtual Memory window.

To open the Virtual Memory window:

• Click the Options button in the Parameters window, then choose Virtual Memory in the pop-up menu.



- Active checkbox: Click to activate the EXS24 mkll's virtual memory feature.
- Disk Drive Speed: Refers to the speed of your hard drive; if you have a 7200RPM or
 faster hard drive for your audio samples, select Fast. If you are using a 5400RPM
 laptop drive for your audio samples, select Medium. Generally, you will not need to
 use the Slow setting with any modern Macintosh.
- Hard Disk Recording Activity: Refers to how much recording and streaming of non-sampler related audio you are doing. For example, if you are recording entire drum kits using over a dozen microphones, streaming live guitars and bass, recording choirs, and so on, you would set your hard disk recording activity to High. On the other hand, if your projects consist mostly of software instruments, with perhaps a recorded instrument or vocal or two, you would set your hard disk recording activity to Low. If you are not sure if you are unsure, keep this parameter set to average.
- Requires Constant RAM Allocation of field: Displays the memory usage required by the
 two parameters mentioned above. The slower your hard drive and the higher your
 hard disk recording activity, the more RAM you will need allocated to virtual memory.

Performance section: Shows the current disk I/O traffic and the data not read from
disk in time. If these numbers start rising, the EXS24 mkll may glitch when trying to
stream your samples from the disk in time with your performance. If you notice these
values rising to high levels, you should change the general settings to free up
additional RAM for virtual memory use. If you continue to experience high levels in
these values and glitching in your audio, you should consider installing more RAM
into your Macintosh.

Using the VSL Performance Tool

The EXS24 mkll features an additional interface for the Vienna Symphonic Library Performance Tool. 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.

You can use the External Instrument to route your external MIDI sound generators through the Logic Pro Mixer, allowing you to process them with Logic Pro effects.

Ideally, you will use a multi input and output audio interface, to avoid constant repatching of devices. The External Instrument can be inserted in software instrument channels in place of a software instrument.

External Instrument Parameters



- *MIDI Destination menu:* Choose the desired MIDI instrument object in your Environment.
- *Input menu:* Choose the inputs of your audio hardware that the MIDI sound generator is connected to.
- Input Volume slider and field: Determines the incoming signal level.

Using the External Instrument

The following section outlines the steps required to route external MIDI sound generators through the Logic Pro Mixer.

To process external MIDI instruments with effects:

1 Connect the output (or output pair) of your MIDI module with an input (pair) on your audio interface.

Note: These can be either analog or digital connections if your audio interface and effects unit are equipped with either, or both.

- 2 Create an instrument channel.
- 3 Click the Instrument slot, and choose External Instrument from the pop-up menu.
- 4 Choose the MIDI Destination from the menu in the External Instrument window.
- 5 Choose the input (of your audio interface) that the MIDI sound generator is connected to from the Input pop-up menu.
- 6 Adjust the Input Volume, if necessary.
- 7 Insert the desired effects in the Insert slots of the channel.

As the track is routed to an instrument channel (which is being used for an external MIDI sound module), it behaves just like a standard software instrument track, which means that you can record and play back MIDI regions on it—with the following benefits:

- You can take advantage of the sounds and synthesis engine of your MIDI module, with no overhead on your Macintosh CPU (apart from the effects used on the channel).
- You can use insert effects, obviously, but can also use Send effects by routing the Instrument channel to aux channels.
- You can bounce your external MIDI instrument parts—with or without effects—to an audio file, in realtime. This makes the creation of a mix, inclusive of all internal and external devices and tracks, a one step process.

Additional Things to Bear in Mind

If using multitimbral MIDI sound sources, please be aware that each External Instrument requires a separate audio output.

Freezing an External Instrument track cannot happen faster than realtime, as per any Bounce operations where MIDI hardware is involved.

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

Klopfgeist is inserted on instrument channel 128 by default, and used to generate the MIDI metronome click.

Theoretically, any other Logic Pro or third-party instrument could be used as a metronome sound source on instrument channel 128. Similarly, Klopfgeist can be inserted on any other instrument channel for use as an instrument.

A look at Klopfgeist's parameters will, however, clearly show that it is a synthesizer designed to create the metronome's clicking sound.



- *Trigger Mode buttons:* Click to operate Klopfgeist as a monophonic or polyphonic (4 voice) instrument.
- Tune knob and field: Tunes Klopfgeist in semitone steps.
- Detune knob and field: Fine-tunes Klopfgeist's in cents.
- *Tonality slider and field*: Changes the sound of Klopfgeist from a short click to a pitched percussion sound—similar to a Wood Block or Claves.
- *Damp slider and field:* Controls the release time. The shortest release time is reached when *Damp* is at its maximum (1.00) value.

• Level Via Vel slider and fields: Determine the velocity sensitivity of Klopfgeist. The upper half of the two-part 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.

Sculpture 28

Sculpture is a synthesizer that generates sounds—based on a simulated string or bar—that is in motion, or vibrating.

To keep things clear, this chapter will always refer to the string, even though many of the sounds you can create with Sculpture have nothing in common with what you'd expect from a stringed instrument!

Sculpture uses a method of synthesis called component modeling. This approach to tone generation shares some aspects and parameters with other synthesis techniques, such as those found in additive and subtractive synthesizers. As such, many of the parameters used by Sculpture will be immediately familiar to you, such as LFOs, vibrato, envelopes, and so on. Many others, however, will be very new.

Sculpture's recordable envelopes provide massive scope for sonic animation, with the payoff being some just plain astonishing sounds! Sculpture's flexibility in this area provides a level of control that will amaze you, even if you're an experienced synthesizer user.

Make full use of every control and parameter that is available—when initially auditioning some of the supplied factory sounds, and when creating new ones of your own. Sculpture is very much a performance synthesizer, that benefits from use of controllers, modulations, and different playing techniques. On the sound design side of things: don't be afraid to experiment—that's what Sculpture was created to do!

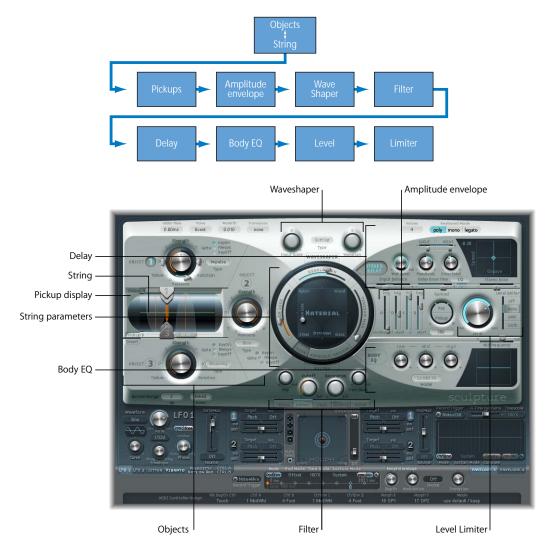
The creation of particular types of sounds is discussed in the "Programming: Quick Start Guide" on page 511.

Before taking a look at the feature set, please read the following brief section on the synthesis core of Sculpture.

The Synthesis Core of Sculpture

This section is designed to give you a feel for the way Sculpture works. Component modeling, as you'll discover, is quite different to traditional synthesis methods—and so are the results!

The layout of parameter descriptions follows the signal path (shown in the diagram) of the core synthesis engine. Please check out the options of each parameter as you read about them. This will give you a better feel for where things are, and what's available.



The *string* is the central synthesis element. It offers a range of parameters that allow you to adjust its material—what it's made of, in other words.

Up to three objects of different types are used to excite or disturb the vibration of the string. These objects can be positioned anywhere along the string, and offer multiple parameters for adjustments to their properties. The string itself doesn't make a sound unless it is stimulated (excited or disturbed) by the objects.

The vibration of the string is passed on to the ensuing processing section—via two moveable pickups (you can view these as being similar to the electromagnetic pickups found on guitars or electric pianos and clavinets).

The processing section consists of the ADSR-equipped amplitude stage, a Waveshaper (with selectable types) and a multi-mode filter.

Note: All elements described above exist on a per voice basis.

The sum of all voice signals is then processed by an integrated Delay effect.

From there, the signal is sent to an EQ-like module (Body EQ), which globally simulates the spectral shape/body response of your instrument.

The resulting signal is then fed to a Level Limiter section.

A vast number of modulation options are also available, from tempo-synced LFOs to jitter generators and recordable envelopes.

A (recordable) Morph function also allows for smooth or abrupt transitions between (up to) five morph snapshots.

The String as a Synthesis Element

Sculpture, as a component modeling synthesizer, provides you with a string—rather than the oscillators found in traditional synthesizers.

The string is considerably more sophisticated in concept than simple oscillators. Basically, you are actually creating the waveform, or base timbre from nothing. This is achieved by (mathematically) describing the string's properties, and that of its environment. These include, amongst others: the material the string is made of, the thickness and tension of the string, its characteristics over time, the atmosphere (water, air, and so on) it is being played in, and the way it is being played—struck, bowed, and so on.

In Sculpture, the general timbre is set by the string parameters. Modules such as the Waveshaper, filter, and Body EQ are useful additions that can help to refine the overall sound.

Sculpture goes far beyond the mere creation of an infinite number of base timbres, however. One of the key differences between Sculpture's string and a traditional synthesizer's waveform is that the base timbre (provided by the string) is in a constant state of flux.

Put another way, if Sculpture's string is still vibrating for a specific note, retriggering that same note will interact with the ongoing vibration. This is not dissimilar to the effect of repeated plucking of a guitar string, where the string is still vibrating when the next note is played. This will alter the harmonic spectrum each time—which is why acoustic guitars sound organic when a note is played repeatedly, and sampled guitars don't.

As you can see, this is quite different to other synthesis methods where the base timbre (waveform), even if modulated, does not harmonically interact with sounding notes when retriggered. What usually happens in traditional synthesizers is that the waveform is restarted—from mid cycle, or from the beginning—with the result being an increase in volume, or a slight cyclical wave shift.

Beyond this aspect of base timbre control and interaction, Sculpture also provides you with the means to model (emulate) a number of other acoustic instrument properties. As examples, you can emulate a bow dragging across a string, a piano hammer strike, or even dropping a coin onto the bridge of a guitar.

There would be no point in bowing or plucking a string, however, if you didn't have a neck or a body to your violin, cello, or guitar. You'd probably want some way to change the tension of the strings, or try steel, rather than nylon or catgut. In other words, to control the physical properties of the string, and its excitation (how it is played).

All of this is possible with Sculpture.

It is an instrument that will require some investment of your time, but it will reward you with beautifully warm organic sounds, evolving soundscapes, or a harsh and metallic Hell's Bells patch—if you're after that sort of thing.

Sculpture's Parameters

This section covers the parameters of Sculpture. Before touching on them, however, a key factor with component modeling is the interaction between various sections of the synthesis engine. This can lead to some truly unique sounds, but can also lead to unexpected results.

Sculpture is very different to traditional synthesizers, and requires a more measured approach to achieve a particular end result. A collection of tips can be found in the "Programming: Quick Start Guide" on page 511, that will assist you in the creation of particular types of sounds. Please ensure that you read this section.

Keep the Sculpture flowchart diagram (on page 456) handy while familiarizing yourself with the interface/programming. If your approach is methodical—and you follow the flowchart, you shouldn't encounter too many surprise results.



The user interface of Sculpture is broken down into three main areas. The large silver section at the top contains the sound engine. It is divided into four sections:

- Global parameters: Light silver area across the top, incorporating Glide, Tune, and other global parameters.
- String and object parameters: Dark silver area to the left, and the Material Pad ring in the center.
- Processing parameters: These parameters include the green-gray Filter area,
 Waveshaper (in light silver area at the top), pickup (in dark silver area to the left) and amplitude envelope (in dark silver area to the right).
- Post-Processing parameters: Dark silver area to the right that contains the Delay, Body EQ and Spread, Level, and Level Limiter parameters. This area also houses the amplitude envelope parameters.

The blue/gray area below houses the polyphonic modulation sources (LFOs, Jitter Generators and recordable envelopes).

Global control sources, incorporating the Morph Pad, are located on the bottom, gray edge. This is where you can assign MIDI controllers to various Sculpture parameters.

Global Parameters

These are found across the top of the Sculpture interface, unless otherwise specified.



Transpose

Transpose is used for coarse tuning of the entire instrument. Given the ability of component modeling to radically alter pitch with certain settings, coarse tuning is limited to octave increments.

Tune

Tune is used for fine tuning of the entire instrument (range: ± 50 cents). A cent is 1/100th of a semitone.

Warmth

Warmth is used to slightly detune the different voices, much like the random fluctuations caused by the components and circuitry of analog synthesizers. As the parameter name suggests, this warms up or thickens the sound.

Glide Time

Values above 0.0 lead to a softer pitch change between played notes.

Voices

When Keyboard Mode is set to Poly, this parameter limits the number of simultaneously sounding voices to the set value. A value of 16 voices is the maximum polyphony of Sculpture.

Keyboard Mode

Here, you can select between the Mono, Legato, or Poly Keyboard Modes. You can switch between mono and poly modes by clicking on the Poly and Mono buttons. The Portamento, the duration of which is set by the Glide Time parameter, affects legato performances. As long as you don't release the key of the previously struck note before clicking a new key, you'll hear the glide or portamento. If you release every note before striking another key (portato or even staccato style), the glide effect will not be audible.

When the Legato button is activated, a further side-effect occurs. The amplitude envelope is not retriggered—the sound is sustained and the attack phase of newly played notes is not retriggered. These behaviors are sometimes referred to as single trigger and multi trigger modes.

All modes simply retrigger a (potentially sounding) voice with the same pitch, instead of allocating a new one. As such, multiple triggering of a given note results in slight timbral variations, depending on the current state of the model at note-on time.

If Sculpture's string is still vibrating for a specific note, retriggering that same note will interact with the ongoing vibration, or current state of the string.

Important: A true retrigger of the vibrating string will only happen if both Attack sliders of the amplitude envelope are set to zero. If either slider is set to any other value, a new voice will be allocated with each retriggered note.

Bender Range Up/Down

These parameters are found below Object 3, on the left-hand side of the Sculpture interface.



Separate settings are available for upwards and downwards pitch bends—using your MIDI keyboard's pitch bend controller.

When Bender Range Down is set to Linked, the Bender Range Up value is used for both (up/down) directions.

Note: Bending the string will, just like the real deal on a guitar, alter the shape of the model, rather than merely act as a simple pitch bend.

String and Object Parameters

The string and object parameters discussed in this section apply on a per-voice basis.

You will note a number of parameter names followed by (morphable). This indicates that the parameters can be morphed between up to five snapshots, called morph points. More details on morphing can be found in "Morphing" on page 500.

Hide, Keyscale, and Release Buttons

This section briefly covers some common parameters that you'll encounter when manipulating the string. The string's default note position is C3 (middle C).

Three buttons are available for your use that will activate and hide the various Keyscale and Release parameters. Simply click on the Keyscale, Release, or Hide button, dependent on the adjustments you would like to make. The corresponding parameters will become visible in the ring surrounding the Material Pad.



- Keyscale parameters can be set for notes that fall below C3, or notes that are positioned above it. In simple terms, the impact of these parameters can be controlled across the keyboard range. As an example: a parameter such as string stiffness (discussed shortly) could be more intense for high notes, and less intense for low notes. In practical terms, this would result in more harmonic (sweeter) sounding bass notes, and inharmonic overtones in treble notes (notes above C3).
- The string Release parameters impact on the vibrations of the string once the key has been released.
- The Hide button is handy for avoiding accidental parameter changes, and simplifies the interface.

The Material Pad

The following two string material parameters determine the general timbre, and are controlled by the ball (which correlates to the X and Y co-ordinates) within the *Material Pad*.



The crosshair is a handle for the Key Scale and Release Scale diamonds in cases where these are hidden by the ball. It also allows you to independently change the keyscaling for one of the two axes (X/Y positions—Inner Loss or Stiffness).

In general synthesizer terms, this could be viewed as a base timbre generator in the oscillator section.

Note: Option-clicking resets all string parameters to their default values.

Inner Loss (Morphable)

This parameter is used for damping of the string, as caused by the string material—steel, glass, nylon, or wood. These are frequency-dependent losses which cause the sound to become more mellow during the decay phase. Simply click-hold, and drag, the Inner Loss and Stiffness ball to the desired position.

Inner Loss Scale Low/High

Allows independent adjustments to the key tracking of inner losses for notes above and below C3.

To adjust Inner Loss keyscaling, first enable the Keyscale button, then click-hold on the green horizontal line for low notes or the blue horizontal line for high notes, and drag up/down to the desired position.

A diamond indicates the intersection between the Inner Loss and Stiffness Low/High Scaling positions. You can click-hold and drag this diamond directly, to adjust both parameters simultaneously.

Inner Loss Scale Release

Values above 1.0 cause the inner losses to increase when the key is released. This is quite unnatural, as this would mean that the string material would change after the note was released.

In practice, however, the use of this parameter in combination with Media Loss Scale Release allows a natural simulation of strings that are dampened at note-off time.

To adjust, first enable the Release button, then click-hold on the blue Release line, and drag up/down to the desired position.

Stiffness (Morphable)

Controls the stiffness (rigidity) of the string. In reality, this is determined by the string material and diameter (or to be more precise: by its geometrical moment of inertia). Stiff strings exhibit an inharmonic vibration, where overtones are not integer multiples of the base frequency. Rather, they have higher frequencies.

In effect, an increase in Stiffness turns the string into a solid metal bar.

Simply click-hold, and drag, the Inner Loss and Stiffness ball to the desired position. Note that as you do so, the thickness of the string—the green horizontal line in the Pickup display—will change.

Stiffness Scale Low/High

Allows independent adjustments to the key tracking of the stiffness parameter, for notes above and below C3.

To adjust Stiffness keyscaling, first select the Keyscale button, then click-hold the green vertical line for low notes or the blue vertical line for high notes, and drag left/right to the desired position.

You can also simultaneously adjust both Stiffness and Inner Loss keyscaling by dragging the diamond that intersects the green lines.

Setting Inner Loss and Stiffness

Low Stiffness values combined with low Inner Loss values lead to metallic sounds. Increasing the Stiffness makes the sound become more bell or glass-like.

Increasing the Inner Loss value, while maintaining a low Stiffness level, corresponds to nylon or catgut strings.

High Stiffness values combined with high Inner Loss values simulate wood-like materials.

The String Parameters Around the Material Pad

The following section covers the string parameters around the Material Pad.



Resolution (Harmonics)

This parameter determines the maximum number of harmonics contained in (and spatial resolution of) the sound at C3. This is roughly proportional to the required CPU power, so the more harmonically-rich/the higher the Resolution setting of the sound, the more processing muscle will be required.

Note: As you alter the Resolution value, you are changing the interaction of the string with the objects. This also changes the frequency of the overtones: very low Resolution values result in inharmonic spectra—even with Stiffness set to zero.

Resolution Scale Low and High

Allows you to set the key tracking resolution (the accuracy of key tracking) separately for notes above and below middle C (C3). The green low slider (down to C0), inside the Material Pad ring, is visible when the Keyscale button is clicked. To adjust, click-hold, and drag. The blue high slider (up to C6) runs along the left-hand side of the outer ring. Click-hold, and drag, to adjust.

Media Loss (Morphable)

Controls dampening of the string caused by the surrounding media, for example; air, water, pea and ham soup, and so on. These losses are independent of frequency. This approach allows control over the duration of the (exponential) amplitude decay, once the excitation of the string has stopped.

Media Loss Scale Low and High

Allows independent key tracking adjustments of media losses for notes above and below middle C (C3). The green low slider (down to C0), inside the Material Pad ring, is visible when the Keyscale button is clicked. To adjust, click-hold, and drag the green arrow. The blue high slider (up to C6) runs along the left-hand side of the outer ring. Click-hold, and drag the blue arrow, to adjust.

Media Loss Release

The blue slider (in the outer ring of the Material Pad) controls the Media Loss Release time. To activate it, you must first enable the Release button, to the bottom right of the Material Pad. Values above 1.0 cause media losses to increase when the key is released. This parameter can be used to simulate a string that is dropped into a bucket of water after initially vibrating in air, for example. Obviously, this is not what the average violinist or pianist would do, but it can be useful for a number of interesting sound variations.

Tension Mod (Morphable)

Strings, such as those of a guitar, exhibit a particularly prominent nonlinear behavior: if the string excursion is large, the string is detuned upwards. As this detuning is caused by the momentary, rather than the average excursion of the string, the detuning occurs very quickly. This phenomenon is known technically as tension modulation nonlinearity. Non-technically, setting or modulating the Tension Mod parameter to values above 0.0 emulates this momentary detuning effect in Sculpture.

Please note that this nonlinear effect can produce some surprising results, and can also make the entire model unstable, especially when combined with low Media Loss and Inner Loss values. So, if you have a sound that spikes or drops out during the decay phase, try reducing Tension Mod (and perhaps Resolution).

Tension Mod Scale Low and High

Allows independent adjustments to tension modulation key tracking, for notes above and below middle C (C3). The green low slider (down to C0), inside the Material Pad ring, is visible when the Keyscale button is enabled. To adjust, click-hold, and drag. The blue high slider (up to C6) runs along the left-hand side of the outer ring. Click-hold, and drag, to adjust.

Note: If you find that your instrument seems slightly sharp or flat as you play up/down the keyboard, look at making adjustments to the Tension Mod and perhaps Media Loss Keyscale parameters.

Excite and Disturb Object Parameters



The following parameters are used to excite, disturb, or dampen the string.

Important: At least one *object must* be used, as the string itself does not make any sound!

As you'll discover shortly, there are a number of different string *excite* models such as blow, pluck, bow, and so on. Needless to say, these quite radically alter the general timbre of the string's attack phase, resulting in bowed or plucked flute and bell sounds, or guitars with a flute-like blown sound.

Judicious use of the object parameters can deliver very accurate emulations of real-world instruments, or sounds that are altogether more other-worldly.

A particular aspect of component modeling to note is that each additional disturb/damp object that is activated will impact on the string. This will, in turn, alter the interaction of *any* other active object with the string, often resulting in a completely different character to your sound.

Obviously, changing the sonic character is the reason why you would use a new object, but the pluck and blow combination you selected may sound like fingernails on a blackboard—rather than the plucked pan flute you were going for—depending on other string settings.

As such, you need to pay special attention to the type and strength of objects. You may find that the flavor of the excite object (1 or 2), for example, has changed significantly—and you may need to adjust or change the parameters of *all* objects (and perhaps several string parameter values) after introducing a new disturb/damp object (2 or 3). Similarly, the selection of a different *type* of excite object will impact on the disturb/damp objects (and the string, obviously), and therefore the character of your sound.

The repositioning of objects changes the timbre of the string. If emulating, say a guitar, changing an object position could be viewed as similar to picking or bowing a string at various spots along a fretboard.

About Objects and Velocity Sensitivity

It is important to note that:

- Object 1 is velocity sensitive.
- Object 2 is only velocity sensitive when a type that actively excites the string is selected. When damping objects are used, Object 2 is not velocity sensitive.
- Object 3 is not velocity sensitive. This is because Object 3 can only be used as a disturber of the string and not as an active excitation element.

Enabling Objects

Click the 1, 2, or 3 button to enable or disable the respective object. When active, each button is lit in a fetching shade of agua blue.





Type

The following two tables list all types available for Sculpture's objects. Click on the Type button for these objects, and choose from the menu.



Note: Object 1 can only make use of the excite types found in the first table. Object 2 can make use of any of the types available in either table. Object 3 can only make use of the disturb/damp types found in the second table.

Excite Types (Objects 1 and 2)

The following table lists all excite types available for Objects 1 and 2, and information on the controls available for each.

Name	Description	Strength controls	Timbre controls	Variation controls
Impulse	A short impulse excitation	Impulse amplitude	Width	Velocity dependency of width
Strike	Short excitation like piano hammer or mallet	Hammer start speed (velocity dependent)	Hammer mass	Felt stiffness
GravStrike	Like hammer but with gravitation towards the string, leading to multiple hammer-string interactions and disturbed string vibrations	Hammer start speed	Felt stiffness	Gravitation
Pick	Finger or plectrum picking	Pickup force and speed	Force/speed ratio	Plectrum stiffness
Bow	Bowing of the string	Bow speed	Bow pressure	Slip stick characteristics
Bow Wide	Same as bow, but wider, resulting in a more mellow tone, especially suited for smooth bow position changes	Bow speed	Bow pressure	Slip stick characteristics
Noise	Noise injected into the string	Noise level	Noise bandwidth/ cutoff frequency	Noise resonance
Blow	Blow into one end of the string (an air column, or tube). At various positions, starting from 0.0 (far left): move the blowing direction and position from along the string, towards one end. The string is blown sideways at the chosen position.	Lip clearance	Blow pressure	Noisiness
External (Only avail. for Object 2)	Feeds sidechain signal into string.	Level	Cutoff frequency of lowpass filter being used to process sidechain signal	Width (size) of the string area being affected by the side chained signal

Disturb and Damp Types (Objects 2 and 3)

The following table lists all disturb/damp types available for Objects 2 and 3.

Name	Description	Strength controls	Timbre controls	Variation controls
Disturb	A disturb object that is placed at a fixed distance from the string's resting position	The hardness of the object	 The distance from the resting position. Negative values: push the string away from the resting position Positive values: string is unaffected when in the resting position 	 Controls width. Negative values: only a small section of the string is affected Positive values: a broader section of the string is affected
Disturb 2-Sided	This parameter is somewhat akin to a ring placed around the string, that limits the string's vibration in all directions	The hardness of the ring	The clearance of the ring (the distance between the ring and string). Negative values: the sides of the damping ring overlap, influencing the string if any movement occurs Positive values: there is an amount of clearance inside the ring. The string will only be influenced if moved sufficiently to actually touch the ring	No impact
Bouncing	Emulates a loose object laying or bouncing on, and interacting with, the vibrating string. This is very random, by nature, and can't be sync-ed.	Controls the gravity constant for the object laying/bouncing on the string	The stiffness of the object	The damping of the object
Bound	A boundary that limits and reflects string movement. This is much like a fingerboard that limits string movement when the string is plucked very firmly	The distance from the boundary center position to the string's resting position	The slope (steepness) of the boundary. A value of 0.0 places the boundary parallel to the string. Other values will position the boundary closer to the string on one end, and further away on the other.	The amount of reflection at the boundary limits
Mass	Used to model an additional mass attached to the string. This can lead to inharmonic sounds, and very interesting results, if the position of this mass is modulated along the string	The mass size/ weight	No effect	No effect
Damp	Localized damper, which is useful for soft damping.	The intensity of the damping	The damping characteristics	The width of the damped string section

Gate

Determines when the object is active—in other words, when it disturbs or excites the string. Settings are:

- KeyOn: Between note on and note off.
- Always: Between note on and the end of the release phase.
- KeyOff: Triggered at note off, and remaining active until the voice is released.

Note: If using an object type such as Gravity Strike, the note may retrigger when you release the key. To avoid this artifact, set Gate mode to Always.

Strength (Morphable)

The central dial adjusts the intensity of the excitation/disturbance, dependent on type. See table above. A value of 0.0 means no excitation/disturbance at all. In contrast to the On/Off button, however, it is possible to fade in the Strength (intensity of excitation/disturbance) via modulation and/or morphing.

Timbre (Morphable)

The fader to the left of each object's controls is responsible for the timbre (tonal color) of the excitation/disturbance, which is type dependent. See the table above. 0.0 is the normal value for the object. Positive values make the sound brighter, while negative values lead to a more mellow sound.

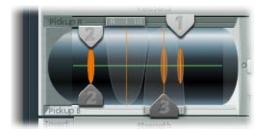
Variation (Morphable)

The Variation slider to the right of each object's controls is an additional timbre parameter, which is again type dependent. Please see the table above for details of its impact on the sound.

VeloSens (Objects 1 and 2 Only)

The excite/disturb objects are velocity sensitive, but this may not be appropriate for all sounds. This parameter, found at the bottom of Objects 1 and 2, allows you to reduce the velocity sensitivity to zero.

Position (Morphable)



Determines the position of each object along the string (A value of 0.0 means one end, and a value of 1.0, the other end of the string).

To adjust, simply click-hold and drag the corresponding numerical slider handle (the 1, 2, or 3 arrows) for each object. Adjustment of these object pickup positions will disturb/excite a given portion of the string.

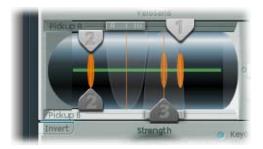
Object 1 can be an exciter. Object 3 can be a damper. You'll note that Object 2 has two arrows. This indicates that this object can be used as either an exciter or damper.

- As you move the object pickups through the Pickup A and B ranges (shown as
 transparent bell curves), you'll find that the intensity of the object disturbance
 increases significantly. This gives rise to a number of changes which can completely
 alter the general timbre of your sound.
- The green horizontal line within the Pickup display represents the string. As the Stiffness of the string is increased, the line will become thicker.
- You can Control-click on the green horizontal line (the string) to activate/deactivate
 string animation. When active, this graphical string will vibrate, making it easier to
 visualize the impact of the objects and pickups. Note that string animation increases
 the CPU overhead, so disable it if your computer is struggling to process all data in
 realtime.
- The vertical orange lines represent the positions of Objects 1, 2, and 3. The thickness and brightness of these lines indicate the strength of the objects.

Processing Parameters

The processing tools covered in this section act on a per-voice basis, as do the string parameters discussed earlier.

Pickup Parameters



- The transparent bell curves represent the position and widths of pickups A and B.
- The green horizontal line within the pickups display represents the string. As the Stiffness of the string is increased, the line will become thicker. The line is animated and will show the range of the string's motion.

• The vertical orange lines represent the positions of disturb/excite objects 1, 2, and 3. The thickness and brightness of these lines increases as the Strength level of each object is raised.

Note: View these pickups as being like the electromagnetic pickups found on an electric guitar. Obviously, changing their positions will alter the tone of your axe, and they'll do the same in Sculpture.

Pickup A Position (Morphable)

The Pickup A slider at the top of the Pickup display determines the position of Pickup A along the string. Just click-hold, and drag, the slider cap to adjust. Values of 0.0 and 1.0 determine the two ends of the string.

Pickup B Position (Morphable)

The Pickup B slider at the bottom of the Pickup display determines the position of Pickup B along the string. Just click-hold, and drag, the slider cap to adjust. Values of 0.0 and 1.0 determine the two ends of the string.

Invert (Pickup B Phase)

The Invert button is found at the bottom left of the Pickup display. Options are: normal or invert(ed).

Note: If the phase of Pickup B is inverted, the sound becomes thinner because portions of the Pickup A and Pickup B signals cancel each other out.

While not found in the actual Pickup display, two further pickup parameters are available next to the Level knob on the left side of Sculpture.



Key Spread

Panning position is determined by MIDI note number. Dependent on settings, the further up/down the keyboard you play, the more the voice is panned left/right. Simply click-drag vertically on the Key button to adjust. Two lines in the ring that surrounds the Spread parameters indicate the values.

In Surround instances, the Surround Range parameter (found in Sculpture's Extended Parameters section) allows you to select the angle range that corresponds to a panning value of ± 1.0 (see "Surround Range and Diversity" on page 484).

Pickup Spread

Spreads the two pickups across the stereo or surround base. In other words, the pickup position, combined with this parameter, will be spread further towards/from the left/right stereo/surround channels. Simply click-drag vertically on the Pickup button to adjust. Two dots in the ring that surrounds the Spread parameters indicate the values.

In Surround instances, the Surround Range parameter (found in Sculpture's Extended Parameter's section) allows you to select the angle range that corresponds to a panning value of ± 1.0 (see "Surround Range and Diversity" on page 484).

Note: You can create animated width and chorus effects by modulating the Pickup Position parameters with an LFO or other modulator.

Amplitude Envelope Parameters

This is a classic analog synthesizer ADSR envelope that scales the pickup signals before passing them on to the Waveshaper and filter. This approach/positioning of the amplitude envelope at this point in the signal path produces more natural sounding results when the Waveshaper is used.



Note: Even with long decay/release times, the sound may decay quickly. This can be caused by high Inner or Media Loss values (in the string material section) or by objects (2 or 3) that are used to damp the string.

Attack—Soft and Hard

The Attack parameter features two sliders. The lower slider determines the attack time, when the keyboard/incoming note data is played at minimum velocity.

The upper Attack slider determines the attack time when the keyboard/incoming note data is played at maximum velocity.

You can adjust both slider halves simultaneously by dragging in the space between them.

Important: The attack time parameters of the amplitude envelope have a major impact on the way a single note is retriggered. When both Attack Soft and Hard are set to value of zero, the vibrating string is retriggered. If either of these parameters is set to a value *above zero*, a new note will be triggered.

Decay

Defines the decay time. The decay time is the amount of time that it takes for the signal to fall to the sustain level, following the initial strike/attack time.

Sustain

This parameter sets the sustain level. The sustain level is held until the key is released.

Release

This parameter determines the length of time that it takes for the signal to fall from the sustain level to a level of zero. Short Release values help to reduce CPU load, as the voice is no longer processed once the release phase has completed.

Waveshaper Parameters

These parameters control and determine any nonlinear processing/polyphonic distortion for the signal, resulting from the panning and level scaling of the two pickup signals.

The Waveshaper provides a nonlinear shaping curve (per voice) that shapes the signal coming from the pickups (and amplitude envelope) and passes this reshaped signal on to the filter. This is quite similar to the waveshaping of oscillators in synthesizers, such as Korg's O1/W.



Waveshaper On/Off Button

The Waveshaper button enables/disables the Waveshaper.

Type

The Type menu above the Waveshaper button allows you to choose different types of waveshaping curves from the list below.

- Soft Saturation
- Vari Drive
- Tube-like distortion
- Scream

Input Scale (Morphable)

This is a bipolar parameter. Negative values attenuate, and positive values amplify, the input signal prior to processing by the Waveshaper. When set positively, this results in a richer harmonic spectrum. The level increase introduced by the parameter is automatically compensated for by the Waveshaper.

Given its impact on the harmonic spectrum, Input Scale should be viewed/used as a timbral control, rather than a level control.

Note: At extreme Input Scale values, processing noise can be introduced at the Waveshaper output.

Variation (Morphable)

The Variation parameter is a bipolar control. Its impact is dependent on the type of Waveshaper selected.

Туре	Variation controls	Value of 0.0	Negative Values	Positive Values
VariDrive	Wet/dry ratio	Provides shaped signal only.	Reduce shaped signal and add dry signal.	Raise shaped signal and add phase inverted dry signal, making sound sharper.
SoftSatTubeDist.Scream	Bias—which alters the symmetry of the shaping curve.	Results in symmetrical shaping.	Alter symmetry.	Alter symmetry.

Filter Parameters

These parameters offer further timbral/spectral control over your sound. They should be pretty familiar to you if you have any experience with synthesizers.

On/Off

The Filter button activates/deactivates the filter.



Filter Type Buttons

The five buttons at the bottom of the filter section determine the filter mode. Choices are:

- *Hipass:* Allows frequencies above the cutoff frequency to pass. As frequencies below the cutoff frequency are suppressed, it's also known as a low cut Filter. The slope of the filter is 12 dB/octave in Highpass mode.
- Lowpass: Allows frequencies which fall below the cutoff frequency to pass. As frequencies above the cutoff frequency are suppressed, it's also known as a high cut filter. The slope of the filter is 12 dB/octave in Lowpass mode.
- *Peak:* This mode allows for an increase in level of a frequency band, the width of which is controlled by the *Resonance* parameter.
- Bandpass: In this mode, only the frequency band directly surrounding the cutoff frequency can pass. All other frequencies are cut. The Resonance parameter controls the width of the frequency band that can pass. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the band.
- *Notch:* In this mode, the frequency band directly surrounding the cutoff frequency is cut. All other frequencies are allowed to pass. The Resonance parameter controls the width of the frequency band that is cut.

Cutoff (Morphable)

Determines the filter cutoff frequency. As an example of its use: In a *lowpass* filter, all frequency portions *above* the cutoff frequency are suppressed, or cut off, hence the name. The cutoff frequency controls the brilliance of the signal. The higher the cutoff frequency is set, the higher the frequencies of signals that are allowed to pass through the lowpass filter.

Resonance (Morphable)

Sets the filter resonance value:

- For highpass and lowpass modes, the Resonance parameter emphasizes the portions of the signal which surround the frequency—as defined by the Cutoff value.
- In Peak, Notch, and Bandpass modes, Resonance controls the bandwidth.

Key

This knob adjusts the key tracking of the cutoff frequency. Put plainly, the further up/down the keyboard you play, the more bright/mellow the sound becomes. Put more technically, the cutoff frequency is modulated by keyboard position.

- A value of 0.0 disables key tracking.
- A value of 1.0 allows the cutoff frequency to proportionately follow the fundamental of the note across the entire keyboard range.

Velo Sens

Determines the velocity sensitivity of the cutoff frequency. The harder you strike the keyboard you're playing, the higher the cutoff frequency (and generally, the brightness of the sound) becomes.

- A value of 0.0 disables velocity sensitivity.
- A value of 1.0 results in maximum velocity sensitivity.

Post Processing

The post processing tools covered in this section impact on the summed signal of *all* voices, rather than on a per-voice basis.

Delay

This is a (project) tempo-syncable stereo or true surround delay. It may also be set to run freely (not synchronized). The Delay section features all the general delay parameters you'd expect from a delay plus the Groove (delay timing) Pad.



General Delay Parameters

The following section covers the general parameters of the Delay section.

On/Off

The Delay button enables or disables the Delay section.

Wet Level

The Wet Level knob sets the level of the Delay output (wet signal). The parameter value is expressed as a percentage (%).

Feedback

Defines the amount of delay signal that is routed back from the output channels (of the delay unit) to the input channels (of the delay unit). Negative values result in phase-inverted feedback.

Xfeed (Crossfeed)

As above, but this parameter defines the amount of delay signal that is fed from the left out to right in and right out to left in (of the delay unit). Negative values result in phase-inverted feedback of the crossfed signal.

In surround instances, the Xfeed control still controls the cross feedback between the delay lines, but offers additional cross feed modes. You can access these in Sculpture's Extended Parameters section:



LoCut

Determines the cutoff frequency of the highpass filter at the delay line output/feedback loop.

HiCut

Determines the cutoff frequency of the lowpass filter at the delay line output/feedback loop.

Input Balance

This parameter allows you to move the stereo center of the Delay input to the left or right, without the loss of any signal components. This makes it ideal for ping-pong delays.

In surround instances, the parameter moves all channels towards the front left or front right channel.

Delay Time

This parameter, coupled with the Sync setting, enables you to set the delay time. This can be in either: musical note values—1/4, 1/4t (t = triplet), and so on—or in milliseconds.

Sync

The Sync button allows you to select either tempo-synced, or tempo-independent delay modes.

Output Width

This parameter allows a reduction of the stereo or surround base of the wet signal.

- A value of 0.0 results in mono output.
- A value of 1.0 results in full stereo or surround output (the left delay line output channels are panned hard left, and the right delay line output channels are panned hard right).

This parameter is used to achieve pure delay grooves, without hard left/right pingpong panning.

The Groove Pad (Stereo)

When used in a stereo instance of Sculpture, the Spread and Groove parameters are combined in the two-dimensional Groove Pad.



Drag the diamond in the center of the crosshair to adjust. You can independently adjust the Spread and Groove parameter values by directly dragging the lines that intersect the diamond.

Spread

Positive values on the y axis (above the default, centered position) increase the delay time of the right delay line and decrease the delay time of the left delay line—in effect, smearing the delay times of the left and right channels. Negative values reverse this. Spread is useful for wide stereo delay effects.

Groove

This parameter (on the x axis) allows you to reduce the delay time of one delay line by a given percentage, while keeping the other delay line constant. Basically, Groove distributes the delay taps to the left/right channels, rather than smearing them, like Spread. Keep an eye on the small help tag while adjusting.

As an example, a value of +50% reduces the right delay time by half. If a value of 1/4 was used as the Delay Time, the right delay would equal 1/8th of a note and the left delay would remain at 1/4 of a note. Needless to say, this parameter is perfect for the creation of interesting rhythmic delays—in stereo.

Note: You can create some truly wide chorus and modulated delay effects by modulating the Pickup Position and Pickup Spread parameters (with an LFO or other modulator), and then feeding this into the delay unit.

The Groove Pad (Surround)

When used in a surround instance of Sculpture, the Delay Time Pad converts into a pure groove pad that controls the delay time relationship between:

- Left and right channels (speakers) in the horizontal direction
- Front and rear channels (speakers) in the vertical direction.

The Spread parameter is accessible separately as a numerical edit field at the top-left of the pad. Click-drag, or double-click and type in, to alter the value.



Clearing and Copying Delay Parameters

You can Control-click on the Delay Time Pad to access a shortcut menu which offers Clear, Copy, and Paste commands for delay settings. These can be used to copy and paste delay settings between multiple Sculpture instances, or between consecutively loaded settings. The Clear option resets the current delay settings.

Body EQ

The Body EQ can work as a simple EQ, as a complex spectral shaper, or as body response simulator. In effect, the Body EQ can emulate the resonant characteristics of a wooden or metallic body—such as that of a guitar or violin.



The various models are derived from impulse response recordings of instrument bodies. These recordings have been separated into their general formant structure and fine structure, allowing you to alter these properties separately.

Body EQ On/Off Button

This button (to the left of the image) enables/disables the spectral shaping section.

Model

The Model menu allows you to choose the Body EQ model. You can choose between various emulations of acoustic instrument bodies and the Basic EQ model. Any selection will be reflected in the graphic display to the right.

Basic EQ Controls

The Basic EQ features different parameters than other EQ models. Please see the following section for details other model parameters.



- Low: Gain of a low shelving filter.
- Mid: Gain of a peak filter (sweepable—see below).
- High: Gain of a high shelving filter.
- *Mid Frequency slider:* Allows you to sweep the center frequency of the mid band between 100 Hz and 10 kHz.

Adjusting the Basic EQ Graphically

You can directly control the Lo Mid Hi model by click-dragging in the Body EQ graph:

- Click-dragging vertically on the left third of the graph allows you to control the Low parameter.
- Click-dragging vertically on the center third of the graph allows you to control the Mid parameter.
- Click-dragging horizontally on the center third of the graph allows you to control the Mid Frequency parameter.
- Click-dragging vertically on the right third of the graph allows you to control the Hi parameter.

Other EO Models

For all other Body EQ models you have the following parameters:



Formant—Intensity

Scales the intensity of the model's formants. In other words, any formants (harmonics) in the model will become louder, or will be inverted, dependent on how this parameter is used.

- A value of 0.0 results in a flat response.
- A value of 1.0 results in strong formants.
- Negative values invert the formants.

Formant—Shift

This parameter shifts the formants logarithmically. A value of -0.3, for example, shifts all formants one octave downwards, and a value of +0.3 shifts the formants up one octave. A value of +1.0 shifts up by a factor of 10—from 500 Hz to 5000 Hz, for example.

Formant—Stretch

Stretches the formant frequencies, relative to each other. In other words, this parameter alters the width of all bands being processed by the Body EQ, extending or narrowing the frequency range.

Low Formant Stretch values move the formants closer together (centered around 1 kHz) while high values move the formants further apart from each other. The control range is expressed as a ratio of the overall bandwidth.

When combined, Formant Stretch and Formant Shift alter the formant structure of the sound, and can result in some interesting timbral changes.

Fine Structure

This parameter enhances the spectral (harmonic) fine structure, making the overall harmonic makeup of the sound more precise. This results in a more detailed sound that is harmonically richer and—dependent on the model selected—more guitar or violin-like, for example. Put another way, the resonant cavities of the instrument become more resonant—somewhat like the increased depth of tone provided by a large-bodied guitar.

- A value of 0.0 denotes no fine structure.
- A value of 1.0 results in enhanced/full fine structure of the selected model.

Please note that heavy use of Fine Structure may be quite CPU intensive.

You should also note that the use of Fine Structure may not actually result in too much of a difference to your sound. This is dependent on the Waveshaper and Body EQ modes, plus other string parameter settings. As always, use your ears!

Adjusting Other Models Graphically

You can directly control the model by click-dragging in the Body EQ graph:

- Click-dragging vertically on the graph allows you to control the Formant Intensity parameter.
- Click-dragging horizontally on the graph allows you to control the Formant Shift parameter.

Surround Range and Diversity

In surround instances, Sculpture's extended parameters feature the Surround Range and Surround Diversity parameters:



- Surround Range: Determines the range of the surround angle. Put another way, this determines the breadth of the surround field. To better explain its use: Imagine an LFO routed to a pickup's pan position with an amount of 1.0. Setting the LFO waveform to sawtooth, and the Surround Range to 360 results in circular movement (around the entire surround circle) of the voice output. The Surround Range parameter also influences the Key and Pickup spread in the same way.
- Surround Diversity: Determines how the output signal is spread across your surround speakers. If you choose a value of 0, only the speakers closest to the original signal's position will carry the signal. A diversity of 1 means that all speakers will carry an identical amount of the signal.

Level Limiter



- Level: Controls the overall output level for the instrument.
- Level Limiter mode: Clicking on the desired button activates or deactivates the integrated limiter. Options are:
 - Off: Disables the limiter.
 - Mono: A monophonic limiter on the summed signal of all voices.
 - Poly: A polyphonic limiter, that processes each voice independently.
 - Both: A combination of both limiter types.

This function is very useful for domesticating some of the more aggressive aspects of Sculpture's component modeling synthesis engine.

Modulation Generators

Sculpture features a comprehensive collection of modulation generators. There are:





- Two assignable LFOs, with (project) tempo-syncable rates
- Random variations can be created via two jitter generators (with adjustable bandwidth).
- An additional LFO that is hard wired to vibrato.
- Two Randomizers that only change values at note start/on.

• Two control envelopes can either be used as standard envelopes or as MIDI controlled modulators—with the ability to record, polyphonically play back (on a per-voice basis), and modify incoming MIDI controller movements.

All modulation assignments take place within the generators.

First order modulations are where the modulation generator modulates core synthesis parameters. Second order modulations are where one modulation generator modulates the parameters of the other modulation generator. These include: LFO Rate Modulation, VariMod, Morph Envelope Modulation, and A Time Velosens.

Each modulation generator allows you to select one (or in most cases, two) of the core synthesis parameters as a modulation *target*. The same target can be selected for all modulators, if desired.

Modules such as the LFOs and control envelopes also offer *via* modulations (a freely modulatable amount/intensity—the scaling factor—of the modulation source output level). These are *sidechain* modulations.

Some modulation generator parameters—for example LFO rates, can be modulated by selecting the desired modulation *source* and amount.

To access the desired modulation generator, click on the corresponding button in this section of the Sculpture interface. Once a modulation source has been activated, the corresponding button label will be lit.

LFO 1 and 2

The two full-featured LFOs offer several possibilities that go beyond the Vibrato LFO and jitter generators described below.



Waveform

This menu allows you to choose the waveform used for LFO modulation. Options are: Sine, Triangle, Sawtooth, Rectangle Unipolar, Rectangle Bipolar, Sample&Hold, Sample&Hold with Lag, Filtered Noise.

• The triangular wave is well suited for vibrato effects.

- The sawtooth is well suited for helicopter and space gun sounds. Intense modulations of incoming frequencies leads to bubbling and boiling, underwater sounds. Intense sawtooth modulations of lowpass filters create rhythmic effects.
- The rectangular waves make the LFO periodically switch between two values (as an
 example—a positive value and zero—Unipolar). The Bipolar Rectangular wave
 switches between a positive and a negative value set to the same amount above or
 below zero.
- The two Sample&Hold waveform settings output *random* values. A random value is selected at regular intervals, defined by the LFO rate. A modulation of pitch leads to the effect commonly referred to as a random pitch pattern generator or sample and hold. Check out very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies! If the Sample&Hold with Lag setting is used, the random wave is delayed, resulting in a fluid changing of values.

Note: The term Sample & Hold (abbreviation—S & H) refers to the procedure of taking samples from a noise signal at regular intervals. The voltage values of these samples are then *held* until the next *sample* is taken. When converting analog audio signals into digital signals, a similar procedure takes place: Samples of the voltage of the analog audio signal are taken at the rate of the sampling frequency.

Rate

The Rate knob determines the rate of LFO modulation, which can be either synced to the current project tempo or set independently in Hz (Hertz) values.

- In Hz mode, the rates range from DC (Direct Current) to 100 Hz.
- In Sync mode, rates range from a periodic duration of 32 bars through to speeds of 1/64-triplets. Triolic and punctuated values are also possible.

The LFOs are ideally suited for rhythmic effects which retain perfect synchronicity even during tempo changes to the project.

Sync/Free Buttons

Part of the Rate parameter, these buttons allow you to select either synchronized or free-running LFO rates. When toggling between modes the value is derived from the project tempo and meter.

Curve

Allows you to define a freely-variable number of waveform variations, resulting in subtle or drastic changes to your modulation waveforms.

The Curve parameter can even influence the sine waveform type.

- Curve value of 0.0: Pure sine wave
- Curve values above 0.0: Wave is smoothly changed into a nearly rectangular wave.

• Curve values below 0.0: The slope at the zero crossing is reduced, resulting in shorter soft pulses to +1 and -1.

Note: The waveform displayed between the Curve knob and the Waveform menu shows the results of these two parameter settings.

Envelope

A simple LFO envelope control that allows either:

- Constant modulation
- A ramping up to the full amount
- A decay to a zero amplitude level

Phase

Allows the choice between strictly monophonic or polyphonic LFO modulations with either similar phases, completely random phase relationships, key-synced phase ... or anything in-between.

- If used polyphonically for modulation of multiple voices, the modulations will *not* be phase-locked. To explain, when used on polyphonic input (a chord played on the keyboard) the modulation is independent for each voice (note). Where the pitch (for example) of one voice may rise, the pitch of another voice might fall and the pitch of a third voice may reach its minimum value.
- When used monophonically, the pitch of all voices will rise and fall synchronously.
- If used randomly, some notes will be modulated synchronously, and others won't.

Note: If you move the Phase knob slightly away from the mono position, you'll get non-locked modulations for all voices running at similar, but not identical, phases. This is ideal for string-section vibratos!

RateMod Source and Amount

The LFO rate can be modulated by the source and amount that you determine with this menu and vertical slider (found to the right of the above mentioned LFO controls).



Target and Via Modulations

Two target modulation destinations can be assigned per LFO, with an optional, additional via modulation. To activate, click on either the 1 or 2 buttons (which will highlight the Target and "via" menus), then choose the desired target from the Target menu.



The "via" menus determine the source that controls the modulation scaling for each LFO.

Modulation Amount Sliders: Amt and Via (Amount)

In cases where the via source is set to off, only one amount slider is visible (the "via (amount)" slider is hidden):

• Amt 1/2: The Amt slider determines the modulation amount.

In cases where any via source other than off is selected, there are two sliders:

- Amt 1/2: The Amt slider determines the modulation amount in cases where the incoming via signal is zero, for example a modulation wheel at its minimum position.
- *Via* (*Amount*) 1/2: The "via (amount)" slider determines the modulation amount in cases where the incoming via signal is at full level, for example a modulation wheel at its maximum position.

Vibrato

One LFO is hard-wired to pitch, for vibrato effects. The strength of the vibrato effect can be adjusted via the MIDI controller assigned in the VibDepth Ctrl menu. This is set in the MIDI Controller Assignment section. More information can be found in the "MIDI Controller Assignments" on page 509.



Waveform Menu

Allows you to choose the waveform used for vibrato, for example sine, triangle, sawtooth, and so on.

There are two special rectangular waves: Rect01 and Rect1—the former switching between values of 0.0 and 1.0 (unipolar), and the latter between values of -1.0 and +1.0 (bipolar, like the other waveforms). See "LFO 1 and 2" on page 486.

Curve

Allows you to define a freely-variable number of waveform variations, resulting in subtle/drastic changes to your modulation waveforms.

The Curve parameter can even influence the Sine waveform type.

- Curve value of 0.0: Pure sine wave
- Curve values above 0.0: Wave is smoothly changed into a nearly rectangular wave.
- Curve values below 0.0: The slope at the zero crossing is reduced, resulting in shorter soft pulses to +1 and -1.

Note: The waveform displayed between the Curve knob and the Waveform menu shows the results of these two parameter settings.

Phase

Allows the choice between strictly monophonic or polyphonic vibrato with either similar phases, completely random phase relationships, key-synced phase—or any value in-between. See "LFO 1 and 2" on page 486, for more details.

Rate

Determines the rate of vibrato, which can be either synced to the current project tempo or set independently in Hz (Hertz) values. See "LFO 1 and 2" on page 486, for more details.

Depth via Vib Ctrl—Max

This slider determines the maximum possible modulation amount.

Depth via Vib Ctrl—Min

This slider determines the minimum possible modulation amount.

Random Variations

Many sounds can benefit from the use of random modulations to parameters. These can emulate small variations that occur when particular instruments are played.

Jitter 1 and Jitter 2

The two jitter generators are special LFOs, designed to produce continuous, random variations—such as those of smooth bow position changes. The jitter generators are equivalent to general purpose LFOs set to a noise waveform.

To activate the routing of the jitter generators, click on the 1 or 2 buttons.

Note: Jitter modulation of pickup positions as the target produces great chorus-like effects.



- *Rate knobs:* These knobs set the speed of the modulation (jitter) signal for each respective jitter generator.
- Target 1 and 2: These menus define modulation destinations 1 and 2.
- Amount 1 and 2: These sliders determine the amount of modulation for each jitter generator.

Note On Random

The two note on random generators are intended for random variations between different notes/voices. Their values are randomized for each note, and remain constant until the voice is released. Such randomizations are useful for adding interest/ thickening the sound when played polyphonically. It is also useful for emulating the random variations a player introduces when playing an instrument—even when repeating the same note.

To activate the routing of the Note On Random modulators, click on the 1 or 2 buttons.



- *Target:* Determines the modulation destination—what parameter will be randomly modulated when a note is played.
- Amount: Sets the modulation amount—the strength of the modulation.

Velocity Modulations

The excite objects and the filter have dedicated velocity sensitivity controls. Many other modulation routings also allow you to select velocity as a via input source.

In some cases, it may be of use to directly control other synthesis core parameters by velocity. This can be done in this section—where two independent destination/amount/velocity curve slots are available. To activate either, click on the 1 or 2 buttons.



- Target 1/2: Click the respective Target menu to choose the destination parameter that you wish to modulate by velocity.
- Amount 1/2: The slider below each target determines the amount of modulation.
- Curve 1/2: Click on the appropriate radio button to choose from concave, linear, and convex velocity curves.

Controller A and B

These parameters allow you to define two discrete modulation targets, and the strength of modulation for both Controller A and B.

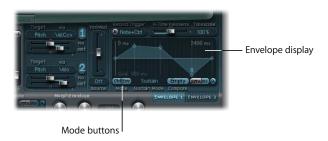


To use, simply click on the 1 or 2 buttons, select the desired target (and Target mode), and adjust the Intensity slider.

Each target features a two-state button:

- Continue: Continuous modulation
- Note On: Modulation value is only updated when a note on message is received.

The Control Envelopes



The two control envelopes are somewhat special, as they can be used as:

- Traditional four segment envelopes
- MIDI controller modulations
- A combination of both: as MIDI controller movement *recorders* (with ADSR-like macro parameters), for polyphonic playback.

The mode buttons (Ctrl and Env) in the lower left corner of the envelope display selects either controller (run mode) or envelope functionality. If both are activated, the controller value is *added* to the envelope output, resulting in a modulation offset.

The envelopes can act as polyphonic modulation recorders and playback units. Each voice is handled independently, with a separate envelope being triggered as each note is played.

To select Envelope 1 or 2, click on the Envelope 1 or Envelope 2 button.

Modulation Routing

As in the LFO section, each envelope offers two modulation target selectors with amount and via amount controls, and a separate via modulation option.

The following target/amount/via settings are available for all run modes.

- Target 1 and 2: Two modulation targets can be assigned per envelope, with an optional, additional *via* modulation. These parameters determine modulation destinations 1 and 2. Either the 1 or 2 buttons must be clicked in order to activate these modulation options. A large number of possible targets are available, including; string, object, pickup, waveshaper, and filter parameters. To choose, simply click on the button below the word Target, and make your choice from the list.
- *Via (source) 1/2:* The "via" parameter defines the modulation amount for Envelopes 1 and 2. The "via" menus allow the selection of sources that are used to scale the modulation amount of the envelopes.
- Amt 1/2 and Via (Amount) sliders: The Amt slider determines the modulation amount. In cases where any via source other than off is selected, there are two sliders. The Amt slider determines the modulation amount in cases where the incoming via signal is zero, for example a modulation wheel at its minimum position. The "via (amount)" slider determines the modulation amount in cases where the incoming via signal is at full level, for example a modulation wheel at its maximum position.

Envelope Display

The envelope curve is shown in the display to the lower right of the Sculpture interface. The envelope display is only active if the envelope functionality is engaged (Mode set to either Env or Ctrl+Env).



• The overall time/length of the envelope is indicated by the numerical entry at the top right of the window (2400 ms in the graphic).

- The maximum time/length of the envelope is 48 bars/40 seconds.
- The lines on the background grid are placed 100 milliseconds apart.
- The background lines are placed 1000 ms apart for very long displayed envelope times. In sync mode, this is displayed as 1 quarter.
- The envelope is *zoomed automatically* after releasing the mouse button. This allows the display of the entire envelope at the highest possible resolution for the graphic envelope display.
- This behavior can be disabled/enabled by clicking the Autozoom button—the small magnifying glass.
- Autozoom is automatically disabled when you perform a manual zoom—by click-holding on the envelope display background, and dragging horizontally. As a reference, the current display width is displayed by the numerical entry at the top right of the display. You can re-engage automatic zooming by clicking on the Autozoom button.
- If you click on the handles (nodes) or lines between the nodes in the envelope display, the current envelope segment will be highlighted. A small help tag also indicates the millisecond value of the current segment.

Envelope Handling

When first opened, a default envelope curve is automatically created for each envelope. To view either, click the Env button in the Mode section.

You will see a few handles (nodes) placed—from left to right—along a straight line within the envelope. These are indicators of the following parameters.

- Node 1: Start level
- *Node 2:* Attack time position/level
- *Node 3:* Loop time position/level (can be freely positioned).
- *Node 4*: Sustain time position/level (can be freely positioned).
- Node 5: End time position/level.

As you move your mouse cursor along the line, or hover over the nodes directly, the current envelope segment is highlighted.

You can create your own envelopes manually, by manipulating the nodes and lines, or you may record an envelope, as discussed in "Recording an Envelope" on page 497.

To adjust the time between nodes, click on the desired handle, and drag it left or right. As you do so, the overall length of the envelope will change—with all following nodes being moved.

You cannot move a node beyond the position of the preceding node. You can, however, move nodes beyond the position of the following node—even beyond the right-hand side of the envelope display—effectively lengthening both the envelope segment and the overall envelope.

When you release the mouse button, the envelope display will automatically zoom to show the entire envelope.

To adjust the *level* of each node, click on the desired handle, and drag it up or down.

To adjust the *shape of the curve* between nodes, click directly on the line that connects them, and drag it up or down.

In the case of recorded envelopes, you may have a more complex curve between nodes. To adjust, Control-click and drag the curve.

Experiment with each node/parameter to get a feel for its operation. You'll find that the envelopes are very intuitive to use.

Copying Envelopes

If you Control-click on the envelope buttons or on the envelope display's background, a shortcut menu opens, offering Copy, Paste, and Clear commands. These allow you to copy and paste (or clear) envelopes between Envelopes 1 and 2, between settings or multiple open Sculpture instances.

Envelope Recording

Before proceeding, it is important to note that you can only record MIDI controller movements of the assigned MIDI controller. MIDI controller assignments for the envelopes must be set in the MIDI Controller Assignment section at the bottom of the Sculpture interface:



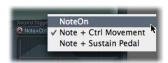
R(ecord) Button

Enables the record functionality for the envelope. This button works in a similar fashion to the record arm buttons in Logic Pro. To stop recording, simply click on the R button a second time, or use the trigger mode functionality described below.



Record Trigger Mode

The menu to the right of the R button is used to choose different record trigger modes to start recording (when R(ecord) is active):



- NoteOn: Recording starts when a note is played.
- Note+ Ctrl Movement: Recording starts when MIDI control change messages (for the assigned controllers—see "MIDI Controller Assignments" on page 509) arrive while a note is held.
- *Note* + *Sustain Pedal:* Recording starts when the sustain pedal is depressed while a note is held.

Recording an Envelope

To provide you with an envelope recording example:

- Set the Record Trigger Mode to Note+Ctrl.
- Enable record by clicking on the R button.
- Play, and hold, a key—and start moving the modwheel or whatever controllers are assigned to envelope controls 1 and/or 2.

Ending an Envelope Recording

An envelope recording ends as soon as at least one of the following conditions is met:

- The record button (R) is disengaged manually by clicking on it.
- All *voices* are released.
- A new note is played after releasing all keys.

Following the recording of a controller movement, R(ecord) is automatically set to off and Mode is set to Env. This ensures that only the recorded movement will be active, regardless of the stop position of the recorded controller.

Playing Back a Recorded Envelope

Polyphonic playback of the recorded envelope occurs when you play a key. The Mode parameter must be set to Env and the R(ecord) parameter must be set to off.

You can also activate both the Env and Ctrl buttons of the Mode parameter, as this will allow you to use controllers assigned to Ctrl Env1 or Ctrl Env2 to manipulate the envelope in real time, alongside playback of the recorded envelope.

Note that if both Env and Ctrl are activated, however, the controller value is added to the envelope output, resulting in a modulation offset.

Preparing the Recorded Envelope for Editing

The envelope segments and handles are set automatically after recording. To change the interpretation of the envelope you can grab and drag the vertical lines that intersect the handles (also see the VariMod paragraph below). Note that this will not change the shape of the envelope.

Envelope Parameters

The following parameters are only active if the envelope functionality is engaged (Mode set to either Env or Ctrl+Env).

A-Time Velosens

This slider is used to set the velocity sensitivity for the attack time of the envelope. Positive values will make the attack time shorter at minimum velocities, and negative values will make the attack time shorter at maximum velocities.



Timescale

This parameter allows you to scale the duration of the entire envelope between 10% (ten times faster) and 1000% (ten times slower). This will also visually impact on the appearance of the envelope curve displayed as it is shortened (sped up) or lengthened (slowed down).

Sustain Mode Menu

Allows you to define the behavior of the envelope while a note is held. Choices are: The usual Sustain mode, Finish mode, or one of three loop modes (Loop Forward, Loop Backward, Loop Alternate).

The envelope can—like any envelope—run in one shot (as long as the note is sustained). It can also run several times or in an infinite cycle, much like an LFO. You can achieve this through the use of loops.

When in any of the loop modes, the loop always cycles between user-defined envelope handles that indicate the loop start point (L icon), and the sustain point (S icon). These handles can be dragged to the desired position.

- When set to Finish, the envelope runs in one shot mode from its beginning to its end—even if the note is released before the envelope has come to its end. The other loop parameters are disabled.
- When set to Loop Forward, the envelope runs to the sustain point and begins to repeat the section between the loop point and sustain point periodically—always in a forward direction.
- When set to Loop Backward, the envelope runs to the sustain point and begins to repeat the section between the sustain point and loop point periodically—always in a backward direction.
- When set to Loop Alternate, the envelope runs to the sustain point and returns to the loop point and back to the sustain point periodically, alternating in both a backward and forward direction.

Note: If the loop point lies behind the sustain point, the loop will start after the key has been released.

Loops can be synchronized to the project tempo automatically via the "sync" and "ms" buttons.

Sync and "ms" Buttons

These buttons allow you to select between a free-running envelope (with segment times displayed in milliseconds) or a tempo-synced envelope with note value options, such as 1/8th or 1/4.

Switching between values forces a recalculation of times to the nearest note value or ms time, respectively, based on the current project tempo.

Compare Button

Following the initial (original) recording of an envelope, and any subsequent edits, this button allows you to toggle between the original recording and the edited version. Obviously, this is only available as an option if an envelope curve has actually been recorded.



VariMod—Source and Amount

VariMod is only available for recorded envelopes. It allows you to select a modulation source, and amount, to control the strength of an envelopes variation.

Variation in the envelopes means the deviation of a recorded envelope path from straight interconnecting lines between the points. Once you have recorded an envelope, you can reduce or exaggerate the amplitude-jitter (variation) of the recording by Command-dragging the curves between points down (to reduce) or up (to exaggerate).

To adjust the amount of variation modulation, simply move the VariMod slider up or down.

Source options can be accessed by clicking on the Source button, and choosing one of the following from the menu: Off, Velocity Concave, Velocity, Velocity Convex, KeyScale, Ctrl A, and Ctrl B.



Morphing

Sculpture includes morph functionality that allows you to smoothly change the sound—in a subtle or radical way—between up to five snapshots, called morph points.

Each morph point includes separate settings for more than 20 important synthesis parameters, including string material parameters, parameters and positions of the excite or disturb objects, pickup positions, filter, and Waveshaper parameters.

All morphable parameters can be automated independently for each morph point.

The current morph point position within the *Morph Pad* can be controlled via MIDI controllers (for example by a vector stick). Such movements can be recorded and played back independently—each voice can be morphed differently.

The morph section consists of two parts:

- The *Morph Pad*, featuring five morph points (center and four corners), plus options for randomizing (via the randomize parameters) and coping or pasting morph points and Morph Pad states via a shortcut menu.
- The *morph envelope*, that can be edited either by segment (with the mouse), or recorded via MIDI controller movements. With a vector stick (Morph X/Y controllers) or mouse movements of the morph cursor (the ball), for example, on the Morph Pad.



Selecting Morph Points

There is always *one* of the five Morph Pad points (A/B/C/D/Center) that is selected for editing. This selected point is indicated by two concentric circles that surround it.



All morphable parameters (all parameters which have an orange value bar, rather than a blue or turquoise one) are shown, allowing you to edit the values of the *selected* Morph Pad point.

When you active Auto Select mode, movements of the morph cursor automatically select the nearest morph point.

You can also click into the circles around A, B, C, D, or Option to select a Morph Pad point manually.

Randomizing Morph Points

The randomize feature allows you to create random variations of selected morph points. When combined with the copy/paste functionality that's also available, randomizing lends itself to using the Morph Pad as a kind of sound cell culture device.

Use of the Morph Pad can yield an interesting composite sound. You can copy this sound to a corner of the Morph Pad (or several corners) and randomize it by a definable amount.

The morphed sound then becomes a new timbral element, creating permutations that can, in turn, be moved to the corners, randomized, and so on ...

In effect what you are doing is "breeding" a sound, while maintaining some control by selecting parent and child sounds.

This approach can result in new, complex sounds without the need to be a programming guru.

Randomization Parameters

The following parameters allow you to define your random variations.



- *Points buttons:* Selects the number of morph points that are to be used for randomization, and indicates which points will be randomized. The bottom button, when selected, limits randomization to the currently selected morph point.
- Randomize button: Click to generate a randomized collection of morph points.
- Int(ensity) slider: Determines the grade of randomization from 1% (slight deviation) to 100% (completely random values).

Using Randomization

The following example will show you how to randomize morph points.

To use randomization:

- 1 Select the top button (five points) in the Points section.
- 2 Ensure that Auto Select is active.
- 3 Set the Int(ensity) slider to a value of around 25%.
- 4 Click the Rnd button.

As you're doing so, keep an eye on the parameters in the core synthesis engine. You will see a number of them move after the mouse click.

- 5 Click-hold on the morph cursor (the ball), and drag it to each of the corners in the Morph Pad. Do this along the edges, as well as through the center of the Morph Pad, and take a mental note of how this impacts on the morph.
- 6 Don't forget to strike a few notes on your MIDI keyboard while doing so.

Note: The morph ball is only visible when the Record Trigger button is active.

As you're moving the morph cursor around, you'll see the real-time movement of the ghost controls in the Pickup display, the ghost ball in the Material Pad, and—if you look closely—you'll also notice a number of red dots moving about in the various string and object parameters.

Each of these indicates the current morph position. This is a handy tool for seeing what parameters have changed, and how they have been changed.

You'll also note that positions on the Morph Pad that fall in-between the various morph points cause the randomized parameters to interpolate between values—which is what morphing is all about.

Check out the next section on the Copy and Paste commands, as these will help you to make use of those in-between settings.

Morph Menu Commands

You can access the Morph shortcut menu by Control-clicking on the Morph Pad.



Copy and Paste Commands

- Copy selected Point: Copies the current morph point into Sculpture's Clipboard.
- Copy current Pad Position: Copies the current morph state into Sculpture's Clipboard.
- Paste to selected Point: Pastes the Clipboard content to the selected point.
- Exchange selected Point: Swaps previously copied data with another point of your choice.
- Paste to all Points: Pastes the Clipboard content to all selected points.

Other Commands

The remaining commands in the Morph Pad shortcut menu are to do with grouping of random parameters. Put another way, the following commands allow you to determine which type of parameters you would like to randomize (via the Rnd button and Int(ensity)) slider.

- All (Random Group): This is your ticket to wacky sounds as all parameters in the
 following three groups are randomized. This can lead to some interesting results, but
 can be uncontrolled. This is less useful for the "sound cell culture" idea discussed in
 the Randomizing section above.
- *All except TensionMod:* Basically the same as All (Random Group) but excluding the somewhat dangerous TensionMod parameter from randomization.
- String Material + Media: In isolation, including Material Pad position, Stiffness, Inner Loss, Media Loss, Resolution, and Tension Modulation.
- *Objects + Pickups:* Alters the positions of objects and pickups, plus the various object parameters.
- Filter + Waveshaper: Randomizes the positions of all Waveshaper and filter parameters.

Morph Envelope Display

The morph envelope offers nine points/eight segments, and recording functionality that is much like that of the controller envelopes.

In the diagram below, you will see a selected point in the lower panel (the Timeline) and a corresponding selected point handle of the trajectory in the Morph Pad. These are shown as an orange diamond in the Timeline and an orange diamond (or ball) in the Morph Pad.



- The overall time/length of the morph envelope is indicated by the numerical entry at the top right of the display.
- The maximum time/length of the morph envelope is 48 bars/40 seconds.
- The lines on the background grid are placed 100 milliseconds apart.
- If you click on the handles (nodes) or lines between the nodes in the envelope display, the current envelope segment will be highlighted. A small help tag also indicates the millisecond value of the current segment.

- As you move your mouse cursor along the line, or hover over the nodes directly, the current envelope segment is highlighted.
- You can create your own envelopes manually, by manipulating the nodes and lines, or you may record an envelope, as discussed in "Recording Morph Envelopes" on page 505.
- To adjust the *time* between nodes, click on the desired handle, and drag it left or right. As you do so, the overall length of the morph envelope will change—with all following nodes—being moved.

You cannot move a node beyond the position of the preceding node. You can, however, move nodes beyond the position of the following node—even beyond the right-hand side of the envelope display—effectively lengthening both the envelope segment and the overall envelope.

Note: Sculpture's Morph Pad displays a moving dot that indicates the current morph position, during a morph on a *monophonically* played sound. The red line in the Timeline below shows the current time position.

Recording Morph Envelopes

The following section outlines the parameters needed to record a morph envelope.



R(ecord) Enable Button

The R button enables the morph envelope record functionality. This button works in a similar fashion to the record arm buttons in Logic Pro.

To make a morph envelope recording, just click the R button, play a note, and start moving the silver ball in the Morph Pad with the mouse. You can also make use of an external controller (see "Morph X and Morph Y Menus" on page 510).

The mode is automatically set to Pad as soon as the R button is clicked (for more information on modes, see "Morph Envelope Parameters" on page 506).

Record Trigger Mode

The menu to the right of the R button is used to select different trigger modes to start recording (when R(ecord) Enable is active):

- NoteOn: Recording starts when a note is played.
- Note + Move Morph Point: Recording starts when MIDI control change messages (as assigned in the Morph X and Y parameters of the MIDI Controller Assign section) arrive while a note is held.

Note + Sustain Pedal: Recording starts when the sustain pedal is depressed while a
note is held.

Recording is stopped by clicking the R(ecord) Enable button (or trigger) a second time. Once all keys are released, and all voices have completed their decay phase, the recording ends. You can stop recording early by releasing all keys, and then pressing a single key.

Following the recording of a controller movement, R(ecord) Enable is automatically set to off and Mode is set to Env only. This ensures that only the recorded movement will be active, regardless of the stop position of the recorded controller.

Morph Envelope Parameters

The following section outlines the morph envelope's parameters.



Mode

These buttons activate the morph envelope, and allow you to choose from the following modes:

- Both buttons off: Morph functionality is disabled.
- *Pad only:* Envelope is deactivated, and morph functionality is controlled by the morph cursor (the ball) or X/Y MIDI controllers only.
- Env only: Envelope is running, but the morph cursor and X/Y MIDI controllers are deactivated.
- Env + Pad: Envelope is running, and the position of the morph cursor or X/Y MIDI controllers is used as an offset for any envelope movements. In this mode (with both the Pad and Env buttons activated), you can choose from the following setting by clicking the offset label next to the mode buttons:



- Offset: The default mode, behavior is as described in the Env+Pad mode described above).
- Point Set: Envelope is running. The selected envelope point can be edited by
 moving the morph cursor in the pad or via MIDI (MorphX and MorphY Controller
 Assignments).
- *Point Solo:* Envelope in a kind of "still picture" mode. The selected envelope point can be edited by moving the morph cursor in the pad.

Time Scale

This parameter scales the duration of the entire envelope between 10% and 1000%.

Sustain Mode Menu

Allows you to select the behavior of the morph envelope while a note is held. Choices are: Sustain mode, Finish mode, one of three loop modes (forward, backward, alternate), or Scan via CtrlB mode.



When in any of the loop modes, the loop always cycles between the loop and sustain envelope handles (the nodes indicated by the small L and S icons). The morph envelope can—like any envelope—run in one shot (as long as the note is sustained). It can also run several times or in an infinite cycle, much like an LFO. You can achieve this through the use of loops.



The loop and sustain point handles can be directly grabbed and repositioned. Note that doing so can potentially alter the loop (and the overall morph envelope) length.

- When set to Finish, the morph envelope runs in one shot mode from its beginning to its end—even if the note is released before the envelope has come to its end point. The other loop parameters are disabled.
- When set to Forward, the envelope runs to the sustain point and begins to repeat the section between the loop point and sustain point periodically—always in a forward direction.
- When set to Backward, the envelope runs to the sustain point and begins to repeat the section between the sustain point and loop start point periodically—always in a backward direction.
- When set to Alternate, the envelope runs to the sustain point and returns to the loop point and back to the sustain point periodically, alternating in both a backward and forward direction.
- When set to Scan via CtrlB, the time line position within the envelope is "disconnected" from real time, and you can "scan" the whole time range by using the MIDI controller chosen for Ctrl B (in the MIDI Controller Assign section).

Dip: It is also possible to manually drag the red time position marker with the mouse.

Note: If one of the three loop modes is selected, and the loop point is positioned before the sustain point, the loop will be active until the key is released. Following key release, the envelope will then continue beyond the sustain point, as per usual. If the loop point is positioned after the sustain point, the loop will be entered as soon as the key is released, and will be cycled infinitely (until the complete voice has finished the amplitude envelope release phase).

Sync and "ms" Buttons

These buttons allow you to select between a free-running envelope (with segment times displayed in milliseconds) or a tempo-synced envelope with note value options, such as 1/8th or 1/4.

Switching between values forces a recalculation of times to the nearest note value or ms time, respectively, based on the current project tempo.

Depth Knob

This parameter scales the amount of morph movement caused by the morph envelope. The effect of the Depth parameter is visually displayed in the Morph Pad. As you increase or decrease the value, the morph trajectory will also be scaled.

Modulation Knob and Source Menu

These parameters allow you to select a modulation source and amount, which are used to scale the movement of the morph envelope.

Transition Knob

Provides control over the transitions between the morph points. This can range from the original (possibly recorded) movement to linear connections, and beyond this to stepped transitions.

The latter means: remaining at one morph state during the entire morph envelope segment, and then abruptly switching to the morph state set at the following envelope point.

This parameter (and the morph envelope itself) can lead to interesting evolving sounds or even rhythmic patches.

Morph Position Display

The red line in the morph envelope's timeline shows the current time position, during a morph. Sculpture's Morph Pad displays a moving dot that indicates the current morph position.



Note: The current morph position is shown as long as only one note is played.

MIDI Controller Assignments

This section allows you to define the MIDI controllers you wish to use for vibrato depth control or morph pad movements, for example. These parameters are saved with each setting. They are only updated if the default setting (that is loaded on instantiating the plug-in) is used, or if the setting was saved with a project.

This approach helps you to adapt all MIDI controllers to the keyboard, without having to edit and save each setting separately.



Learning a MIDI Message

All parameters that allow you to select a MIDI controller (VibDepth Ctrl, CtrlA, CtrlB, CtrlEnv1, CtrlEnv2, MorphX, Morph Y) offer a Learn command. If chosen, the parameter will automatically be set to use the first appropriate incoming MIDI message. The Learn mode features a 20 second time-out facility: If Sculpture does not receive a MIDI message within 20 seconds, the parameter will revert to its original MIDI controller assignment.

Vib Depth Ctrl Menu

Defines the MIDI controller used for vibrato depth control.

Ctrl A and Ctrl B Menus

Allow the assignment of two controllers that can be used as "via" modulation signals, for direct modulation routings found on the CtrlA and CtrlB tab or for side chain modulations

CtrlEnv 1 and CtrlEnv 2 Menus

Set the controller assignments for the two control envelopes—used as a modulation signal or an offset—in cases where the control envelope is set to Ctrl only or Ctrl+Env modes. It also is used to define the source for recording controller movements.

Morph X and Morph Y Menus

Determine the controller assignments for the X and Y co-ordinates of the Morph Pad. Once assigned, the controller can be used to: manually move the morph point, program single morph envelope points, shift the entire morph envelope and as source for recording morph movements.

Mode Menu

The two Mode menu entries allow you to select whether the MIDI controller assignments shall be taken from a setting or left as they were, when loading a setting. Switching between modes toggles between the original assignments saved with the setting and the default assignments (taken from #default.pst setting—which is loaded on instantiation of Sculpture, if it exists).

Programming: Quick Start Guide

This section of the manual contains a collection of programming guidelines, tips, tricks, and information to assist you in creating particular types of sounds. A more involved look at programming can be found in "Programming: In Depth" on page 527.

Approaches to Programming

Given the flexibility of Sculpture's synthesis core, you can take a number of different approaches to sound design.

If you're the type who prefers to sculpt a sound from scratch—parameter by parameter—you can.

If you prefer to make use of Sculpture's morphing capabilities to create new sounds, you can also do this. This is discussed in "Using Randomization" on page 502.

If you're more of a "tweaker" of factory or user patches, then the parameters that affect the entire instrument may be more your style. These include, the Body EQ and Filter sections, plus the Modulators, for example.

Whatever camp you fall into, you'll be able to achieve new (and hopefully interesting) results.

You should experiment, and familiarize yourself, with each approach. You will find that each has its strengths and weaknesses, and that a combination of methods may strike the balance between satisfying sounds and a social life.

Basics

Throughout the manual, the signal flow of the core synthesis engine has been followed.

When programming from scratch, this is the approach you should also take, working on each component of the sound in isolation.

Obviously, when you're starting out with Sculpture, you won't be familiar with the impact of each parameter on your end results. Don't sweat it; this (and following) section(s) provide pointers when particular types of sounds.

First up, you'll need a plain vanilla or from scratch patch. When you first launch Sculpture, this is exactly what you'll get—a default set of neutral parameters. Sonically, this patch won't set your heart racing, but it will provide you with a starting point for all of the examples in this chapter.

Note: This patch is saved as the "#default" settings file. It is advisable to save a copy of this setting (now!), and name it as desired—maybe "neutral," "plain vanilla" or "from scratch"? This setting can be reloaded as you work through the examples.

The Core Engine

The signal flow was discussed in "The Synthesis Core of Sculpture" on page 456. To recap, and explain in a more hands on way, please follow these steps. This section is intentionally simplified, but please work through it. Knowing the mechanics of Sculpture is essential to your success:

The String

The string is the central synthesis element. It offers a range of parameters that allow you to adjust its material—what it's made of and what environment it's being played in (water, air, and so on).

Each parameter can be explored further in the "String and Object Parameters" on page 462.

Before starting, Control-click on the string (the green horizontal line in the Pickup display) to activate/deactivate string animation. When active, the string will vibrate, making it easier to visualize the impact of the objects and pickups.



- Click the Keyscale button at the bottom of the Material Pad ring, as shown in the graphic.
- Strike and hold or repeatedly strike middle C on your keyboard.

Middle C is the default pitch of the string.

- While doing so, grab hold of the ball on the Material Pad by click-holding on it with your mouse, and move it around. Listen to the sonic changes as you move between the Nylon, Wood, Steel, and Glass materials. Keep an eye on the string (the green horizontal line in the Pickup display, to the left) as you're doing so.
- Release the mouse button once you've found a basic tone that you like.
- Now, experiment with the values of each of the sliders that surround the Material Pad—namely the Media Loss, Tension Mod and Resolution parameters (while continuing to strike middle C). Take note of the changes that each makes to the sound, and also the string animation in the Pickup display. Play a few notes above and below middle C, again keeping your eye on the string.

- You probably noticed that moving the Media Loss, Tension Mod, and Resolution sliders also had an impact on the green and blue Keyscale sliders inside and outside the ring. Grab, and drag each of these keyscale slider arrowheads to different positions—one by one—and play a few notes either side of middle C as you're doing so. Note the changes that happen up or down the keyboard range.
- Once done, click the Release button at the bottom of the Material Pad ring, and adjust the blue Media Loss Release slider while striking notes.

The Objects

Up to three objects of different types are used to excite or disturb the vibration of the string. Each parameter can be explored further in the "String and Object Parameters" on page 462.

- Please reload your default or "plain vanilla" setting.
- Now click the (Object) 1 button (so that it turns gray), while striking a key. You'll note that you hear nothing. The string itself doesn't make a sound unless it is stimulated by the objects. Reactivate the button by clicking on it again.
- Now, click on Object 1's Type menu, and choose each entry in the list. Strike a note repeatedly while doing so to hear the impact of each object type on the string. As always, keep an eye on the string animation.
- Adjust the Strength knob, by click-holding it, and moving your mouse vertically for large changes, or horizontally for fine adjustments. Strike a note repeatedly while doing so.



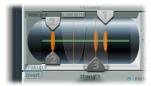
The three string object dials/controls are shown, along with the Pickup section at the center left.

- Drag the Timbre and VeloSens arrowheads to different positions while striking a key
 to audition the changes that they bring. The impact of the Variation parameter is
 indicated on the charts found in "Type" on page 468.
- Try out each of the Gate setting.
- Once you've settled on a particular group of settings for Object 1, activate Object 2 by clicking on the 2 button.

- Change parameters for this object as desired, and note the interaction of the two objects with each other and the string.
- Do the same for Object 3.

The Pickups

The vibration of the string is captured by two movable pickups. The Pickup section also houses three sliders—one for each object.



- Reload the "plain vanilla" setting.
- Click-hold on the Object 1 pickup handle—the down arrowhead with the number "1" on it—and drag it left or right while striking a key. You'll note that adjustments to the object pickup position alters the tonal characteristics of the string.
- Adjust Object 1's Strength control to hear things better, or adjust the tone, if desired. You may also wish to make use of the Object's Timbre and Variation parameters to alter the tone. Use the table in "Type" on page 468.
- Do the same with the Pickup A and Pickup B sliders. You'll note that changes to the pickup positions result in quite different string vibrations (and tonal qualities). Adjust the Level control (directly opposite the Pickup section, on the right of Sculpture's interface) to increase volume, if desired.
- Adjust the pickups of the other two objects. Adjust each object's Strength, Timbre, and Variation parameters to alter the tone. Make use of the tables in "Type" on page 468.

Note that Object 1 can only make use of the excite types found in the first table. Object 2 can make use of any of the types available in either table. Object 3 can only make use of the disturb and damp types found in the second table.

You can disable any of the objects at any time by clicking on their numerical buttons (1, 2, and 3).

About Parameter Interactions

As you're probably discovering, each parameter has an impact on the overall tone of the string—and more often than not—an impact on the string interaction of other parameters.

In effect, each parameter that you introduce or make changes to, will affect the modelled string. This will, in turn, affect the interaction of each parameter with the modelled string. As such, parameter settings that you have already made for, say Object 1, may need to be adjusted when Object 2 is activated.

Generally, such adjustments won't need to be radical, and may only involve a small tweak to the Strength parameters, or perhaps the pickup positions of each object, for example. These parameters have the greatest impact on the tone and level of the objects, and should be the first things you look at if the introduction of Object 2 results in an unwanted change to the color of your sound.

You may want to fine tune the objects further through use of the Timbre and Variation controls.

Small changes—rather than radical ones—will retain the general tonal character of the string (and Object 1), while introducing the new flavor of Object 2.

Back to it

Processing

From the pickups, the signal is sent to the processing section, which consists of the ADSR-equipped amplitude stage, a Waveshaper (with selectable types of waveshaping curves) and a multi-mode filter.

These processors and the pickup parameters are covered in "Processing Parameters" on page 472. Feel free to experiment with these, while referring to the individual parameter descriptions.

All elements covered thus far exist on a per-voice basis.

Additional Processing

All voice signals coming from the *pickups* are summed, and then processed by an integrated Delay effect.

From there, the signal is sent to an EQ-like module (Body EQ), which globally simulates the spectral shape or body response of your instrument. There are several body types to choose from.

The resulting signal is then fed to a Level/Limiter section.

Please explore all of the parameters available in these processing sections at your leisure—using the "plain vanilla" patch each time. This will give you a general feel for each parameter, and its impact on the sounds you hear.

All other parameters on the lower portions of the Sculpture interface (Modulation, Morph, Envelope, and Controller Assignments) are not part of the core synthesis engine, although they can obviously impact upon it. Some tips and uses of these parameters are discussed below.

Creating Basic Sounds

This section covers the creation of basic types of sounds, such as organs, basses, guitars, and so on.

The idea here is to provide you with a starting point for your own experimentation, and to introduce you to different approaches for tone creation with Sculpture.

As you become familiar with the synthesizer, and component modeling, you'll find that there are many ways to achieve an end result. In other words, each component of the sound can be modelled using different techniques and tools available in Sculpture.

This flexible approach allows you to create, say a brass sound, in several ways—using the Waveshaper as a major tonal element in one patch. In another brass patch, the filter and Body EQ can be used to emulate the same sonic component that the Waveshaper provided in the first patch.

A good understanding of the physical properties of the instrument that you are trying to emulate is obviously advantageous. This type of knowledge is pretty specialized, but it can easily be found on the Internet.

You can certainly do some detailed research, but for most sound creation tasks with Sculpture, you can follow this general breakdown formula when creating your string.

How is the sound of the instrument created?

This means:

- Is it a string that is vibrating and resonating in a box (guitar or violin, as examples)?
- Is it a column of air that is vibrating in a tube (flute, trumpet)?
- Is it a solid object that is struck, causing vibration (woodblock)?
- Is it a hollow object that is struck, causing vibration/resonance? (drum, bell)?

What are the physical properties of the instrument? In other words, what is it made of?

When answering this question, don't just consider the body of the instrument. Take into account the string material—nylon or steel on a guitar, or perhaps the thickness and material of the reed in a clarinet or oboe, or a mute in a trumpet.

Is it polyphonic or monophonic?

This is a pretty significant factor, that ties into the next question. Apart from the obvious things such as the inability to play chords on a flute, a modelled string will interact with any currently active string. This, of course, can't happen in a flute. It's strictly a one-note instrument.

How is it played?

Is it bowed, blown, struck, plucked?

Are there other characteristics that contribute to the sonic character of the instrument?

Examples of this are:

- Changes to lip pressure and mouth position with brass and wind instruments
- · Breath or mechanical noises
- Momentary pitch changes—as an example, when fingers are pressed into a fretboard, or when a string is plucked.
- Momentary tonal or level changes—such as when brass players are running out of breath, or fluttering the valves.

Once you've mentally, or physically, written down your list of properties, try to emulate each *component* that contributes to the sound's character. This is what component modeling is all about.

Important: Before beginning, it should be stressed that the following examples are just that—examples. There are many ways to model each component of the sound. Experiment with the settings that are suggested to create your own versions of patches. You'll note that specific parameter values are rarely given, and if they are, feel free to use another.

Just to balance the ledger a little. Subtle changes—particularly when it comes to Keyscaling parameters—result in more controlled sounds. Take your time, and try everything as you're following these examples.

You should make use of other users patches, and the factory settings that ship with the instrument. Close study of these will provide an insight into how the sound was created. Enable/disable different parts of Sculpture to see what each does, and then use this knowledge for your next sonic masterpiece.

Have fun and take risks—you can't break anything!

Basses

Creating bass sounds with Sculpture is pretty straightforward.

- Load up your "plain vanilla" setting.
- Set the Transpose parameter to 1 Oct., and play a few notes around C2. You'll note that the general color of an acoustic bass is already there.
- You can certainly adjust the ball on the Material Pad towards the Nylon entry, but before doing so, change Object 1's type to Pick.
- Have a play on the keyboard, and adjust the ball position while doing so.
- Now take a look at the Strength, Variation, Timbre, and VeloSens parameters of Object 1. Adjust each in turn, to taste.
- You may also wish to adjust the amplitude envelope's Release parameter.
- To make your bass more woody, adjust Object 1's pickup position towards the right. At extreme positions (left or right-hand end), the bottom-end of your bass will be gutted. Try it out!
- Now, adjust the position of Pickup A and Pickup B. As you'll hear, you can quickly recreate a picked acoustic or electric bass sound.
- To instantly make it a hybrid (or full-on) synthesizer bass, click the WaveShaper button, and select from the different types.
- Save Setting as... with new names as you go. You'll probably come up with several new sounds in just a few minutes. Once again, these can be used as templates for future bass sounds.

Bells

At a basic level, bell-like sounds are quite easy to produce with Sculpture. The creation of truly interesting bells involves a little more effort, but the harmonic richness and detuning during the decay/release phase makes all the difference ...

- Load your "plain vanilla" patch.
- Set Object 1's type to Strike.
- Move the Material Pad ball to the very bottom of the pad, and place it pretty much halfway between the Steel and Glass entries. If you play a few notes, you'll hear that the sound is already more bell like.
- Now grab the Media Loss slider, and drag it nearly all the way down. Again, play a few notes, and you'll hear that the release phase of the sound is considerably longer.
- Drag the Resolution slider all the way to the right.
- Drag Pickup A's position to around halfway (0.48).
- Drag Object 1's pickup position to a value of 0.10. You should be starting to get pretty bells now ... play a few notes.
- Now click on the Delay button to activate the Delay unit.
- Click the (Delay) "sync" button, and drag the Delay Time slider to a value of 20 ms.
- Increase the (Delay) Wet Level to 66%.

- Activate the Body EQ by clicking the Body EQ button. Ensure that the Lo Mid Hi
 model is selected.
- Adjust the Low level to 0.55, the Mid to 0.32, and the Hi to 0.20.
- At this point, you will have a working bell sound—but—you'll probably find that there is a tuning issue below C3, in particular. This programming approach was taken because the harmonics of the sound are most noticeable after all other parameters have been set. The solution to the tuning issue primarily lies in the Inner Loss and Stiffness KeyScaling parameters. To adjust, first select the Keyscale button, then clickhold on the green horizontal line (within the Material Pad) for low notes, or the blue horizontal line for high notes—and drag up/down to the desired position.
- Save setting as... with a new name, and use it as the basis for new bell sounds, or your next Christmas album.

Brass

Brass instruments are notoriously difficult to recreate with electronic instruments. Samplers do a reasonable job in the right hands, and with the right sample library, but they lack the organic warmth of a real brass player. This is a simple and generic brass patch that can be played as a solo instrument or as a brass section.

- Load your "plain vanilla" patch.
- Set Object 1's type to Blow.
- Activate Object 2, and set its type to Noise.
- Adjust the Strength of Object 1 to around 0.90.
- Set Object 1 VeloSens to around 0.30.
- Drag the Material Pad ball to a position that is diagonally between the "I" of Inner Loss, and the "I" of the word Steel, while playing middle "C." The sound should be quite brassy.
- Now play the "E" above middle "C," and you'll hear a weird mandolin meets a telephone" kind of sound.
- Drag the Resolution slider to the left/right while playing middle "C" and a few notes down an octave or so. You'll discover a range of sounds that cover everything from sitars to flutes is possible, just through manipulation of this parameter.
- Now click on the Keyscale button and—while playing up/down the keyboard—independently adjust the Resolution slider, plus the Resolution Low/High Keyscale sliders until the range of the keyboard that you wish to play (say an octave or so around middle "C") doesn't suffer from those mandolin/phone artifacts. Oh, and make sure that your sound is still brassy.
- Move Pickup A's position to around 77%.
- Turn on the Waveshaper and select Scream as your preferred type. Adjust the Input Scale and Variation parameters to taste.
- Turn on the Filter. Select HiPass mode, and adjust the Cutoff, Resonance, and other filter parameters to taste. (As a suggestion, Cutoff at 0.30 and Resonance at 0.41).

• Save setting as... with a new name.

Please explore this patch much further. There are a great number of directions that it could be taken in—as a muted trumpet, french horns and even sitars or flutes. The Waveshaper has a significant impact on this sound, and this is the first place you should look to radically alter it.

Use the Delay to create space and the Body EQ to cut the lows and boost the Mids and Hi's.

Adjust the Material Pad ball position towards the Nylon entry, choose Blow as Object 2's type, and then experiment with the Object 1 and 2 positions. This can also result in different brass sounds.

Flute

This can be used as the basis for most instruments in the wind family, including flutes, clarinets, shakuhachi, pan pipes, and so on.

- Load your "plain vanilla" patch.
- Keyboard Mode should, theoretically at least, be set to mono, as flutes and other
 wind instruments are monophonic. After you've set up the patch, experiment with
 this parameter while playing, and make your choice.
- Set Object 1's type to Blow.
- Set Object 2's type to Noise.
- Set the Gate of both objects to Always.
- Adjust Object 2's Strength to a value of around 0.25.
- Adjust Object 1's Velosens parameter to a value around 0.33.
- Move the Material Pad ball to a position that is pretty much at the end of the Inner Loss entry (below the word Nylon).
- Play the keyboard and you should hear a flute-like sound, but with a long release which obviously isn't ideal. Drag the Amplitude Envelope Release slider down to around 0.99 ms.
- Pickup A should be set to a value of 1.00 (far right).
- Set Object 1's pickup position to around 0.27.
- Set Object 2's pickup position to around 0.57.
- Now activate the Waveshaper by clicking on the Waveshaper button, and select the Tube-like Distortion type.
- Play a few notes, and adjust the Waveshaper Input Scale and Variation parameters to taste. (In. ScI = 0.16/Var 0.55, as examples).
- As you play sustained notes, you'll probably notice a distinct lack of interesting timbral shifts (typical of real flute sounds—due to changes in the player's breath, lip position and so on) as the note is held.

- A number of approaches can be taken to add interest to the sustained sound. These
 include; using the vibrato modulator (assigned to aftertouch, perhaps), or perhaps
 recording or drawing in an envelope, and controlling the Waveshape Inner Scale via
 Velocity and/or String Media Loss. You could even use the Loop Alternate Sustain
 Mode. Feel free to experiment!
- Save setting as... with a new name.

Guitar

Guitar, lute, mandolin, and other plucked type instruments, including harps, can be created from this basic patch.

- Load your "plain vanilla" patch.
- Set the Voices parameter to a value of 6—there's only 6 strings on a guitar. Obviously, pick 7 for a banjo, as many as possible for a harp.
- Set Object 1's type to Impulse, if it's not already there.
- Activate Object 2 and set its type to Pick.
- Now move Pickup A's position to the extreme right.
- Move Object 2's Pickup position to a value of 0.14.
- Activate the Body EQ, and select one of the Guitar models.
- Adjust the various Body EQ parameters. These have a major impact on the overall brightness and tone of your guitar sound. (Suggestions—Guitar 2, Int—0.46, Shift—0.38 and Stretch—0.20).
- Finestructure should be at a value around 0.30 to 0.35—but use your ears to judge this. Remember that an increase in the Finestructure value results in a greater load on the computer CPU.
- Click-hold on the Spread Pickup button, and drag your mouse upwards—to increase the perception of stereo width (a value around the 10 o'clock/2 o'clock mark is nice).
- Activate the Filter, and select Lo Pass mode.
- Adjust the Cutoff and Resonance parameters to taste. (suggestion—both at 0.81).
- Adjust the Tension Mod slider upwards, and play the keyboard to see how the momentary detuning effect caused by this parameter affects the sound. Set it to an appropriate amount.
- Set the Level Limiter to Both.
- Save setting as... with a new name.

You may have noticed that a "detour" from the signal path of the Core synthesis engine was taken in the creation of this setting. The reason for this is the major impact that the Body EQ model has on the sound.

In some cases, like this one, it may be better to work slightly out of sequence, rather than strictly follow the signal flow.

Obviously, this is but one guitar sound. You can make use of the Object Strength, Variation, and Timbre parameters, not to mention repositioning the Material Pad ball to create a completely different tone to your guitar.

For quick and easy mandolins, make use of the Delay (or Vibrato) to emulate the double-strike picking that is associated with the instrument.

Organ

Organ sounds are amongst the easiest and quickest sounds to emulate in Sculpture as they have no release phase. This simplifies programming as there is no real need to set up Keyscaling parameters to create the basic tone. You may, however, wish to do so at a later stage—for modulation routing or specific sound design purposes.

- Tip: Play notes/chords while you are adjusting parameters. This way, you can hear what each parameter is doing to the sound.
 - Load your "plain vanilla" patch. (Object 1's type should be set to Impulse. If it isn't, change it now.)
 - Increase the Voices parameter to a value of 8 (or higher if you wish).
 - Move the Material Pad ball into the top left corner.
 - Activate Object 2 by clicking the 2 button.
 - Change Object 2's type to Bow.
 - Set Object 2's Gate Time to Always.
 - Pull the Release slider of the Amplitude Envelope all the way down.
 - Play a "C" chord, and you'll hear a flute-like sound.
 - Drag Pickup A to the extreme right.
 - Play a "C" chord, and you'll hear a cheesy organ sound. As you can see Pickup A's position has a significant impact on the overall sonic character of the sound.
 - Grab Object 2's pickup, and move it around while holding down the "C" chord. Once you've found a position that meets your "that sounds like an organ" criteria, release the object pickup.
 - Now, very slightly adjust Object 2's Timbre parameter upwards.
 - Adjust Object 2's Variation parameter slightly downwards, and upwards, until you find a tone that you like.
 - You may, at this point, want to move the Object 2 pickup parameter to another position. Hold down a chord while doing so.
 - Further tweaks can be made to the Variation and Timbre parameters of Object 2.
 - To introduce a little key click, change Object 1's type to Strike, and adjust its Strength and Timbre.
 - To add a little of that detuned organ vibe, set the Warmth parameter between 0.150 to 0.200.

At this point, you should have a basic organ tone. Save setting as... with a new name. You can use this as the basis for your next organ patch.

You'll probably notice some intermodulations that are introduced when you're playing chords. Apart from the pitch differences between notes in the chord, this is a result of the interactions between each *voice* being produced by Sculpture. These slight variations between each voice (or string, if you will), and their harmonic interaction with each other are not dissimilar to the harmonic interactions of a violin section in an orchestra—even when playing identical lines.

Percussion

Percussive sounds, such as drums, tend to share a similar type of envelope. They contain a strike element, where most of the sonic character is exhibited, followed by a short decay phase. The release phase will vary—dependent on the instrument itself (a snare drum as opposed to a woodblock), and the ambient space it is placed in—a cavern, a bathroom and so on.

- Load your "plain vanilla" patch.
- Set Object 1's type to Strike.
- Activate Object 2, and set its type to Disturb 2.
- Set Object 2's Gate to Always.
- Object 1's Strength should be about 0.84.
- Object 2's Strength should be about 0.34.
- Adjust the Media Loss value upwards/downwards while playing to hear its impact.
 Find a suitable setting.
- Similarly, you can alter the Material Pad ball position—although its impact on the overall tone of the sound is heavily reliant on the Media Loss value.
- Activate the Body EQ and/or Filter, and adjust to your heart's desire.
- Save setting as... with a new name.

This sound can be used as the starting point for a vast range of percussive sounds—including drums, blocks, industrial percussion and even rhythmic sequenced synth sounds.

Just adjust the ball position in the Material Pad, and alter the Media Loss slider position.

Solo Strings—Cello Bowed

Solo stringed instruments, such as violins and cellos, that are played with a bow can be created in much the same way. This sound can also be played polyphonically.

- Load your "plain vanilla" patch.
- Set Transpose to -1 Oct.
- Set Object 1's type to Bow.
- Play the lower half of your MIDI keyboard, and you'll hear a viola/cello like sound. This could obviously be improved.

- Set the Object 1 Velosens slider to match your playing style and that of the music, as you're playing the keyboard. Adjust later, if desired.
- Grab the Tension Mod slider, and move it slightly upwards, so that the arrowhead covers the "D." This emulates the momentary detuning effect of the bow stretching the string.
- Move Pickup A to a position around 0.90.
- Move Object 1's pickup position to a value around 0.48.
- Activate the Body EQ, and select the Violin 1 model.
- Set the Body EQ parameters as follows: Int—0.73, Shift—+1.00, Stretch—+1.00.
- Adjust the Fine Structure to taste.
- Click-hold on the Spread Pickup dial, and drag upwards until the 10.30/1.30 positions are reached.
- Set the Level Limiter to both.
- Save setting as... with a new name.

Set up your own modulations for this sound. The most obvious thing that springs to mind is the introduction of vibrato into the sound after a short period.

The creation of higher pitched solo string instruments is much the same as the example above but special attention must be given to *all* Keyscaling parameters. The Body EQ also has a large bearing on the upper octaves, so take care with its parameters.

Tip: Simply change Object 1's type to Pick, and you'll have a round and rubbery synth bass sound in the lower octaves, and a passable harp across the rest of the keyboard.

Synthesizers

One of Sculpture's great strengths is the creation of endlessly evolving pad and atmospheric sounds. It can also easily do fat synth basses (which you hopefully discovered while following "Basses" on page 518), powerful leads and other types of typical synthesizer sounds.

Sculpture has an advantage over traditional synthesizers as its core synthesis engine produces a wider variety of basic tones, and these tones have an organic quality and richness to them.

Basic Pad

- Load your "plain vanilla" patch.
- Set the Voices parameter to 16.
- Set Object 1's type to Bow.
- Set Object 2's type to Bow Wide.
- Grab the Material Pad ball, and position it at the extreme left of the Pad, exactly halfway between the top/bottom—on the Material line.

- Play a "C" chord (middle "C"), and you'll hear a pad sound.
- Move Pickup A to a position around 0.75, and the pad will become a little sweeter.
- Move Object 1's position to a value of 0.84.
- Move Object 2's position to a value of 0.34.
- As a final step, click on the Points icon that features five dots in the Morph Pad section.
- Set the Int slider in the Morph Pad Randomize section to a value of say 25%.
- Click once on the Morph Rnd button.
- Save setting as... with a new name—say "vanilla_pad."

You'll be using this basic pad sound for the following examples. Don't be shy about doctoring the "vanilla pad"—anything goes, so make use of the Filter, the Delay, EQ, and Waveshaper to create new sounds.

Evolver

- Load your "vanilla_pad" patch.
- Click on the LFO 1 tab at the bottom of the interface.
- Click the 1 button, and play the keyboard. Not much difference, there, huh?
- Now, click-hold on the Min/Max sliders, and drag left and right, while holding down a chord. Finally settle on a value of 0.15.
- Click-hold on the Target (Pitch), and select Object 1 Strength. You'll hear a fluttering sound.
- Now click on the "sync" button, and adjust the Rate knob to a value of 1/8t.
- Activate the second LFO 1 object by clicking the 1 button, and select Object 1
 Position as the Target.
- If you play the keyboard, there's not much that's different.
- Set the via option of the second LFO 1 object to Velocity.
- Play the keyboard at different velocities, and you'll hear some shifting of the Object 1 pickup position ... and now, to make it interesting ...
- Change the Waveform to Sample&Hold, and play the keyboard at different velocities. If you've got a sustain pedal, use it. Listen to the endlessly evolving sound.
- You may like to experiment with the project tempo and LFO Rate.
- You may want to alter the Spread Pickup value, and introduce LFO 2 or the other modulators.

Morpher

- Load your "vanilla_pad" patch.
- Click on the R(ecord) button in the Morph Trigger section.
- Play a chord on the keyboard, and drag the Morph Pad ball in a circle.
- Once you're done, press the R(ecord) button.

- Now change the Morph Mode to Env only, and you should see your Morph circle.
- Play the keyboard. There's your morphed pad!
- Feel free to adjust the morph envelope parameters.

Remember when you were asked to use the Morph Points, Intensity, and Rnd parameters while setting up the "vanilla_pad"? This was to ensure that there were several morph points already available for your morphing pleasure.

You can, if you wish, retain the path of your morphed pad, and continue to click the RND button and adjust the Intensity slider for an endless variety of sounds.

Modulations

The modulation options can be very important for the emulation of acoustic instruments. As a simple example, the introduction of vibrato into a trumpet sound over time.

Many classic synthesizer sounds also rely as much on modulation as they do on the basic sound source components—the VCO, VCF, and VCA.

Here's a number of quick modulation tips:

- Imagine that you want to modulate the timbre of Object 2, with the LFO, for example. To do so, click on the LFO 1 or 2 Tab, press the 1 or 2 button, select the desired Source/Target and value. That's it!
- To control any modulation with an external controller, such as your keyboard's modulation wheel, set the desired "via" parameter to CtrlA or CtrlB respectively. By default, the Mod Wheel is set to CtrlA.
- The Bouncing damp type available to Object 3 affects the sound in a very interesting way, but it cannot be run synchronously with the project tempo. To create a similar effect to the Bouncing Object—but in-sync, you could use a Disturb object type, and move it by modulating its vertical position (Timbre) with an LFO.

Breath control is available to all users of Sculpture, even if you don't own a breath controller. To do so, record breath controller modulations into the recordable envelopes (using a mod wheel or other controller), and then reassign the recorded modulation path (use the CtrlEnv 1 and/or 2 parameters) with each NoteOn.

Programming: In Depth

This tutorial explains how you can program sounds with Sculpture from scratch. Based on Sculpture's string model, you'll learn how to use the individual sound shaping parameters in order to recreate the physical properties of an instrument in detail.

Note: You will find these tutorial settings in the Factory > Tutorial Settings sub-folder of the Settings menu (in the header of the Sculpture plug-in window).

Programming Electric Basses With Sculpture

This section concentrates on a single instrument type: the electric bass, including all of its important variations and articulations. The physical nature of electric basses is not as complex—and encumbered with acoustic issues—as is the case for many acoustic instruments. This instrument is therefore an excellent choice for the sound programming tutorial, the goal of which is to acquaint you with the art of using Sculpture to accurately reproduce detailed sounds.

In order to build a bass (including all components) in Sculpture, it is necessary to understand the basic, physical process of sound production within the instrument. Before looking at the practical programming process within Sculpture, you'll find detailed information on the construction of electric basses in the next section.

The Most Important Aspects of Electric Basses

In general, the electric bass has four strings. The lowest string is usually tuned to E 0 or E (MIDI note number 28). The strings above the low E are tuned in fourths, thus A, D, and G. You can, of course, find basses that have five, six, and even more strings. As Sculpture has no tonal limits, this is of little concern.

What is much more important for sound programming is the overtone content of the bass sound. This depends primarily on the qualities of the strings.

- Round wound strings: A very fine wire is wound around a steel cable core which results in a wiry, metallic sound that's full of overtones.
- Flat wound strings: The fine wire wrapping is ground down or polished smooth, and the sound has far fewer overtones in comparison. (These are much less popular today.)

In contrast to guitar strings, the structure and workmanship are the same for all strings in a set. Sets combining wound and non-wound strings do not exist.

The relationship between string length and string tension has a significant impact on the overtone content. Disregarding basses that can be adjusted to different scale lengths (different vibrating string lengths), the actual playing position that is used plays an important role. When you play D at the tenth fret on the low E string, it sounds more muffled than the same pitch played on the open D string.

The number of frets differs from bass to bass and depends on the scale length. Don't worry about pitches higher than a single ledger line C; the actual functional range of this instrument is primarily in its two lower octaves—between E 0 and E 2.

Also worth mentioning is the fretless electric bass. Like all instruments of this type, it is freely tunable and possesses a distinctive, individual sound. Over the course of this tutorial you will discover how to program this type of instrument sound in Sculpture.

There are three types of articulations that will be discussed:

- Fingered: The strings are played with the alternating index and middle fingers.
- *Picked*: The strings are played with a pick.
- *Thumbed/Slapped:* The strings are either played with the (side of the) thumb on the fingerboard or plucked strongly with the fingers.

The vibration of the strings is captured by an electromagnetic pickup. When the string is vibrating, its steel core affects the magnetic field. The pickups are almost always found some distance to the side, nearer to the bridge and stop tailpiece. There are different pickup concepts for electric basses and often two or more pickups are combined to make the sound. To avoid getting into too much detail at this point, there is a rule of thumb that applies:

The further you move the pickup towards the middle of the string, the bassier the sound will be, and the more hollow it will sound. The further you move the pickup towards the end of the string, the more the sound's overtone content will increase, becoming more dense and compact. The sound will have more mid-range frequencies or buzz, and less bass. If the pickup is positioned at the very end of the string, the sound becomes very thin. This behavior mirrors the actual playing position of a real string: If you play more towards the middle of the string, you'll get a smooth, even, and powerful sound which contains limited harmonic denseness (overtones). If the string is played at the bridge, the sound develops a nasal twang and features more buzz and more overtones.

Now to the body of the instrument, and its resonant properties. Almost all electric basses have a steel rod running through the neck, to strengthen it, and a body made of solid wood. This construction allows the strings to vibrate relatively freely (sustain), even though very little direct sound is generated. The pickups and the amplifier and speaker systems are responsible for the actual sound of the instrument.

The acoustic interaction between body, strings, and external sound sources is much less complex than with pure acoustic instruments.

The vibration of the strings is, of course, naturally hampered by several physical factors: the radius of motion of the string (antenode) is impeded by the left bridge or by the first fret that's pressed down upon (and the frets in between). This can lead to the development of overtones which can take the form of anything from a slight humming or buzzing to a strong scraping or scratching sound.

In addition, factors such as the material properties of the strings and the instrument, as well as the softness of your fingertips also serve to dampen the vibration of the string

Programming a Basic Bass

This section covers programming of a basic bass sound. This will serve as the foundation for the different bass sounds which you will be creating thereafter.

Heed the following steps to create the proper working environment for design of your own homemade bass:

1 Make sure the range from C 0 to C 3 is available on your keyboard by either; transposing your master keyboard, or using the Transpose function in the Region Parameter box of Logic Pro.

Note: You can, of course, transpose sounds within Sculpture but this isn't the best solution in this case. The reason is that sounds would not be compatible with MIDI regions in which note number 60 as middle C is considered to be the measure of all things.

2 Choose the default setting within Sculpture.

In order to recreate the sound characteristics of a typical bass instrument:

- 1 Set the Attack value of the amplitude envelope to its minimum value (0.00 ms). You'll find the Attack slider just to the right of the Material Pad.
- 2 Shorten the Release time of the amplitude envelope to a value between 4 and 5 ms. Play a key on your keyboard. The note should stop abruptly when you release the key and should be free of artifacts (a digital crackle or snap). If you encounter any artifacts during the course of this tutorial, please carefully increase the Release time.
- 3 Play some sustained notes in the range above E 0. These will die away (too) quickly. Correct this quick die-out with the Media Loss parameter by pulling the slider located to the left of the Material Pad almost all the way down to the bottom. For your information: the low E string on a high quality bass can sound for over a minute! Our basic bass should simulate a fingered articulation, which means the sound is created by striking the strings with the fingers.
- 4 Choose the Pick entry from the Type menu of Object 1.

Don't be confused by the name of the object type: despite its name, this model is appropriate for simulating the playing of strings with your fingers.

Play some notes in the lower range. you'll note that the sound is very muffled, hollow, and distorted. Before you adjust further parameters in Object 1, you need to set the position of the pickup.

This is accomplished in Sculpture's pickup display located to the left of the Material Pad. You'll find three trapezoidally shaped sliders, representing Objects 1 to 3. Both of the transparent bell-shaped curves help you to visualize the position and width of Pickup A and Pickup B.

On electric basses, the pickups are found quite a way off to the side and near the bridge. This particular bass only has a single pickup.

The behavior of a single pickup is simulated by placing both pickups at exactly the same position.

5 Keep an eye on the help tag, and drag Pickup B to the exact position of Pickup A. The two thin orange lines should overlap perfectly. As a suitable value for the example, set both pickups to 0.10.



Note: Make sure that the Invert switch to the lower left of the pickup display isn't turned on, as this would cause the pickups to completely cancel each other out.

It's now time to determine the playing position:

- 6 Grab the Object 1 slider in the pickup display and move it in a horizontal direction. Play the keyboard while doing so, to hear the changes it makes.
- 7 You'll quickly realize that you can only achieve a precise, crisp sound when relatively far away from the middle of the string. Move Object 1 closer to the pickup (position 0.15—see GUI detail below).



8 The low notes are still distorted. You can remedy this by adjusting the Level dial found to the right of the amplitude envelope. Set a value of –10 dB.

Although you can already recognize the sound of an electric bass, it doesn't sound wiry enough yet. Now it's time to focus on the bass strings themselves.

In order to recreate the material properties of a set of round wound strings:

- 1 Move the ball in the Material Pad up and down at the left edge. Pay attention to how the overtones react. Move the ball to the lower left hand corner. The sound should vaguely remind you of the sound of a low piano string. As the overtones sustain too long, the tone sounds somewhat unnatural.
- 2 Move the ball upwards until you hear an acceptable sound. Try the position shown in the example:



Note: In general, a splaying of the overtones in low wound strings is typical. You can recognize it by the slightly impure, metallic sound. This occurs because the partials (overtones) are not exact whole number multiples of the fundamental frequency, but rather they are shifted somewhat higher. An example of this effect in the real world of electro-acoustic instruments are the low strings on a Yamaha CP70. This is overkill, however, but your bass model will benefit from a small amount of this effect.

How to splay overtones in Sculpture:

- 1 Move the ball in the Material Pad gradually to the right. The sound takes on a more pure, bell-like character.
- 2 To realistically simulate the splaying of overtones, try the following example setting:



The vibration of a bass string does not occur in a vacuum. The antenode of the string frequently encounters the natural, physical limitations of the instrument. This is heard as the typical buzzing and rattling that occurs when the strings touch the frets.

To simulate these disturbing elements with Object 2:

- 1 Activate Object 2, and choose the Bouncing type menu item. The sound should now vaguely remind you of a mandolin tremolo. This is way too strong an effect for this kind of sound.
- 2 Move Object 2 all the way to the right (a value of 1.00).
- 3 Experiment with Object 2's parameters. A discrete and realistic result can be achieved with the following parameter values: Strength: 0.33, Timbre: –1.00 and Variation: –0.69.

Play some low notes, and you'll find that, once again, the overtones sustain a little too long (somewhat like the lowest notes/strings on a piano).

Use Object 3 to dampen these overtones:

- 1 Activate Object 3 and choose the Damp type menu option.
- 2 Also move Object 3 all the way to the right (value 1.00).
- 3 Set the Strength parameter to 0.18.

Note: Experiment with how the Strength parameter of Object 3 interacts with the Inner Loss Material Pad parameter. The higher the Inner Loss value, the smaller the Strength value can be, and vice versa.

In order to more realistically replicate the different tonal ranges of the bass, use Sculpture's scaling function.

First up, activate the scaling function display:

 Click on the Keyscale button located at the bottom of the Material Pad. The key-scaling below C3 is displayed in green, the range above in light blue. The Material Pad with its key-scaling parameters activated:



Note: The most relevant performance range for basses is found exclusively below C3. For this reason, you should make use of the green sliders to set the actual timbre of the sound. The primary sliders found around the ring determine the timbre of the sound above C3. For the moment, ignore the blue sliders (which control high key-scaling) and simply set them to the same positions as the main sliders.

Once activated, the key-scaling function is used to adjust the timbre of the sound, independent of pitch. To do this, you should initially employ the Resolution parameter. This is normally used to set the balance between DSP load and sound quality. As the overtone spectrum is reduced at low Resolution values, this parameter can also be used to shape the sound.

To use the Resolution parameter to control the timbre, independent of pitch:

- 1 Play some notes at the higher end of the bass's range (around C2), then move the Resolution slider all the way to the right and then gradually back towards the left.
- 2 You can hear how the sound loses overtones, yet simultaneously becomes louder. At low Resolution values, an inharmonic metallic rattling is heard in the sound.
- 3 Increase the Resolution value until the metallic rattling disappears. Set the slider to the following position:



4 Play some notes in the bottom range (around E 0). You'll note that the sound is quite muffled and vintage-like. Move the green Low Keyscale slider (found below the main Resolution slider) all the way to the right; the low range should now sound a little more wiry.

With most stringed instruments, the overtone content decreases as the pitch becomes higher. Strictly speaking, this is only true of open strings, and even then, in a limited sense. If the strings are fingered, the length of the string is shortened (especially in the high register) and the effect becomes more significant.

Use the Inner Loss parameter to scale the overtone content, dependent on pitch:

- 1 Move the Material Pad ball above the words Inner Loss. Try to move the ball solely in a vertical direction, in order to maintain a constant Stiffness value.
- 2 Grab the green line next to the ball, and pull it towards the bottom until the small green diamond is located directly above the word Steel.

When playing, you'll recognize the smooth transition that takes place between the wiry, overtone-rich sound at the bottom end and the extremely dampened sound in the upper register. This (exaggerated) setting was chosen to clearly demonstrate the scaling principle in stringed instruments. In order to achieve an authentic sound and timbre, try the following setting:



In basses in particular, low notes sustain far longer than the high notes. Sculpture allows you to authentically and convincingly simulate this behavior with the Media Loss parameter.

To use the Media Loss parameter to scale the fading phase of the note, dependent on pitch:

- 1 Play a few held notes in the range around C2 and above. You'll hear that these notes die out much too slowly. Move the Media Loss slider up until this range begins to fade out quickly enough. The downside is that the lower notes now die out too quickly!
- 2 Pull the green Media Loss Key Scale slider down until the fade-out phase of the lower range is sufficiently long enough.
- 3 Compare your results with these recommended values:



You've completed this section, and created a basic bass that's articulated with your fingers. Save this as E-Bass Fingered Basic. In the following sections, you'll be using this basic bass as a foundation for the construction of further bass sounds.

Modifying the Frequency Spectrum of Your Basic Bass

The scope for sound design, by altering the frequency spectrum of electromagnetic instruments, is far more flexible than that offered by acoustic instruments. In addition to the number of pickups, the choice of amplifier, the equalization setting within the amplifier and, last but not least, the physical properties of the speakers and their enclosing cabinet also play a major role.

The central features of your electric bass sound are complete, but the sound can be improved by paying close attention to some details. Here are a few general suggestions:

- Vary the position of the pickups. Try placing each of them in different positions. This will cancel out certain frequencies, and others will be summed together.
- Try turning on the Invert switch, even though this effect is not typical for electric basses.
- What is typical for bass sounds is the placement of the pickups in the outer left third of the string model. The farther you move them to the left, the thinner and more nasal the sound will become.
- Shifting Object 1 will have a similar effect. Try different combinations here as well.

The Body EQ is ideal for giving the bass sound that final, finishing touch. Your electric bass sound could be a little less smooth, and be a bit more precise in its attack phase. Bassists like to use the terms drier and more bite to describe this phenomenon.

To alter the frequency spectrum of your basic bass with the Body EQ:

- 1 Load the E-Bass Fingered Basic setting.
- 2 Select the standard Lo Mid Hi Body EQ model.
- 3 Reduce the low bass frequencies by setting the Low dial to a value of -0.30.
- 4 Boost the mid-range frequencies substantially by setting the Mid dial to a value of 0.50. Set the Mid frequency slider to a value of 0.26.
- 5 You'll probably find that the boosting of the low mid frequencies is a little too strong at this point, so please return the Mid value to 0.30.



6 The sound could stand to be a little more wiry, so set the High dial to a value of 0.30.

- 7 To finish off, set the Level dial (to the right of the amplitude envelope) to a value of 3 dB. The sound is now as loud as possible, without the low notes distorting.
- 8 Save this sound setting, as you'll need it for further modifications later. Please name it E-Bass Fingered Basic EQ1.

Pick Bass

The basic bass is played with the fingers. In the following example you will simulate playing the strings with a pick, using the Pick object type. The Timbre parameter will be used to adjust the relationship between the speed and intensity at which the string is struck. The Variation parameter will be used to define the virtual material density (or hardness) of the pick.

If you imagine the fingers to be very soft picks, it makes sense to alter the Pick parameters so that a hard plastic pick is the outcome.

To simulate playing with a pick:

- 1 Please load the E-Bass Fingered Basic setting.
- 2 Set the Timbre parameter of Object 1 to its maximum value of 1.00. You'll note that the attack is now stronger.
- 3 Try several different Variation settings to get a feel for the material qualities of the pick.

Note: Not all positions will deliver usable results for the entire range of the instrument.

4 You'll get a consistent, working setting for the two octaves above E 0 with the following parameter settings: Position 0.17 (Pickup display), Strength 1.00 (maximum), Timbre 0.90, Variation 0.56.

When these settings are used, you'll find that the sound has become softer and very thin. In fact, it's somewhat reminiscent of a clavinet.

To compensate for this side effect with the Body EQ:

- 1 Activate the Body EQ and add a healthy portion of bottom end to the sound by setting the Low parameter to 0.60. Mid should be set to 0.33.
- 2 Set the High dial to –0.45 as the sound is now so bright that rolling off a few of the highs can't hurt.
- 3 Now bring the volume into line. If you adjust the Level dial to 2.5 dB, nothing should be distorting. If this isn't the case, try reducing some more of the bottom end with the Low dial.
- 4 Save this setting as Pick Open Roundwound.

Damping

Playing with a pick is often combined with a damping technique that employs the ball of the thumb. The right hand, which also holds the pick, should physically lay on top of the strings at the bridge. This technique results in the sound having less overtone content but become more percussive and punchy at the same time. You can variably control the timbre of the sound through the angle and pressure of your hand while playing.

Object 3 will be used to emulate the virtual ball of the thumb in this example. The *Timbre* parameter determines the kind of damping that occurs. *Variation* dictates the length of the string section that is being dampened.

To achieve this effect, proceed as follows:

- 1 Set the Object 3 type to Damp.
- 2 Set Object 3's Strength parameter to 0.50.
- 3 Move Object 3 a little bit to the right in the Pickup display (to position 0.95) to simulate the width and position of the ball of the thumb lying on the bridge.
- 4 Set Timbre to its minimum value (-1.00) to achieve a very soft damping effect.
- 5 Set the Variation parameter to its maximum value of 1.00.
 - You'll note a metallic ringing that occurs during the attack phase still can be heard in the octave above E0.
- 6 To suppress it the ringing, Move the small green diamond on the Material Pad to a position directly under the ball. In doing so, you've just increased the Inner Loss value for the low key range.
 - *Note:* In order to place the diamond exactly under the ball, you can also click it while pressing the Option key.
- 7 Save this setting as Pick Bass Half muted.

Harmonics

Harmonics are single partials (overtones) of the overall sound. They can be heard by damping certain points along the string. This is done by lightly laying the fingers of the left hand (assuming a right-handed bass player) on the string (not pressing down) before the note is articulated. The first overtone, the octave, is achieved by placing your finger at the exact middle of the string, in effect separating the string into two halves. The next overtone is the fifth above the octave and the position of your finger should divide the string into a ratio of one-third to two-thirds. The next highest overtone separates the string into proportions of one-quarter to three-quarters and so on.

To simulate fingers lightly touching the strings:

- 1 Object 3 is used as a damper. Select the Damp type.
- 2 Adjust Object 3's Timbre parameter to its maximum value of 1.00.
- 3 Variation must be set to its initial value of 0.00. Simply click on the slider while holding down the Option key to do so.
- 4 Move Object 3 to the exact middle (0.50) of the Pickup display. Play the keyboard, and you'll hear the first overtone as a harmonic.
- 5 While playing, very slowly move Object 3 towards the left of the Pickup display. In doing this, you are effectively scrolling through the overtone series, so to speak.
- 6 Save this setting as Flageolet Xmple.

Vintage Flat Wound Pick Bass

Now, in just a few easy steps, you can transform the pick bass into a vintage pick bass with flat wound strings. This bass sound is typical for the funk and soul music of the 70s, but you'll also find it in many easy listening arrangements.

Proceed as follows:

- 1 First load the Pick Bass Half Muted setting.
- 2 Drag the Material Pad ball upwards and the sound becomes more muffled.
- 3 Increase the Object 3 Strength parameter to 0.70. The result is a muted pick bass with flat wound strings.
- Tip: If you turn Object 3 off, you'll hear a sound that is reminiscent of a 1970s' Fender Precision Bass.
- 4 Save this setting as Flatwound Pick Damped.

To get a nice percussive sound à la Bert Kaempfert, proceed as follows:

- 1 Turn Object 3 back on.
- 2 Move both pickups a little to the left (position 0.08).
- 3 Our virtual pick (Object 1) can also be moved a little further to the outside (position 0.10).
- 4 You can add the icing to the cake with the Body EQ. Turn the Low dial to its maximum value (1.00).



- 5 To remove the smacking in the attack phase, use the graphical display to choose a value of 0.48 for the Body EQ Mid frequency, then use the dial to increase this value to 0.51. Option-click on the Body EQ High parameter to set it to a value of 0.00.
- 6 Save this setting as Easy Listening Pick Bass.

Slap Bass

You're actually dealing with two different articulations here. The low notes originate when the thumb literally slaps the strings on the upper part of the fingerboard. The high notes are produced when the strings are strongly plucked or popped with the fingers. This is achieved by hooking a finger under the string, pulling it away from the instrument and then allowing it to slap back onto the fingerboard. In conjunction, these articulation methods make up the typically aggressive and overtone-rich slap bass sound.

To simulate a slap bass:

- 1 Load the E-Bass Fingered Basic EQ1 setting.
- 2 Turn off the Body EQ.
- 3 Also turn off Object 2 and Object 3, for now.

As the basic sound of a slap bass is brighter than a standard fingered bass, you need to adjust some Material Pad settings:

- 4 Return the Low Key Scale parameter to its initial value by clicking on the little green triangle, while holding down the Option key.
- 5 Move the ball down a little, and the sound becomes more wiry. The ball should now be located directly above the word Steel on the horizontal axis.



From the models at your disposal, Strike is the most suitable for simulating a thumb physically striking the strings from above. This model is not, however, as appropriate for the slapped (popped) strings. It makes the most sense to choose the Pick model for this purpose.

- 6 To be safe, turn the Level dial to -25 dB.
- 7 Select the Pick model for Object 1.

8 Move Object 1 to position 0.90 in the Pickup display. This position corresponds to a playing position above or on the fingerboard.

Note: Given its universal concept, Sculpture will not react exactly like a bass, where one would tend to play in the middle of the string on the upper part of the fingerboard. Try moving Object 1 to this position and see how it sounds. You'll find that the sound is little too smooth.

Setting the parameters for Object 1:

- 1 Set Timbre to a value of 0.38; this corresponds to a rapid attack.
- 2 Set the Strength parameter to 0.53.
- 3 Set the Variation parameter to –0.69; this defines the softer material that constitutes the fleshy part on the side of your slapping thumb.

You're probably familiar with the sound of low notes when played with your thumb. What's missing, thus far, is the typical bright rattling that is created when the string strikes the fingerboard. Use Object 2 to this end, and select the Bound type menu option. Bound limits the antenode of the string in exactly the same way as the fingerboard on a real electric bass.

To review the functions of these parameters: Timbre determines the angle of the obstacle to the string while Variation defines the type and degree of reflection.

Adjust Object 2's parameters to the following settings:

- 1 Set Timbre to 0.39. This corresponds to a fingerboard that runs almost parallel to the string.
- 2 Set the Strength parameter to 0.33.

Note: Try some higher values as well. You'll see that the sound becomes softer and softer until it's completely dampened by the obstacle.

3 An appropriate value for Variation is 0.64. Despite the overtone-rich reflection, the string can still vibrate freely.

Note: Try some negative values: you'll see that the reflections can no longer develop in an unhindered fashion.

- 4 Set the Level dial to -3 dB; the Bound obstacle has made the sound softer.
- 5 The sound is still too smooth for a real slap bass; try using the Body EQ again. Switch it on, and adjust the parameters to the following settings: Low = 0.25, Mid Frequency = 0.59, Mid = 0.43, High = 0.51.
- 6 Save this sound as Slap Bass Basic#1.

Fretless Bass

With the exception of shared playing techniques, the fretless bass differs from a normal bass through its buzzing, singing sound. As the frets on the fingerboard of a standard bass function as a collection of mini-bridges and allow the string to vibrate in an unobstructed fashion, the direct collision of the string's antenode with the fingerboard on a fretless bass is responsible for its typical sound. The string length on a fretless bass is markedly shorter than that of an acoustic double bass. The upshot of this is that a controlled buzzing is produced, even when played with a weak attack. This buzzing can be consistently reproduced in the high register, even on fretless basses that have very short string lengths. The use of the comparatively soft tip of your finger, instead of a hard, metallic fret, to divide or shorten the string also plays a role.

This is how you program a fretless bass:

- 1 Load the E-Bass Fingered Basic EQ1 setting.
- 2 Turn Object 3 off. You'll come back to it later.
- 3 Choose the Object 2 Disturb type menu option.
- Tip: The Disturb model functions in the following way: the Timbre parameter determines how far the string is deflected from its resting position by the obstacle. Positive values precipitate no deflection of the vibration from its resting position. Variation defines the length of the string section that is disturbed: positive values correspond to a longer section of string, negative values to a correspondingly shorter string section.
- 4 Adjust Object 2's parameters to the following values: Strength 0.14, Timbre –0.05, Variation –1.00.
- 5 Object 2's pickup position remains at the far right; please enter a value of 0.99. You'll note that the range between C2 and C3 already sounds quite acceptable, but the buzzing in the lower notes is still too strong. It is somewhat sitar-like, so keep this disturb model in mind when it comes to creating a home-spun sitar.
- 6 Try different settings for the Strength parameter for both the higher and lower playing ranges. You'll see that, at best, only a compromise is possible. The buzzing is either too loud in the low range or not present enough in the high range.
 - Obviously, the effect needs to be scaled over the relevant tonal range. Unlike the parameters for the string, Objects 1 to 3 don't have a directly adressable key scaling function. There's a clever way around this: Both LFOs offer a key scaling function. As you probably don't want the buzzing to be modulated by a periodic oscillation, you need to reduce the LFO speed to infinitely slow or 0. In this way, you can deactivate the LFO itself, but still use its modulation matrix.
- 7 Activate LFO2 by clicking on the LFO2 button at the bottom left, and set the Rate dial to a value of 0.00 Hz.

8 Click on the button marked 1 (next to the Rate slider, to the upper right) to activate the first modulation target.



- 9 Choose Object2 Strength as the Target parameter.
- 10 Select the KeyScale entry in the via column.
- 11 Move the lower slider labeled Amt (amount) to the right while you are playing. You'll quickly realize that the singing buzzing fades out in the lower range, while gradually being retained as you move towards C3. Set the slider to a value of 0.15. The buzzing will now be far more moderate in the low range.
- 12 Switch Object 3 back on. Set Timbre to its minimum value (–1.00) and Variation to its maximum value (1.00). Object 3 should be positioned all the way to the right, at a value of 1.00.
- 13 Vary the Object 3 Strength parameter. You'll discover that the overtone content of the buzzing can be controlled very effectively. A Strength value of 0.25 is recommended here.
- 14 Save this setting as Fretless Roundwound#1.

Modulation and Detuning

Detuning and ensemble effects are normally achieved using a modulation effect or by combining doubling and detuning. When using a fretless bass for a solo part, a broad chorus effect adds a nice touch.

As Sculpture can only synthesize one note at a time at any certain pitch, simple doubling isn't an option. There are, however, alternatives for bringing movement and life into the sound. Almost all of the *type* parameters of the different objects can be modulated by LFOs, resulting in a vast number of possible combinations.

Emulating a chorus effect by modulating the pickup positions:

- 1 Make sure that you've got the Fretless Roundwound#1 setting loaded.
- 2 Adjust the position of Pickup B to 0.20.
- 3 The Spread control element is located to the left of the Level dial. Click-hold on the (Spread) Pickup semicircle, and move the mouse in an upwards direction. Both of the light blue dots move downwards towards the letters L and R.

You can hear how the stereo breadth of the fretless sound has increased. Pickup A is sent out on the right channel, while Pickup B occupies the left channel.



Note: Although only modern basses offer such stereophonic features, it's still fun to process conventional sounds (such as those created in the previous examples) with this effect. Note that not all pickup positions are monophonic compatible; you can check this by returning the Spread Pickup setting to mono (by clicking on the Pickup semicircle while holding the Option key).

To make the pickups move:

- 1 Select LFO1.
- 2 Activate the first modulation target by clicking on the "1" button (next to the Rate slider, to the upper right).
- 3 Choose Pickup Position A-B as the modulation target.
- 4 Set the Rate dial to 1.00 Hz.
- In order to hear the effect, the modulation intensity (amount) has to be set. Familiarize yourself with this effect by moving the slider labeled *Amt* (amount) gradually to the right. Set it to a final value of 0.15, a moderate rate that doesn't wobble too much.
- 6 Save this setting as Fretless Chorus Dry.
- Tip: At the maximum stereo breadth, effects based on detuning are not as prominent, especially when the beats heard in the sound result from signal differences between the left and right channels. This is only valid to a certain degree because the motion of the pickup doesn't create a true chorus or harmonizer effect. Try it out and see what happens when the stereo breadth is reduced a little. Also test other modulation targets: Pickup Pos A+B, Pickup Pan A+B, Pickup Pan A-B, and String Stiffness are recommended. Have fun!

Reverb and Reflections

As a rule, basses are mixed without effects (dry) and you probably haven't missed any reverb or delay effects in the examples, so far. Having said this, a little bit of reverb can be quite appealing on a fretless bass, when it's used as a solo instrument. Use Sculpture's Delay section to emulate this.

In order to create an unobtrusive atmospheric space, proceed as follows:

- 1 Load the Fretless Chorus Dry setting.
- 2 Turn on the Delay section by clicking on the Delay button.
- 3 Set the Input Balance slider to 1.00.
- 4 Switch off the tempo synchronization of the delay by deactivating the small Sync button (found directly to the right of the Delay Time slider).



- 5 Set the Delay Time slider to 90 ms.
- 6 Set the Xfeed knob to 0.30.

The individual reflections are still too brash. In order to make the effect more discrete and unobtrusive, adjust the frequency spectrum and amplitude of the reflections. Start with the frequency spectrum:

7 Set LoCut to 200 Hz and HiCut to 1000 Hz (in the Delay section).

The LoCut at 200 Hz excludes the low frequencies in the reflections, thus avoiding a muddy sound. The comparatively drastic cut to the highs with the HiCut parameter blurs the individual reflections, thereby creating the impression of a small room with soft surfaces.

- 8 Reduce the total level of the effect by setting the Wet Level dial to a value of 25%.
- 9 Save this setting as Fretless Chorus+Ambience.

This example shows that the Delay section can be used as a substitute reverb for small spaces. For sophisticated reverb effects, it's best to process Sculpture's output with one of Logic Studio's reverb plug-ins.

Creating a drowned in delay effect:

- 1 Reload the Fretless Chorus Dry setting.
- 2 Switch the Delay section on.
- 3 Move the Input Balance slider all the way to the right (1.00).
- 4 Set the Delay Time value to 1/4t (quarter note triplet).
- 5 Set the Feedback dial to a value of 0.20.
- 6 Adjust the Xfeed knob to a value of 0.30.
- 7 Set the LoCut to 200 Hz and the HiCut to 1600 Hz.
- 8 Now adjust the overall level of the effect; try setting the Wet Level dial to a value of 45%.
- 9 Vary the stereo position and rhythmical structure of the delay, by moving the small light blue diamond around the Delay Pad.
- 10 Save this setting as Fretless Chorus+Wet Delay.

Synthesized Sounds

In the preceding sections, you learned how to program natural bass sounds with Sculpture: by authentically reproducing the real physical interaction that occurs between a string and the exciting agent that acts upon it. While producing such lifelike models is undoubtedly a forte of Sculpture's architecture, its sonic capabilities extend to the creation of very different sounds as well.

Sculpture contains a number of functions that you can use to create new and novel synthesized sounds. This includes the Morph Pad, which can be automated, as well as recordable and programmable envelopes that can be used in a rhythmic context.

Such features are usually unnecessary when reproducing natural bass sounds, as no electric bass that exists can alter the tonal characteristics of the string during the decay phase of a note—perhaps from wood to metal—and rhythmically synchronize this change to the tempo of the project. These functions are very useful, however, when creating sustained, atmospheric sounds where slow and interesting modulations help it to come alive.

In the following sections, Sculpture's ability to create somewhat spacy and less organic sounds, using several pad patches as examples, will be demonstrated. After having studied the modeling of bass sounds in the preceding sections, you'll now be introduced you to a totally different group of sounds. Provided that you're willing to supply the necessary level of curiosity and time investment for experimentation, you'll discover a wide variety of interesting and animated sounds.

Within the framework of these short experiments, it's of course impossible to comprehensively cover all of Sculpture's possibilities. Please experiment with the suggested settings and closely observe the results of the changes you make. In this way you can learn a lot about the instrument, and hopefully be inspired to create new sounds and variations.

Note: You will find the settings for these tutorials in the Tutorial Settings folder in the settings menu (in the head of the Sculpture plug-in window).

The Sustained Sound

First, please load Sculpture's default setting again; the very simple sound consisting of a plucked string that vibrates and fades away. Obviously, this sound needs some drastic changes as you're going for a sustained or extended sound for pads, rather than one that dies away.

Have a look at the three objects: you can see that only Object 1 is active, and acts on the string with an Impulse. As in the pick example in the bass section, the string is briefly excited when the note is played, and then the sound decays. A sustained pad sound requires an exciting agent that constantly acts upon the string; the appropriate object types are Bow or Bow wide (the string is played with either short or long, extended bow strokes), Noise (excited by a random noise signal) or Blow (excited by being blown—much like a clarinet or flute).

Please test the above mentioned object types one after the other, and move the Object 1 (Pickup) slider, responsible for the exact position of the exciting agent, up and down the string while you're playing. You will come to two conclusions. First, the sound is now sustained for as long as you hold the key down. Secondly, shifting the Object 1 slider, with the Bow type selected, results in the most pronounced sonic changes. This setting promises the most rewarding possibilities for varying the sound, and that's why this type has been chosen.

Recording an Envelope

The sonic variations created by the Bow type are very appealing when the virtual bow stroke is moved along the string. You can control this movement through the use of an envelope, thus creating the foundation of your pad sound.

It makes more sense (and it is more convenient) to record the envelope, rather than programming it, even if the latter is easily achieved with the graphic display.

Proceed as follows to record an envelope:

- 1 Move the Object 1 slider all the way to the left. Starting from this position (where it only generates an overtone-rich scratch), start animating it by using the envelope.
- 2 Locate the Envelope section in the lower right corner of the Sculpture window. Choose the first of the two envelopes by clicking on the envelope 1 button, if necessary. In the left part of the Envelope section, you can see two routing possibilities that allow you to assign a modulation target to the envelope.
- 3 Activate the first routing link by clicking on the 1 button, and choose Object1 Position as the modulation target in the Target menu. Set the modulation intensity to its maximum value by moving the horizontal slider all the way to the right.



The envelope can now be recorded. It is assumed that your MIDI keyboard has a modulation wheel which outputs the corresponding MIDI controller message (CC number 1) and that option 1 ModWh is selected for control of Envelope 1 (CtrlEnv 1) in the dark bottom edge of the Sculpture window.

4 Click on the button labeled *R*, located in the upper right corner of the Envelope section below Record Trigger, to prepare the envelope for recording. Choose the Note+Ctrl Movement option for the recording. This option specifies that the recording of the controller messages from the modulation wheel will begin the instant the first note is played.



- 5 Play a note when you want to start the recording, and move the modulation wheel slowly upwards while keeping the key on the keyboard depressed. Pay attention to the sound variations you create while moving the modulation wheel.
- 6 At the end of the recording, return the wheel to its initial starting position and, after you've released the depressed note, click on the *R* button to deactivate the recording mode.

You can now see the curve you recorded in the graphic display. You'll note that the curve arches exclusively above the zero axis—this is no surprise as the modulation wheel only sends unipolar values, which means between zero and up to a positive maximum value. As the Object 1 slider has already been moved all the way to the left end of the string, it can only be shifted all the way to the right by the envelope when the maximum modulation intensity is reached.

Play a note, or better yet, an entire chord and listen to the modulation you recorded. If you're not satisfied, you can, of course, repeat the procedure described above as often as you'd like. The maximum available recording time is 40 seconds. It is therefore possible, if you'd like, to control parameters and create modulations that extend far beyond the well-worn path of a simple ADSR envelope progression.

Note: By moving the junction points, you can edit the shape of the envelope when the need arises. By clicking in the empty part of the graphic display and moving the mouse to the right or left, you can zoom in and out of the display. When you activate the small Sync button, the junction points snap to a rhythmic grid (further information about these parameters and their functions can be found on page 498).

Choose Loop Alternate in the Sustain mode menu below the Envelope graphic display. As the sustain point is found at the end of the envelope, the envelope repeatedly travels from the beginning to the end and backwards from the end to the beginning, creating a continuous flux within the sound.



Summing up: you now have a rudimentary, but appealing and organic-sounding pad which will be used as the foundation for further shaping and refinement. Save this rough version of the sound as a basis for further experiments (Preset 0001 raw pad).

Increasing Stereo Breadth and Chorus

To give this very dry sounding pad a little more stereo breadth and chorus effect, use the trick discussed while creating bass sounds. To refresh your memory, modulate the Pickup positions, and assign them to the left and right channels.

Here's a quick description of the process:

1 Click-hold on the Pickup semicircle in the Spread control element, and move the mouse upwards to separate the stereo pan positions of the Pickups. Both of the light blue dots should come to rest near the line that separates both semicircles.



2 Activate both of the modulation links in LFO1 by clicking on the buttons labeled $\it 1$ and $\it 2$. For the first link, modulate the position of Pickup A (Target: PickA Pos) with a small positive value and with the second link, the position of pickup B (Target: PickB Pos) with a small negative value (intensity at ± 0.15). Reduce the Rate of the LFO to about 0.3 Hz.



You will hear a pleasant beating or chorus effect in the sound, which makes it broader and more full, alleviating the unpleasant, dry character (preset 0002 Ifo>pick pad). Another unpleasant aspect is that the sound is too strong in the mid frequency range and could use some equalization. You can use the Body EQ to correct this.

Activate the Body EQ, and experiment with the Lo Mid Hi model (which is the standard setting). Try these settings: reduce Mid to –0.5 and set Mid Frequency to 0.37.



To give the pad a little depth, activate the Delay. Set the Delay Time to 1/4 and adjust the Xfeed knob to 30%. The pad now has a pleasant and unobtrusive ambience; you can leave the other Delay parameters at their original values (preset 0003 egfx pad).



Finally, to optimize the sound so that it is a little more animated. The end result you should be aiming for is subtle, which makes the jitter modulators the perfect tool for the job. The jitter modulators are basically LFOs that use a random waveform.

Using the jitter modulators to make the sound become more lively:

- 1 Activate the display for both of the jitter modulators by clicking on the Jitter button below the LFO.
- 2 Turn on the first link in Jitter 1 by clicking on the button labeled 1, and choose Object1 Timbre in the Target menu.
- 3 Adjust the Intensity to -0.40 with the slider below the Target menu, and reduce the Rate parameter to 1 Hz. There should be subtle inconsistencies in the pressure applied by the bow to the string. To better recognize the effect, temporarily increase the Intensity level.
 - You can use the second jitter modulator for random position deviations with the modulation target Pickup Pos A+B (pickup position A and B).
- 4 Activate Jitter 2 and choose the Pickup Pos A+B setting in the Target menu.



5 Set the slider under the Target menu to an Intensity of about 0.2, and adjust the Rate dial to 1.5 Hz. As you increase the Intensity, the sound develops a distinct clinking or rattling—use this effect to taste. You now have a satisfactory pad sound, which you should leave alone at this point, even though a few Sculpture features such as; the Filter and the Waveshaper lie idle, not to mention the two additional Objects—but sometimes it's smart to quit while you're ahead.

The last function, morphing, has been saved for the end. Follow this section to bend and twist your pad sound (a little or a lot).

Morphing

In the middle of the lower part of Sculpture's window you can see the Morph Pad. In each corner, it can contain a different setting for a diverse number of parameters and by moving the red ball—which can be seen in the center of the Morph Pad—you can crossfade between these settings and morph the sound.

Choose the Paste to all Points command in the shortcut menu (accessible via a Controlclick on the Morph Pad) to copy the current patch setting into all four corners of the Morph Pad. If you move the ball in the Morph Pad now, you won't hear any changes in the sound because the settings (in each corner) are all the same. Not for long ...



To vary the sound with the Morph Pad:

- 1 When you move the ball to one corner, the corresponding partial sound is selected; you can recognize this by the gray blue arches that light up in the graphic display. Choose each of the four corners one after the other, and vary the sound by altering several parameters directly in Sculpture's GUI.
- 2 Use of the Material Pad allows you to achieve an especially noticeable variation: carefully move the ball around in the Material Pad, and try to find a position where your pad sound takes on a new and interesting character. Also try the extreme corners, for example.

As soon as you've chosen different settings for the Morph Pad corners (A to D), moving the morph ball will create marked sound variations—even though the intermediate stages will not all exhibit a tonal character. You can automate the morphing process by assigning two MIDI controllers to the MorphX and MorphY menus at Sculpture's bottom edge. You can also automate the Morph Pad using a recorded envelope—you'll find further information about this on page page 496.

Surrounding the Morph Pad, you'll find a randomizing function which randomly varies sounds to a chosen intensity level (or amount of randomization). This is especially useful for subtle changes to natural sounds, but it can also provide for rewarding variations to synthesized sounds as well.

To use the randomizing function:

- 1 On the left side of the Morph Pad, choose the number of corners that are to be varied by selecting one of the cubes.
- 2 Use the slider to the right of the Morph Pad to adjust the intensity of the random deviations.
- 3 Perform the randomization by clicking on the *Rnd* button. When you next move the morph ball, you'll hear the variations you just created.

This ends the programming tutorial. By demonstrating how to create basic sounds, detailed emulations of various electric bass sounds and explaining how to approach the generation of synthesized sounds, it is hoped that you now have an insight into the interplay of Sculpture's functions and parameters. Good luck with creating your own sonic sculptures!

Ultrabeat 29

Ultrabeat is a rhythm synthesizer that incorporates a step sequencer.

Ultrabeat's synthesis engine is optimized for creating electronic and acoustic drum and percussion sounds. It includes an impressive diversity of synthesis engines: phase oscillators, sample playback, FM (frequency modulation), and physical modeling. You can even use an audio side chain input as a sound source!

Ultrabeat features an integrated step sequencer, which you can use to create rhythmic grooves comprised of patterns. The sequencer displays running light style controls like those of classic drum machines. Ultrabeat's sequencer also offers step automation functions, allowing you to vary the timbre of the sound dependent on your performance dynamics or programmed rhythms. The sequencer plays an important role in the dynamic shaping of rhythms and sounds produced with Ultrabeat.

In addition to loading audio samples and its own settings, Ultrabeat can import EXS instruments. This allows you further sound design and rhythm processing options for EXS instruments, as well as letting you use Ultrabeat's intuitive drum mixer layout to play EXS drum kits.

Finally, Ultrabeat features truly comprehensive modulation functions—nearly every aspect of Ultrabeat can be modulated.

The Structure of Ultrabeat

Most software synthesizers offer one synthesizer per plug-in instance. Ultrabeat, however, places 25 independent synthesizers at your disposal. These synthesizers—called drum voices in Ultrabeat—are optimized for the generation of drum and percussion sounds.

The distribution of drum voices across the MIDI keyboard is simple and easily explained: the first (starting from the bottom) 24 MIDI keys are each assigned a single drum voice. The 25th drum voice is an exception, and can be played chromatically.



You can compare Ultrabeat to a drum machine that features 24 drum pads plus a built-in keyboard.

Ultrabeat's 24 drum pads are assigned to the first 24 keys of a standard MIDI keyboard (corresponding to MIDI notes C1-B2). This MIDI note mapping is compatible with the Roland GM Drum MIDI note mapping. The chromatic keyboard for the 25th synthesizer begins (lowest note in the range) at C3.

Note: If your MIDI keyboard is not large enough or does not support octave shifting to play Ultrabeat's lower or upper octaves, you can use the Transpose parameter in the Logic Pro Inspector to shift incoming MIDI notes.

For the sake of simplicity (and to stay with the drum machine analogy), this manual refers to the independent synthesizers (drum voices) as sounds which, combined, form an Ultrabeat drum kit.

Overview of Ultrabeat

Ultrabeat's user interface is divided into three functional sections.



- Assignment section: Displays all drum sounds of a drum kit, allowing you to select, rename, and organize them. It also includes a small mixer, which you can use to adjust each sound's volume and pan position.
- Synthesizer section: The majority of Ultrabeat's user interface is dedicated to creating and shaping individual drum sounds; in short, Ultrabeat's synthesizer. The parameters of the drum sound selected in the Assignment section are displayed in this Synthesizer section.
- Step sequencer: Here you can create and control sequences and patterns. A sequence triggers a single drum sound and can consist of up to 32 steps. A pattern contains the total sequences of all 25 sounds. You can use the step sequencer in place of, or in addition to, MIDI notes entering Ultrabeat (from Logic Pro) to control sounds. The step sequencer is described more fully in "The Step Sequencer" on page 590.

Loading and Saving Sounds

You can use the same methods to save and load settings in Ultrabeat as in all other Logic Pro instruments. For more information, see the Logic Pro 8 User Manual.



An Ultrabeat setting contains:

- The drum kit, which consists of 25 sounds, inclusive of assignment and mixer settings.
- The complete settings of all parameters for all 25 sounds.
- The sequencer settings and all 24 patterns, including the step automation, trigger, velocity, and gate rows for all 25 sounds.

Note: The joint recall of all of this data when you load an Ultrabeat setting makes a lot of sense as the musical effect of the patterns, especially those with sequenced gate and velocity parameters, are often intimately connected to the sounds being used.

The Assignment Section

The Assignment section displays all drum sounds of a drum kit, and also includes a small mixer. It allows you to:

- Select, organize, and name drum sounds
- Import drum sounds from other Ultrabeat settings or EXS instruments
- · Mix drum sounds

Selecting Sounds

The 25 sounds of an Ultrabeat drum kit are mapped to the onscreen keyboard found on the left hand side of the Ultrabeat window. The order of sounds on the keyboard corresponds to notes on a connected MIDI keyboard, starting with C1 for the first (bottom) sound.

Click on the name of the sound in the Assignment section to select the relevant drum sound. The sound's parameters will be displayed in the Synthesizer section to the right.

Make sure that the sound you're playing via MIDI is also the one you're editing: you can recognize the selected sound by the gray frame that appears around its name in assignment area. The corresponding key on the onscreen keyboard to the left of the sound name turns blue when it receives appropriate MIDI information. You can also click these keys to play the sound.



In this example, drum sound 2 is being played (the blue key) while drum sound 4 is selected (the gray frame).

You can also select sounds using MIDI note input. To do so, activate the Voice Auto Select button in the upper left hand corner of the Ultrabeat window.



Note: Given the obvious situation of Ultrabeat receiving numerous trigger notes from Logic Pro or the integrated step sequencer, automatic sound selection would result in constant, visually disturbing changes to the parameter display. To avoid this, the Voice Select function turns off automatically when a rapid succession of trigger notes occurs.

Naming and Organizing Sounds

Double-clicking on the name of a drum sounds opens its text entry field, allowing you to rename it. Press Return or click anywhere outside the text entry field to complete the naming operation.



Swapping and copying drum sounds within an Ultrabeat kit can be achieved via a drag and drop operation or via a shortcut menu.

To swap or copy drum sounds using drag and drop:

1 Click-hold the drum sound in the Assignment section (not on a button or menu).



- 2 Drag it to the desired position.
 - A simple drag and drop swaps the two drum sounds (including Mixer settings: volume, pan, mute, solo, and output configuration).
 Sequences are not swapped!
 - Holding Command during the drag and drop operation swaps the two drum sounds. Sequences *are* swapped!
 - Holding Option during the drag and drop operation copies the sound. Sequences are *not* copied!
 - Holding Option-Command during the drag and drop operation copies the sound. Sequences *are* copied!

To swap or copy drum sounds via a shortcut menu:

1 Control-click or right-click the drum sound in the Assignment section.



2 Choose the desired command from the shortcut menu that appears:

- Copy (Voice & Seq): This command copies the selected sound—inclusive of Mixer settings and sequences—to the Clipboard.
- *Paste Voice*: This command replaces the selected sound with the sound from the Clipboard, without changing its sequences.
- Paste Sequence > (submenu): The Paste Sequence submenu allows you to replace all, or individual sequences, of the target drum sound. Paste Sequence has no impact on the sound parameters of the drum sound. Pasting a single sequence replaces the currently active sequence (as set in the Pattern menu) of the target drum sound. This allows you to copy sequences into any of the 24 possible pattern locations.
- *Swap with Clipboard:* This command exchanges and replaces the selected sound with the sound from the Clipboard.
- Init > (submenu): The Init submenu contains a few starting-point (Init) sounds.
 Selecting one of them will replace the target drum sound. The Sample Init sound initializes the filter and pitch parameters to neutral settings as a starting point for programming sample-based drum sounds.

Note: Naturally, the shortcut menu's Paste and Swap with Clipboard commands require an initial Copy command (to place data in the Clipboard) before functioning.

These commands only affect the selected drum sound, all sequence and sound data of the other 24 drum sounds remain untouched.

Importing Drum Sounds and EXS Instruments

You can import drum sounds and sequences from other Ultrabeat settings into your currently active drum kit. You can also import sounds from EXS instruments: Ultrabeat reproduces the EXS layout as closely as possible. Layered EXS zones are set up as layered drum sounds, using the sample playback mode of Osc 2.

To open an Ultrabeat setting or EXS instrument in the import list:

1 Click on the Import field in Ultrabeat's upper left corner.



2 Browse to, and select the Ultrabeat setting or EXS instrument from which you want to import sounds, then click Open.

A list of all the sounds found in the selected setting or EXS instrument will be shown next to the Mixer section.



Note: If you import EXS instruments that include more than 25 sounds, you will be able to page through sets of 25 sounds using the up and down arrows to the left and right of the EXS instrument name at the top of the import list.



There are two methods to transfer sounds from the import list to the Assignment section. The simplest method is to drag sounds from the import list to the Mixer section. Holding Command while doing so includes all sequences. You can also use a shortcut menu.

To transfer sounds via a shortcut menu:

- 1 Control-click (or right-click) the sound's name in the import list.
- 2 Choose Copy (Voice & Seq) in the ensuing shortcut menu.
 This copies the selected sound and its sequences to the Clipboard.
- 3 Control-click (or right-click) the sound you want to replace in the current drum kit, then choose one of the following commands in the ensuing shortcut menu:
 - Paste Voice: Replaces the target drum sound, without changing its sequences.

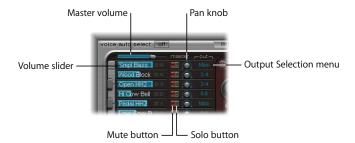
- Paste Sequence > (submenu): Use to replace all, or individual sequences, of the target drum sound. Paste Sequence has no impact on the target drum sound. Pasting a single sequence replaces the currently active sequence (as set in the Pattern menu) of the target drum sound. This allows you to import drum sound sequences into any of the 24 possible pattern locations. Note that these operations only affect the selected drum sound, all sequence and sound data of the other 24 drum sounds remain untouched.
- Swap with Clipboard: Swaps the content of the Clipboard with the target drum sound. As this command swaps both drum sound and sequence data, it's very useful for reorganization of your drum kit.

Dragging EXS Instruments Into the Assignment Section

You can also drag an EXS instrument directly onto the Assignment section to import the entire EXS instrument. This method does not allow paging through the EXS instrument if it contains more than 25 sounds of sample material. Ultrabeat will only map those sample zones and layers within Ultrabeat's drum sound range of C1 to C3, and ignore the rest.

The Mixer

The Assignment section contains a mixer for the 25 sounds found in an Ultrabeat drum kit. It allows you to adjust each sound's volume and pan position, and also offers a Mute and Solo button.



Volume

The individual volumes of all sounds are indicated by blue bars, providing a complete overview of all levels within the kit. You can adjust the volume of the sound, in relation to Ultrabeat's total output level, by dragging the blue bar beneath the sound name. The Master (Volume) control is found above the 25th drum sound and controls the levels of all drum sounds in the kit: put another way, the overall mix level of all drum sounds.

Mute

You can mute individual sounds in a drum kit by clicking the Mute button (M) to the right of the name.

Solo

You can listen to sounds in isolation by clicking the Solo button (S), found beside the Mute button.

Pan

The rotary knob to the right of the Mute and Solo buttons controls the placement of the signal in the stereo field (Panorama).

Individual Outputs

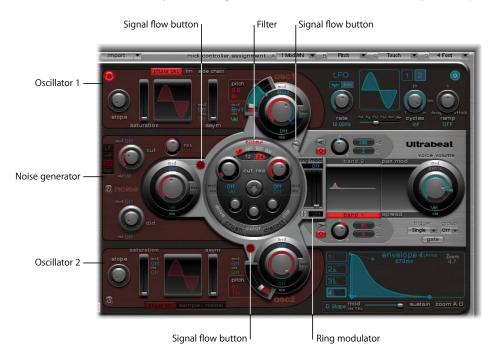
Ultrabeat features eight separate stereo and mono outputs, and can be inserted as a multi output instrument. In this situation, each drum sound can be independently routed to individual outputs (or output pairs) by using the Out(put Selection) menu, found beside the Panorama knob. Drum sounds that are routed to an output pair other than main are automatically removed from the main outputs. Control of output pairs other than 1–2 is via aux channels.

The Synthesizer Section

The Synthesizer section is the heart and soul of Ultrabeat. As noted above, each drum sound has its own Synthesizer section. Don't be intimidated by how many parameters the Ultrabeat synthesizer squeezes into one plug-in window; in fact, its signal flow is quite easy to understand.

The Signal Flow

Ultrabeat's synthesis engine is based on classic subtractive synthesis principles.



If you look at the Synthesizer section from left to right, you'll recognize the traditional structure and signal flow of a subtractive synthesizer. First, the basic tonal material is created by the oscillators and noise generator. A filter then takes away certain frequencies from the raw sound, followed by volume shaping (envelopes). The details of Ultrabeat's functions and their importance become more apparent when you look at the three dimensional interface, and recognize the different levels from front to back:

The elevated Filter section is in the middle. It's a large, round control object. Its placement and design are both symbolic and practical, as the Filter section plays a central role in Ultrabeat.

The filter receives its signal from the following sound sources: Oscillator 1, Oscillator 2, the noise generator and the ring modulator. Their output sections are displayed by four objects that sit adjacent to the filter (three round objects and the smaller, rectangular ring modulator to the right of the filter). One level down, you'll find the control elements for these sound sources.

On each of the objects that adjoins the filter you'll find a small, red signal flow button which indicates whether the signals should proceed through the filter or bypass it on their way to the Output section of the synthesizer. Along the output path to the right, the signals pass through two equalizers and a stage for stereo expansion or panoramic modulation.

The output of the drum sound is then passed along to the mixer that is integrated into the Assignment section (see "The Mixer" on page 561).

Common Parameters of Oscillators 1 and 2

To use Oscillator 1 or 2, you first need to turn them on. This can be done with the On/ Off button in the upper left corner of the Oscillator 1 or 2 section. When active, the button is red.



Note: When programming a drum sound, you can turn the individual sound sources on or off with the On/Off buttons. You can also listen to the individual components of the sound separately this way, and remove them from the patch if necessary.

The volume of Oscillator 1 or 2 is controlled by the Volume knob shown on the right edge of the Oscillator 1 or 2, respectively, sections. Volume can be modulated by the sources found in the Mod and Via menus. For more information, see "Modulation" on page 581.



The curved slider to the left of the Volume knob controls the pitch of the oscillators in half step intervals. If you press Shift, you can adjust the pitch in cent intervals.

The pitch value is displayed to the left of the slider. You can change the displayed value by click-holding directly on the value field, and moving the mouse vertically.

Pitch can also be modulated by the sources found in the Mod and Via menus.

You'll find a signal flow switch between each of the oscillators and the Filter section that controls routing (filter bypass button). Repeated mouse clicks will send the signal from the respective oscillator to the filter (filter bypass button turns red), or will bypass the filter and send it directly to the EQ section (filter bypass button remains gray).



The direction of the arrow on the filter bypass button illustrates the routing.

Note: The filter bypass button simply determines the signal flow. It doesn't turn the oscillator on or off. Use the oscillator's On/Off button to activate it (see above).

Unique Properties of Oscillator 1

Oscillator 1 can be switched between three different types of synthesis engines: Phase Oscillator, FM, and Side Chain (external audio input). This can be done by clicking the appropriately labeled buttons at the upper edge of the Oscillator 1 section.



Phase Oscillator

The waveform of the phase oscillator can be twisted with the Slope, Saturation, and Asymmetry parameters, and shaped into almost any basic synthesizer waveform. The effects of these three parameters are graphically illustrated in the waveform display within the Oscillator section. Setting all three parameters to zero values will cause the oscillator to produce a sine wave.



• *Slope*: Determines the slope or steepness of the waveform. The higher the Slope value, the steeper the waveform. The resulting sound takes on an increasingly nasal character as the incline becomes more vertical.

- Saturation: Increasing Saturation values clip the waveform, gradually molding its shape towards a rectangular waveform. This results in a corresponding increase in odd numbered overtones.
- Asym (Asymmetry): Tilts the waveform towards a sawtooth wave, making the sound more edgy. Asym can be modulated by the sources found in the Mod and Via menus. This allows you to create dynamic sound changes at the oscillator level. For more information, see "Modulation" on page 581.

Note: The basic waveforms of classic analog synthesizers can be easily reproduced with the phase oscillator: sine, rectangular, and sawtooth waves are each the result of setting the Slope, Saturation, and Asym parameters in different combinations. For example, if you set Slope and Saturation at their maximum, and Asym at its minimum, you will create a classic square wave. With Slope set to -0.20, Saturation at its minimum, and Asym at its maximum value, you will create a saw wave.

FM (Frequency Modulation)

In FM mode, Oscillator 1 generates a sine wave. Its frequency is modulated by the waveform of Oscillator 2. Please remember that Oscillator 2 must be switched on to do this. The more complex the Oscillator 2 waveform, the more partials will be created (by increasing the FM Amount) during the FM process. You can watch the display to see how the sine wave takes on an increasingly complex shape.



The FM Amount parameter can be modulated by the sources found in the Mod and Via menus. For more information, see "Modulation" on page 581.

Note: While the phase oscillator is well-suited to simulations of analog waveforms and analog-style sounds, FM mode offers bell-like digital tones and metallic sounds.

Side Chain

In Side Chain mode, Ultrabeat uses an external side chain input as its source for Oscillator 1. This means that you send any audio channel, aux, or live input channel through Ultrabeat's filters, envelopes, LFO, and step sequencer.



Once you select Side Chain mode for Oscillator 1, you need to select which channel you wish to use as your side chain input. You do this in Ultrabeat as you would with any plug-in with side chain input by choosing the desired channel in the Side Chain menu.



Side Chain mode greatly expands your creative options when using Ultrabeat. You can use an audio input from Oscillator 1 along with the synthesis engine of Oscillator 2 to create a part live audio, part synthesized drum sound. Since Ultrabeat's other drum sounds and sequencer operates normally, you can blend analog drum programming and side chain processing—for example, setting up one drum sound to step filter external audio input to a programmed groove.

Note: A side chain audio signal alone is not enough to trigger Ultrabeat's processors. To hear the side chain audio signal Ultrabeat needs to be triggered by MIDI or its internal step sequencer.

Unique Properties of Oscillator 2

Oscillator 2 can be switched between three different types of synthesis engines: Phase Oscillator, Sample, and Model. This can be done by clicking the appropriately labeled buttons at the lower edge of the Oscillator 2 section.



Phase Oscillator

Oscillator 2's phase oscillator operates nearly identically to the phase oscillator of Oscillator 1. The difference is that when in Phase Oscillator mode, Saturation can be modulated in Oscillator 2 instead of Asymmetry in Oscillator 1. This means that when both oscillators are in Phase Oscillator mode, they can produce different sounds.

For all other details of Phase Oscillator mode, please see "Phase Oscillator" in the "Unique Properties of Oscillator 1" section.

Sample

In Sample mode, Oscillator 2 uses an audio file as sound source.



Clicking the arrow in the upper left corner of the waveform display opens a menu, that allows you to load and unload samples, or to display the loaded sample in the Finder.

The Reverse arrow changes the playback direction of the sample (forwards/backwards).

The two Min/Max (Velocity) horizontal sliders determine the start point of the sample—dependent on the dynamics of the performance. Min determines the start point of the sample at the minimum velocity level (velocity = 1), Max at the maximum level (velocity = 127). If Min and Max are set to the same value, this corresponds to a static setting of the sample start point.

Both factory Ultrabeat samples and sounds imported from EXS instruments often consist of different layers that are velocity switched depending on the dynamics of your performance. The precise sample layer that incoming velocity values switch to is determined by the small Vel Layer slider on the right. This slider determines which layer is triggered at the minimal level (velocity = 1). The second small slider on the left determines which layer sounds at the maximum level (velocity = 127). If you have loaded a single sample that does not have multiple layers, the Vel Layer slider has no effect.

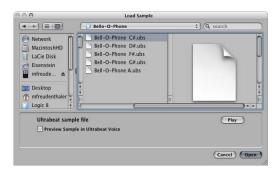
To load an audio file:

1 Click the arrow in the upper left corner of the waveform display, then choose Load Sample from the pop-up menu.



2 In the Load Sample window, browse to, and select the desired audio file.

A selection of multi-layer drum and percussion samples that were specially created for Ultrabeat and its function set are included with Ultrabeat. You can also load your own samples in AIFF, WAV, CAF, or SDII stereo interleaved format. It should be noted, however, that the velocity layering function is not available for such samples.



The Play button in the Load Sample window allows you to preview audio files (AIFF, WAV, SD2, CAF, UBS) before loading.

- Press the Play button to loop playback of the currently selected sample file. The sample is played directly from the hard disk, with no manipulation: filters, EQ, envelopes, and other synthesizer parameters are ignored.
- Clicking the button again stops playback.

You can audition multiple files by clicking Play once, and then stepping through the files.

Note: In the case of multi-layer UBS files, the sample is played with a fixed velocity of 75%. Independent auditioning of all layers is not possible. Only the layer addressed by this velocity value will be played.

The Preview Sample in Ultrabeat Voice option temporarily replaces the sample files in the currently selected drum sound. The drum sound is not directly triggered by activating this option, but it can as usually be triggered via MIDI notes (played notes, MIDI region events, or Ultrabeat sequencer events) while the Load Sample window is open and different files are being selected. The selected sample can be heard as part of the current drum sound, inclusive of all synthesizer processing. Once you find a sample you'd like to use, click Open to load it. Click Cancel to return to the previously loaded sample.

Important: Any effects inserted into the Ultrabeat instrument channel affect previews.

When saving a drum kit using the Settings menu, the location of the sample is saved with the setting. The Ultrabeat setting doesn't actually save the audio files themselves—only a reference to their location. If you load a setting that contains a reference to a sample that has been moved or erased, Ultrabeat will present you with a dialog box that requests you to find it. To avoid this problem, it is highly recommended that you use a dedicated Ultrabeat sample folder.

Model

This oscillator type offers a physical model of a string instrument for the creation of percussive sounds. The parameters at your disposal are based on the physical properties of a real string.



Two contrasting exciters, each with different sound characteristics, are available. You can switch between them with the corresponding buttons (Type 1 and Type 2).

Note: In Ultrabeat's Oscillator 2 model, an exciter is the agent or triggering device used to initiate the vibration of the string. It should not be confused with the effect plug-in of the same name.

In the Material Pad, you can set the Inner Loss and Stiffness string parameters. These parameters actually determines the material qualities of the physical model.

Inner Loss determines the damping of the string which, in the real world, depends on the material the string is made of (for example steel, glass, nylon, or wood). Damping primarily affects high frequencies, and forces the sound to become more muffled and smooth during the decay phase.

Stiffness controls the stiffness or rigidity of the string. In the real world, this depends on the material the strings are made of and their diameter (or, more precisely: their speed of vibration or response to being struck or plucked and so on). Rigid strings create an inharmonic vibration where the overtones do not represent whole number multiples of the fundamental frequency. These overtones are, in fact, slightly higher. Marked increases in rigidity (stiffness) ultimately transforms the string into a metal rod.

Along the x-axis of the Material Pad you'll find the value range for the Stiffness parameter, and the value range for the Inner Loss parameter along the y-axis. To adjust the parameters, click-hold on the dot in the Material Pad and move it.

Note: Click on the dot in the Material Pad while holding down Option if you want to return the string parameters to their default values.

To the right of the Material Pad you'll find the Resolution parameter. In contrast to the other parameters of the Model oscillator, Resolution does not reproduce a pre-defined real-world property of the physical model, but affects the modeling process itself: higher values lead to an improved calculation resolution which results in more overtones. Lower values reduce the precision of the calculations, leading to fewer overtones and often to inharmonic spectra.

The Ring Modulator

The ring modulator functions as its own sound source; its signal can bypass or be sent into the filter, independent of Oscillators 1 and 2. Its volume can also be regulated. Please note that both oscillators need to be switched on to use it.

The sound of the ring modulator is largely dependent on both of the oscillators, as it modulates the output signals of both. Parameter changes, especially the tuning relationships of each oscillator, have a direct effect on the sound of the ring modulator. The individual volumes of the oscillators have no effect on the process of ring modulation.

The ring modulator doesn't have an On/Off button like the oscillators. It is activated by clicking directly on the Ring Mod label itself. When switched on, the label is red and when off, it's gray.



Note: As the ring modulator needs the signals of both oscillators to produce its output, the ring modulator is muted when one of the oscillators is switched off. If you want to hear the ring modulator's signal in isolation (in order to better judge your settings), temporarily set the volume of both oscillators to 0.

The slider adjusts the output volume of the ring modulator. Volume can be modulated by the sources found in the Mod and Via menus. For more information, see "Modulation" on page 581.

Between the ring modulator and the Filter section you'll find a signal flow switch that controls the routing (filter bypass button). Repeated mouse clicks will send the signal to the filter (filter bypass button turns red), or bypass the filter and send it directly to the EQ section (filter bypass button remains gray). The direction of the arrow on the filter bypass button illustrates the routing.

Note: The filter bypass button determines the signal flow. It doesn't turn the ring modulator on or off. Use the Ring Mod field for this (see above).

The Noise Generator

The fourth synth engine is the noise generator.

Noise contains—in a technical sense—all tonal frequencies; that's why human hearing can't recognize any tonality in a noise signal. Despite this (or as a direct result of it), noise is an indispensable ingredient when creating drum sounds. For this reason, Ultrabeat's noise generator is outfitted with extensive features.



To use the noise generator, you first need to turn it on. This can be done with the On/Off button. When in an active state, the button is red.

The volume of the noise generator is controlled by the Volume knob. Volume can be modulated by the sources found in the Mod and Via menus. For more information, see "Modulation" on page 581

The noise generator has its own filter which functions independently of Ultrabeat's multimode filter. The four type buttons LP, HP, BP, and Byp allow you to switch the filter between lowpass, highpass, or bandpass modes, or deactivate it (Byp).



The names of the filter types illustrate how they work: A lowpass (LP) filter allows frequencies that are lower than the cutoff frequency (see below) to pass. This filter type dampens higher frequencies, and makes the sound less sharp and bright.

A highpass (HP) filter has exactly the opposite effect. It filters out the lower frequencies while leaving the higher frequencies untouched.

The bandpass (BP) filter only allows a certain frequency range (a frequency band) centered around the cutoff frequency to pass. It can be used in the upper, as well as at the lower, end of the frequency spectrum to reduce the highs and lows of a sound.

The Cut knob determines the cutoff frequency, defining the point in the frequency spectrum where reduction begins. Depending on the type of filter you select, you can make a sound darker (LP), thinner (HP) or more nasal (BP) by adjusting the Cut value.

Cutoff can be modulated by the sources found in the Mod and Via menus. For more information, see "Modulation" on page 581.

Increasing Resonance boosts frequencies that surround the cutoff frequency. Values range from 0 (no increase) to self-oscillation of the filter at high Resonance values.

Note: Self-oscillation is typical of analog filter circuits. It occurs when the filter feeds back into itself and begins to oscillate at its natural frequency, when high resonance values are used.

Dirt is a parameter developed especially for the noise generator. Turning up the Dirt knob roughens up the pure, white noise appreciably, making it more grainy.

Note: The Dirt parameter is especially effective at high Resonance values.

Between the noise generator and the Filter section you'll find a signal flow switch that controls the routing (filter bypass button). Repeated mouse clicks will send the signal to the filter (filter bypass button turns red), or bypass the filter and send it directly to the EQ section (filter bypass button remains gray). The direction of the arrow on the filter bypass button illustrates the routing.

Note: The filter bypass button determines the signal flow. It doesn't turn the noise generator on or off. Use the On/Off button for this (see above).

The filter bypass button has no effect on the independent filter contained in the noise generator. This is deactivated with the Byp button in the Noise Generator Filter section.

It is therefore possible to filter the noise generator signal twice. In many instances, you may want the noise generator signal to bypass the main filter, freeing the main filter for other duties—an important element when programming drum sounds.

The Filter Section

The output signals of both oscillators, the ring modulator, and the noise generator are passed on to Ultrabeat's central Filter section (if they haven't bypassed it through use of the various filter bypass buttons). The Filter section offers a multimode filter and a distortion unit.



The order that sounds are passed through the filter and distortion unit is determined by the arrow found at the "equator" of the Filter section. Clicking the arrow switches between the arrow up (distortion, then filter) and arrow down (filter, then distortion) settings.



Note: A description of Ultrabeat's filter parameters and a discussion on the basic concepts of subtractive synthesis and analog filters follows. If you're new to synthesizers, please read "Synthesizer Basics" on page 621 for more information.

The Multimode Filter

A click on the word "filter" in the upper middle section activates or deactivates the multimode filter. In a deactivated state (the word filter is gray, and red when active), all synthesis engine signals pass the filter unprocessed and are forwarded to the distortion unit.

The multimode filter offers the following filter types: lowpass (LP), highpass (HP), bandpass (BP), and band rejection (BR).

You can switch between the filter types by clicking the corresponding button directly beneath the word filter.



The names of the individual filters illustrate their function: A lowpass (LP) filter allows frequencies lower than the cutoff frequency to pass. It removes (cuts) the highs of a sound, making it darker and less bright.

A highpass (HP) filter allows frequencies higher than the cutoff frequency to pass. The lows of the sound are cut.

A bandpass (BP) filter allows a frequency band centered around the cutoff frequency to pass. Frequencies that lie further away (the lows and highs outside the band) are filtered out. A sound with a lot of mid frequency content results.

The abbreviation BR stands for band rejection filter. In this mode, the area (the frequency band, to be more exact) around the cutoff frequency is filtered out while frequencies that lie further away (from the cutoff frequency) are allowed to pass. The mid frequencies become softer and the low and high frequencies remain unchanged.

Below the filter type buttons, you'll find two buttons labeled 12 and 24. These allow you to select the slope of a filter. All of Ultrabeat's filter types offer a filter slope of either 12 or 24 dB/octave.

Note: Filters don't completely remove the parts of the signal that you wish to filter out and always work with limited precision in the selected band. The steepness or slope is measured in decibels of damping per octave (dB/oct). Frequencies that are located close to the cutoff frequency are generally reduced less than those that are farther away. The higher the slope value, the more apparent the level difference is between frequencies that are nearer the Cutoff frequency and those that are further away from it.

The Cut knob determines the cutoff frequency of the filter.

Note: Adjusting the Cutoff frequency can make a sound darker (LP), thinner (HP), more nasal (BP) or more transparent (BR), dependent on the type of filter chosen.

Cutoff can be modulated by the sources found in the Mod and Via menus. For more information, see "Modulation" on page 581.

Increasing Resonance boosts frequencies that surround the cutoff frequency. Values range from 0 (no increase) to self-oscillation of the filter at high resonance values. Resonance can also be modulated by the sources found in the Mod and Via menus.

Note: Self-oscillation is typical of analog filter circuits. It occurs when the filter feeds back into itself and begins to oscillate at its natural frequency, when high resonance values are used.

The Distortion Unit

Depending on the order determined by the arrow in the Filter section, the distortion unit is inserted either before or after the multimode filter. It provides either a bit crusher or distortion effect.



The desired mode is activated by clicking on the Crush or Distort button. The active effect is indicated in red. If neither button is red the distortion unit is bypassed

The distortion effect is modeled on an analog distortion unit, which distorts the sound by overdriving the level. The bit crusher uses a digital process that deliberately reduces the digital resolution of the sound (measured in bits), achieving an intentional digital coloration of the sound. Both methods lead to distortions that are as tonally divergent as the two approaches. Distortion offers a more analog feel while the bit crusher can't hide its digital origins (nor is it supposed to!).

The bit crusher and distortion effect are adjusted with the same three dials:

- *Drive*: Turning this parameter up increases the degree of distortion.
- *Color:* Determines the basic sound of the distortion. Higher values help you achieve a brighter sound and lower values lead to a darker, warmer tone.
- Level/Clip: The output volume is set here (Level) when in Distortion mode. In Bit Crusher mode, this dial determines the level required before distortion (Clip) begins.

Output Section

Depending on the status of each filter bypass button, the output signals of both oscillators, the ring modulator and the noise generator are routed either; directly or via the Filter section to the Output section of Ultrabeat. The Output section passes signals through both equalizers (EQ) and the Pan Modulation and Stereo Spread section (in a pre-configured order) before the final level is set, and the trigger behavior (of the signals) is adjusted.

Two Band EO

Both equalizer bands have almost identical features. Their parameters are explained jointly, but you can, of course, adjust band 1 (the upper EQ in the Output section) and band 2 separately.



Clicking on the Band 1 and Band 2 labels turns the individual band on or off. When active, the field is red. If neither EQ is activated, the signal passes through unaffected.

The EQ type buttons switch between two different types of EQs: shelving and peak.



In shelving mode, all frequencies above or below the set frequency are either increased or reduced. In peak mode, only frequencies located near the set frequency are affected.

The shelving EQ is activated by clicking the upper of the two EQ type buttons. The peak EQ is activated by clicking the lower of the two EQ type buttons.

Band 1 offers a low shelving EQ while band 2 features a high shelving EQ. Low shelving means that the frequencies below the set frequency are affected. High shelving affects frequencies above the set frequency.

Note: Shelving EQs function similarly to synthesizer lowpass and highpass filters. The fundamental difference: while lowpass and highpass filters merely dampen certain frequencies (filter them out), shelving EQs also allow these frequencies to be boosted.

The EQ Gain knob is bipolar. Positive values boost a certain frequency range as determined by the EQ type and Hz settings. Negative gain values lower the gain of the frequency range. If the Gain knob set to the mean value of 0, the EQ has no effect.

Note: You can also return this knob to its neutral position by Option-clicking on it. Alternately, you can click on the tiny 0 above the EQ Gain knob.

The frequency is set by click-dragging vertically on the Hz parameter field. This determines the frequency range to be boosted or reduced.

Note: Option-clicking the Hz parameter returns its value to a neutral position. This is 200 Hz for the first band and 2000 Hz for the second. The selection of these default frequencies was made in accordance with the different shelving characteristics of each frequency band. Band 1 is designed to filter low frequencies and band 2, high frequencies.

The Q factor is regulated by click-dragging vertically on the Q parameter field. The effect of Q on the sound is heavily dependent on the selected EQ type:

- With shelving filters, as the Q value goes up, the area around the threshold frequency becomes more pronounced.
- With the peak EQ, Q determines the width of the frequency band selection: low Q values select a broad band while high Q values select a very narrow band to be boosted or reduced with the Gain control.

Editing the EQ Bands Graphically

The EQ bands each have their own display which shows changes on a frequency response curve. The display provides immediate access to the Gain, Hz, and Q parameters. Just grab the graphical frequency response curve with the mouse, and alter it by moving the mouse vertically or horizontally! Horizontal moves change the EQ frequency, and vertical moves influence the Gain. At the peak (maximum point) of the EQ, a handle can be dragged vertically to change the Q factor.

Pan Modulation and Stereo Spread

The EQ's output signal is passed along to the Pan Modulation and Stereo Spread section. In the Pan Modulation and Stereo Spread section, the placement of the sound in the stereo field (set in the Assignment section's mixer) can be modulated (Pan Modulation mode), or the stereo basis of the sound can be broadened (Stereo Spread mode). Activate the desired mode by clicking on the appropriate button (Pan Mod or Spread).



If neither mode is activated, the signal passes through unaffected.

Pan Modulation

Pan Modulation varies the panorama position of a drum sound dependent on a Mod and Via source. The modulation set here is relative to the panorama position set in Ultrabeat's mixer.

The panorama position set in the mixer is represented here by a thin, red line. To the left and right of the line, small sliders (and corresponding menus) allow the adjustment of the Mod and Via modulation routings.



Note: You cannot directly grab and move the red line (that represents panorama position) shown in this section. In order to move the line, rotate the pan knob in the Mixer section.

Stereo Spread

Stereo Spread broadens the stereo image, making it wider and more spacious.



Low Frequency applies the (spreading) effect to the bass frequencies: the higher the value, the more prominent the effect becomes. Hi Frequency allows you to apply the effect to the high frequencies.

Voice Volume

This rotary knob adjusts the output volume of the individual drum sounds. To be more exact, you are controlling the Voice Volume with Env 4, thereby adjusting the maximum volume level attained after the attack phase of Env 4.

The effect the envelope has on Voice Volume can also be modulated by a Via source.

Note: The leveling stage for Voice Volume precedes the sliders in the mixer. This approach allows the starting volume of the individual drum sounds to be set independently of their relative levels in the drum kit mix.

Trigger and Group Menus

The manner in which Ultrabeat reacts to a succession of incoming notes is individually defined for each sound. These parameters are found in the Output section, below the Voice Volume knob.



Clicking the button below the Trigger label opens the Trigger menu, allowing you to choose between Single and Multi trigger modes.

- Single: A new trigger note cuts off the (same) note that is currently playing.
- *Multi:* When a new note is played, preceding (currently playing) notes continue to decay in accordance with their respective amplitude envelope settings (Env 4).

Clicking the button below the Group label opens the Group menu, allowing a choice between the Off and group 1 to 8 settings. If two different sounds are assigned to the same group, they will cut each other off. A typical use of this facility is when you're programming hi-hat sounds: when playing a real hi-hat, the closed hi-hat note cuts off and mutes the ringing of the open hi-hat. This function is often referred to as hi-hat group mode.

Note: While in Single Trigger mode, only the currently sounding note of the same sound is cut off. A sound that is assigned to a group cuts off all other sounds (regardless of note) in the group.

Clicking on the Gate button turns the Gate function on and off. If active, the sound is immediately cut off as soon as the MIDI note is released (MIDI note off), regardless of envelope settings.

Note: The Gate function ensures that a specific sound does not play beyond a note off event, as defined in the sequencer. A corresponding rhythmic definition of the exact note off time is achieved with the Gate Length parameter in Ultrabeat's step sequencer. Logic Pro's sequencer allows you to quantize note off events, or precisely edit them manually. Note length can be an important creative element when programming rhythm tracks.

Modulation

Numerous sound parameters can be controlled dynamically (modulated) in Ultrabeat. Ultrabeat provides two powerful LFOs, four envelope generators, velocity, and four freely-definable MIDI controllers as modulation sources. The setting of modulation routings follows a universal principle that is explained in this chapter.

The Principle of the Modulation Routings

Ultrabeat's modulation routings feature three key players:

- The synthesizer parameter that you want to modulate (the modulation target)
- The source of the modulation (the modulation source)
- A second modulation source that can influence the intensity of the first modulation—this modulation source is called Via modulation.

Mod and Via Modulations

You can modulate a sound parameter using an adjustable value (called modulation depth) with the Mod parameter. You can choose between two LFOs, four envelope generators, and Max, as sources for this modulation.

Via allows you to further tailor the modulation effect. To explain, the modulation depth of the first modulation (Mod) can be modulated by a separate, independent source. The intensity of this effect is set with the Via parameter. The sources for Via modulations include velocity and four freely definable MIDI controllers.

A typical application for Via function usage is increasing a pitch sweep as you play at higher velocities, for example. To this end, an envelope (Env) is chosen as the Mod source for the pitch of an oscillator, and velocity (Vel) is chosen as the Via source. The firmer the key is played, the higher (in pitch) it will sound—this is typical of synthesized tom-tom sounds.

Exceptional Modulation Features

The design of the Mod and Via modulation options are substantially different when comparing Ultrabeat and other, more traditional, synthesizers. Ultrabeat's Mod and Via options are given a target value that can be reached by modulation of the respective target parameters, rather than indicating a modulation or effect intensity as a percentage. The result of such modulation routing—its minimum and maximum effect on the modulated parameter—can be set simply, and grasped at a glance, making the complex subject of primary and secondary modulations an intuitive task.

Have a look at the following example to better understand how this works:

The Cut (Cutoff) parameter has a mean (default) value of 0.50. It's not being modulated yet as no modulation source has been selected in either the red Mod or blue Via menu (set to Off).



As soon as a modulation source is selected in the Mod menu (Env 1 in this example), the ring around the rotary knob is activated. Grabbing and moving this ring with the mouse allows you to set the value that this parameter will be increased to by the Mod source (0.70 in the example).



As soon as a modulation source is selected in the Via menu (Vel in this example), a movable slider appears on the Mod ring. Grabbing and moving this slider with the mouse allows you to set the maximum modulation value that can be reached through use of the Via source (0.90 in this example).



So much for the settings. What do the markings found around the Cut knob mean, and what is happening to the sound?

The Mod and Via controls indicate the minimum and maximum values that the modulated parameter can attain (in comparison to the mean value). Put another way, the result of the modulation. These controls don't show—as is customary with other synthesizers—a percentage value describing their intensity, but state very clearly: "These are the minimum and maximum possible values of the modulated parameter."

Back to the example; the frequency of the filter is set to the mean value of 0.50. When the Mod source Env 1 enters the equation, the Env 1 envelope generator drives the Cut value up from 0.50 to 0.70 (during the attack phase) and back down to 0.50 (during the decay phase).

Note: You can view the exact values in the help tags that appear when you grab the individual handles of various parameters.

If the Via source Ctrl A is introduced, the following interplay occurs: when Ctrl A remains at its minimum value, nothing changes (yet); Cutoff continues to be modulated between values of 0.50 and 0.70 by the envelope. A maximum value for Ctrl A causes the envelope generator to vary the parameter between the values of 0.50 (the mean value) and 0.90 (the Via amount).

You can see, at a glance, the degree of maximum influence on basic parameters by the Mod and Via modulation sources: the area between the Mod and Via points shows the amount that the modulation depth can be (further) altered by the Via modulation source. In the example, the Cutoff can reach values between 0.70 and 0.90 depending on the value being sent by Ctrl A.

Here's another example:







Cutoff is again set to 0.50, Env 1 now drives the value down to 0.25, and a maximum Ctrl A value reduces the Cutoff frequency down to 0.

Here is another example that illustrates the simplicity and speed of Ultrabeat's modulation options:



In this example, you won't just be changing the modulation intensity of Env 1 (which affects Cutoff) with the dynamics of your performance (Vel), but you'll also control its direction as well. Try this setting in Ultrabeat to create some extremely interesting sounds.

Setting the Modulation Routing

Clicking on the Mod label opens the Mod menu. This is where you can choose one of the LFOs or envelope generators (Env) as a modulation source.



The Off setting deactivates the Mod routing, and the Mod ring can no longer be adjusted. In this situation, no Via modulation can occur either (this is because Via no longer has a modulation target) and the Via slider disappears.

Note: The Max setting produces a static modulation at maximum level. When the Mod value is set to Max, the Via parameter is routed directly to the modulation target. This way, velocity can be used as a direct modulation source, even though Vel is not available as a source in the Mod menu.

Another example would be to set up an external MIDI fader unit with Ctrl A, B, C, or D (see below). You could then use the Max item in the Mod menu to route the respective Via source—Ctrl A, B, C, or D—to the parameter you'd like to control with one of the faders on your MIDI fader box.

Clicking on the Via label opens the Via menu. This is where you choose the Vel or Ctrl A to Ctrl D parameters.



Vel represents velocity.

Ctrl A to Ctrl D are four continuous controllers that can be assigned to four external MIDI controllers. These assignments are made in the MIDI Controller Assignments area at the upper right edge of the Ultrabeat window (see below). The assignments apply to all sounds in the current Ultrabeat plug-in instance.

MIDI Controllers A-D

In the MIDI Controller Assignments area at the upper edge of the Ultrabeat window you can assign a standard MIDI controller to each of the four controller slots: Ctrl A, B, C, or D. Ctrl A, B, C, and D can be used as Via modulation sources within Ultrabeat.



Use these assignments to set up your external MIDI controller hardware to operate with Ultrabeat. As examples: aftertouch or the modulation wheel of your MIDI keyboard.

Note: All MIDI Controller Assignment menus feature a Learn option. If this is selected, the parameter will automatically be assigned to the first appropriate incoming MIDI data message. The Learn mode features a 20 second time-out facility: If Ultrabeat does not receive a MIDI message within 20 seconds, the parameter will revert to its original MIDI controller assignment.

LFO 1/2

Among other items, two LFOs are available as modulation sources in the Mod menu.

LFO is the abbreviation for Low Frequency Oscillator. The LFO signal is used as a modulation source. In an analog synthesizer, LFO frequency generally ranges between 0.1 and 20 Hz, which is outside the audible frequency spectrum.

Note: The speed of the LFO in Ultrabeat can reach up to 100 Hz which, when compared to analog synthesizers, offers a number of far-reaching possibilities.

Ultrabeat has two LFOs that offer identical feature sets. The parameters for each are described jointly; you can, of course, adjust LFO 1 and LFO 2 completely independently of each other.



The buttons labeled 1 and 2 select the corresponding LFO, allowing adjustments of each LFOs parameters. The blue On/Off button activates and deactivates the selected LFO.

The LFO section display shows the LFO waveform, the shape of which is governed by the Shape slider located underneath it. Dragging the slider from left to right causes the waveform to fluidly morph from a sine to a triangle, and then finally to a square wave (with variable pulse width), including all variations in-between. At the far right hand position of the Shape slider, the LFO produces random waveforms.

The LFO speed (Rate) can be set independently (Free) or synchronized (Sync) to the Logic Pro tempo. Clicking either button once activates the corresponding mode.

The Rate knob determines the speed of the LFO. Depending on the Free/Sync setting, Rate is displayed in either Hz or musical (measure) units.

The Ramp knob determines whether the output signal of the LFO is faded in or out. Ramp works in a bipolar fashion: turning it to the right increases the Attack time of the LFO, turning it to the left decreases the Decay time. In its middle position, Ramp has no effect on the LFO. The Ramp value is displayed in milliseconds in its parameter field.

An LFO normally oscillates constantly. On percussive signals it can, however, be interesting to limit the LFO cycles to a defined number. Ultrabeat allows you to set the number of LFO cycles with the Cycles parameter. After completing the defined number of cycles, the LFO stops oscillating. Try small Cycles values, and route the LFO to Osc Volume to create typical drum flams or hand claps.

The range of Cycle values extends from 1 to 100. Turning the knob to its maximum value (all the way to the right) results in a permanent oscillation (an infinite number of cycles). A Cycle value of 1 allows the LFO to function as an additional (albeit simple) envelope generator.

The Cycle parameter can also determine whether the LFO (waveform) is started from the beginning (at a zero-crossing point) with each note trigger, or whether it simply continues oscillating. A Cycle value of Inf (Infinity) forces the LFO to run freely. It is not reset by incoming MIDI note on messages. When Cycle is set to values under 100, the LFO will be reset by each new MIDI note on message (Note On Reset).

It's really a question of taste as to whether or not you choose to trigger an LFO cycle from the same spot or just allow it to oscillate freely, regardless of phase. The random element of free-running LFOs can make many sounds fatter. This, however, can come at the expense of a percussive attack—an often undesirable quality in a drum synthesizer.

Note: You can, of course, use minor shifting of the LFO phase—with the Cycle value set to Infinity—to your advantage, adding an analog character to a drum sound, for example.

Env 1 to 4

Further modulation sources available to you in the Mod menu include four identically specified envelope generators. Envelope parameters are described in this section.

Note: In addition to its potential use in the Mod menus of various sound parameters, Env 4 is permanently connected to the Voice Volume. In other words, Ultrabeat has a hard-wired volume envelope generator.

Structure of the Envelope Generators

The default setting of the envelope generators is known as the *one shot* envelope mode: after a key is pressed (note on message), the envelopes run their course, regardless of how long the note is held. This setting is ideal for percussive signals because it allows simple emulations of the behavior of natural percussion sounds. For special cases such as sustained pad or cymbal sounds, you can activate a sustain mode where the envelopes take the lengths of the played notes into account.

Editing the Envelopes Graphically

Before delving into the individual parameters, please take the time to familiarize yourself with the graphical depiction of an envelope, shown below.

Ultrabeat's envelope display provides a new type of envelope design, consisting of Bezier curves in which two segments—attack and decay—make up the entire envelope.



In the envelope graphic, you can see various junction points of two different sizes. Both of the larger handles on the x-axis (the horizontal, or time axis) control the attack and decay times, respectively. A vertical line extends up from the first of the two handles, and divides the envelope into an attack and decay phase. Both segments each have two small curved junction points. You can move these in any direction to deform the contour of the envelope, and freely shape its amplitude.

In order to move the curved junction points, simply grab them and drag them to a new position. Experiment with the different junction points and you'll quickly realize that handling is very intuitive. You can also directly click-hold and drag any curve edge on the envelope itself.

Envelope Parameters

In order to edit the envelope parameters, first select an envelope by clicking on the desired 1 to 4 buttons. The parameters of the corresponding envelope can now be directly changed in the envelope display window.

Attack Time

Attack time defines the period of time the envelope needs to reach its maximum value. This is measured from the instant you press a key (note on). This period is called the attack phase.

Grab the attack junction point (the left-most of the two handles found on the x-axis) with the mouse, and move it to shorten or lengthen the attack time.

Note: To change the shape of the envelope in the attack phase, you can edit both junction points found in this segment. It is also possible to directly grab the curve with the mouse and alter it.

Decay Time

Decay time defines the period of time the envelope needs to fall back to a zero amplitude, after it has reached its maximum value (defined in the Attack phase). Moving the second junction point on the x-axis shortens or lengthens the decay phase.

Note: To change the shape of the envelope in the decay phase, you can edit both junction points found in this segment. It is also possible to directly grab the curve with the mouse and alter it.

Envelope Modulation

The time and shape of the envelopes can be modulated by velocity. Clicking in the menu field below the envelope 1 to 4 buttons opens the Env Mod menu. Choose either Time or Shape of the (A)ttack or (D)ecay phase as the modulation target.

The intensity of modulation is adjusted with the mod slider below the envelope display.

Note: When you modulate Shape, low velocity values lead to a sagging envelope shape, while higher values cause the selected envelope segment to bulge.

Note: When you modulate Time, increasing velocity values lead to a reduction in length of the envelope segment. Lower velocity values increase the length of the envelope segment.

Sustain

Activation of the Sustain button causes a red handle (and vertical line) to appear on the x-axis. This can be moved horizontally—but only within the decay segment area. The amplitude that the envelope reaches at the Sustain junction point is retained until the MIDI note is released. After receiving the MIDI note off command, the envelope continues for the remaining decay time.

Note: If the Sustain button is not activated, the envelope functions in one shot mode, and the note length (MIDI note off command) is disregarded.

Zoom (to fit)

When you select the Zoom button, the envelope is enlarged to fill the entire width of the display, making it easier to adjust junction points and curves. The graphic display is quickly redrawn after any change is made to the Attack or Decay values.

Note: When the Zoom function is selected, the decay junction point can be dragged beyond the right-hand edge of the display area, in order to lengthen the decay time. After you release the mouse button, the envelope graphic is automatically resized to fit the display area.

Zoom A/D

The Zoom A button only shows the attack phase across the entire width of the display, and the Zoom D button only shows the decay phase. This allows easier and more accurate edits to envelope shapes, even down to millisecond values.

Modulation Target Display of LFOs and Envelopes

Ultrabeat's user interface features a function that makes finding the modulation targets of the LFOs and envelopes faster: Simply press the numerical field of the desired modulation source to highlight all of its modulation targets.

Highlighted modulationtarget of Envelope 1



Click here to highlight all I modulation targets of Envelope 1.

The Step Sequencer

The integrated step sequencer allows all Ultrabeat sounds to be combined in sequences, based on patterns. Its design and use (step programming input) are based on analog predecessors. Unlike Ultrabeat's analog predecessors, however, you can also program automated changes for nearly every synthesizer parameter!

Dependent on your personal taste and favored musical style, you'll want to control Ultrabeat from either the integrated step sequencer or from Logic Pro, when programming rhythms. Combining both sequencers is also possible; they both can be active at the same time, and are automatically synchronized with each other. The Logic Pro tempo stipulates the tempo of Ultrabeat's internal step sequencer.

If you are not familiar with the concept of step sequencing, please read the following brief look back at the early days of sequencing. This will help you understand the design of Ultrabeat's step sequencer.

The Step Sequencer Principle

The basic idea behind analog step sequencers was to set up a progression of control voltages, and output these step by step. In early analog sequencers, three control voltages were usually created per step, in order to drive different parameters. The most common usage was control of a sound's pitch, amplitude, and timbre (cutoff) per step.

The control surface of analog sequencers often contained three rows of knobs or switches on top of (or beside) each other, each row with 16 steps. Each row had its own control voltage output, and the parameter it controlled was determined by the control input (on the synthesizer) it was connected to. A trigger pulse determined the step tempo. A running light (an LED) indicated the current step. This principle helped to create an electronic music style whose appeal stems from the mesmerizing effect that repeating patterns can have.

The running light programming concept also appeared in later drum computers. The most well-known representatives of this category being the very popular Roland TR series drum machines.

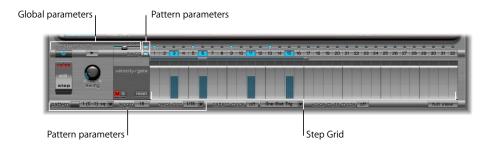
The introduction of the MIDI standard and increasing use of personal computers in music led to a rapid decline in the step sequencer and related technology. More modern concepts that didn't adhere to the step and pattern principle came into vogue.

Despite this, step sequencers haven't disappeared completely. Hardware groove boxes have experienced a renaissance over the last couple of years. Their intuitive nature has made them a favorite tool for rhythm programming.

Ultrabeat provides an integrated step sequencer of the newest design, that time-warps the advantages of its analog predecessors to the present day. As part of the "dynamic duo" with Logic Pro, it raises modern rhythm programming to a new level.

Step Sequencing With Ultrabeat

Ultrabeat's step sequencer contains 24 sequences—each consisting of up to 32 steps. The sequencer is divided into three sections.



- *Global parameters*: These parameters globally control the pattern and sounds, independent of the individual steps and patterns.
- Pattern parameters: Control the currently selected pattern.
- Step Grid: Here, the actual sequencing takes place. A pattern of 32 steps is shown for each sound. The pattern grid of the sound that is currently selected in the Assignment section is shown. You can add or remove events to the grid by simply clicking at the desired step position. Parameter values in the individual steps are altered by grabbing and moving them with the mouse.

Global Parameters

A description of the parameters that apply globally to all patterns follows.



On/Off Button

This button turns the step sequencer on or off.

Edit Mode Switch

This switch toggles between Voice and Step mode:

• *Voice mode (default):* In Voice mode, editing a drum sound's parameters sets the parameters of the drum sound itself.

• Step mode: In Step mode, you can automate a sound's parameter from one step to the next. Values are offset, instead of set—all of your original drum sound settings remain unaltered by any parameter changes performed in Step mode. Step automation changes only operate on parameters when the sequencer is running. These parameter changes occur individually per step. This means that if the sequencer is turned off, you'll hear the original sound. For more information see "Step Mode" on page 597.

Transport Button

The Transport button starts and stops the sequencer pattern. This allows you to inspect the step sequencer pattern while Logic Pro is stopped, for example. The step sequencer is always synchronized to the tempo of Logic Pro.

Note: If the Transport button is displayed in blue, the sequencer interprets incoming MIDI notes between C–1 and B0 as performance information. For more information see "Using MIDI to Control the Sequencer" on page 601.

Swing

This rotary knob globally determines the swing intensity for all sounds that have the Swing function activated (see "Swing Enable" on page 593).

Swing changes the distance between notes: notes on odd-numbered steps remain unchanged, while even-numbered notes are slightly shifted. At a 0 setting (the knob all the way to the left), the Swing function is not active. Turning the Swing knob up shifts the affected notes towards the following note.

Note: Swing is only active at grid resolutions of 1/8 and 1/16.

Pattern Parameters

A pattern has a maximum number of 32 steps and contains the total events of all 25 sounds. At the bottom edge of the Ultrabeat window you can select from one of 24 patterns, and set parameters for each pattern that apply to all sounds.

Pattern Menu

Allows you to choose one of the 24 patterns.

Length

This parameter defines the length of the pattern. The length of the grid can be adjusted by dragging the value in the Length parameter field or the bar beneath the swing buttons.



Resolution

This parameter determines the resolution of the pattern. It defines the metric unit of a measure that is represented by the individual steps.

As an example: The 1/8 setting means that each step of the grid represents an eighth note. Given a pattern length of 32 steps, the pattern would run for 4 measures (32 \div 8).

The Resolution setting applies to the entire grid, and therefore, equally to all sounds.

Note: The interplay between the Length and Resolution values allows the creation of different kinds of time signatures. Here are a few examples: The values Length = 14 and Resolution = 1/16 result in 7/8 time, Length = 12 and Resolution = 1/16 in 3/4 time, and Length = 12 and Resolution = 1/16 in 120 and Resolution = 1210 in 1210 time.

Accent

Individual steps can be strongly emphasized, or accentuated.



To turn on the Accent function, click the blue LED to the right of the Accent slider. The Accent slider globally determines the volume of the programmed accents.

In order to program an accent for a particular step, click on the blue LED above the desired step, and the step in question will be played louder.

Note: The Accent setting can be switched on or off individually per drum sound. This allows you to, for example, turn accent on for cymbals, but turn it off for the kick drum.

Swing Enable

Activation of the blue Swing button below the Accent button stipulates that the grid of the currently selected sound will be played in accordance with the Swing knob setting.

Only even-numbered steps are affected by the Swing parameter; exactly which beats this corresponds to depends on the selected Resolution parameter setting, as demonstrated by the following example.

At a Resolution of 1/8 and a Length of 8, the notes on steps 1, 3, 5, and 7 represent quarter notes in the measure. These remain unchanged. Only the eighth notes found between them (steps 2, 4, and so on) are shifted by the Swing function. The amount of shift is equal to the swing intensity (see "Swing" on page 592).

Note: Swing is only active for grid resolutions of 1/8 and 1/16.

Step Grid

In the step grid, the pattern is displayed in numerous rows and steps. The rows always correspond to the sound that is currently selected in the assignment area. Choosing a different sound switches the sequencer display to show the rows that correspond to the newly-selected sound.

The step grid area contains two rows—each with 32 fields.



- Trigger row: Click a button to activate or deactivate the sound on that corresponding beat.
- Velocity/Gate row: Here you set the length (gate time) and the velocity of the notes entered in the trigger row. Both parameters are displayed as a single graphical bar. The bar's height represents the velocity, its length depicts the note length.

The Trigger Row

In this row, consisting of the 1 to 32 buttons, trigger events are placed on corresponding steps. In other words: this is where you designate when (on which beat) the selected sound plays.



A click on one of the 1 to 32 buttons activates or deactivates the sound on that corresponding beat. In the example shown above, these are steps 1, 4, 8, 9, and 14.

Note: Dragging the mouse across the buttons allows corresponding triggers to be quickly turned on or off.

Trigger Shortcut Menu

Control-clicking (or right-clicking) one of the trigger buttons opens the Trigger menu, which offers the following commands:

- *Copy:* Copies all activated triggers to the Clipboard.
- Paste: Pastes all triggers from the Clipboard.
- Clear: Turns off all activated triggers.

- Add Every Downbeat: Adds triggers on every downbeat until the sequence is filled.
 The exact determination of which steps are downbeats depends on the grid resolution. For example, if the resolution is set to 1/16, Add Every Downbeat would create triggers on every 4th step. Starting with the initial downbeat at step 1, this would create trigger events on step 5, step 9, step 13, and so on. This command does not erase existing trigger events, it only adds trigger events.
- Add Every Upbeat: Adds triggers on every upbeat until the sequence is filled. The exact determination of which steps are upbeats depends on the grid resolution. For example, if the resolution is set to 1/16, Add Every Upbeat would create triggers on every 4th step. Starting with the initial upbeat at step 3, this would create trigger events on step 7, step 11, step 15, and so on. This command doesn't erase existing trigger events, it only adds trigger events.
- *Alter Existing Randomly:* Randomly reorders the sequencer steps while retaining the number of activated triggers.
- Reverse Existing: Reverses the order of existing sequencer steps.
- Shift Left by 1 Step: Shifts the sequencer data one step to the left.
- Shift Left by 1 Beat: Shifts the sequencer data one beat to the left. The precise number of steps that equals a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps; at a resolution of 1/8, a beat equals two steps, and so on.
- Shift Left by 1/2 Beat: Shifts the sequencer data one-half beat to the left. The precise number of steps that equals one half of a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps, so half of one beat is two steps; at a resolution of 1/8, a beat equals two steps, so half of one beat equals one step, and so on.
- Shift Right by 1 Step: Shifts the sequencer data one step to the right.
- Shift Right by 1 Beat: Shifts the sequencer data one beat to the right. The precise number of steps that equals a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps; at a resolution of 1/8, a beat equals two steps, and so on.
- Shift Right by 1/2 Beat: Shifts the sequencer data one-half beat to the left. The precise number of steps that equals one half of a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps, so half of one beat is two steps; at a resolution of 1/8, a beat equals two steps, so one half of one beat is one step, and so on.
- Create & Replace Randomly: Erases then randomly creates new sequencer triggers; in other words, a brand new sequence is created. How many events are created depends on the grid resolution.
- Create & Replace Few: Same as Create & Replace Randomly above, however it only creates a small number of trigger events. How many events are created depends on the grid resolution.

- Create & Replace Some: Same as Create & Replace Randomly above, however it creates less trigger events. How many events are created depends on the grid resolution.
- Create & Replace Many: Same as Create & Replace Randomly above, however it creates more trigger events, filling the sequencer with events.

Velocity/Gate Row

In this row, you set the length (gate time) and the velocity of the notes entered in the trigger row. Both parameters are displayed as a single graphical bar. The bar's height represents the velocity, its length depicts the note length.



Click-dragging on the desired bar allows you to change the length and velocity values for each step.

The gate time is divided into four equal sections, making it easy to set rhythmically accurate note lengths. In order for the one shot envelope to react to gate time, it is necessary to either; activate the Gate function in the sound itself (see "Trigger and Group Menus" on page 580) or use envelopes in sustain mode (see "Sustain" on page 588), in conjunction with rhythmically useful (short) decay times.

Reset

The Reset button located to the left of the velocity/gate row returns all velocity and gate values to their default settings (default velocity setting: 75 percent; default gate time: are all four sections are active).

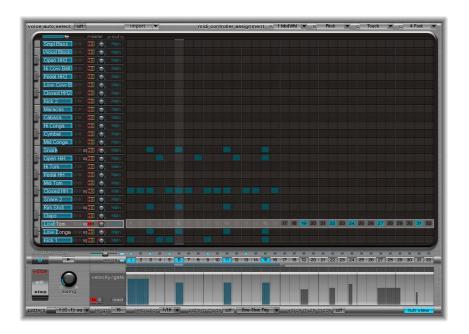
Velocity/Gate Shortcut Menu

Control-clicking (or right-clicking) steps in the velocity/gate row opens a shortcut menu which offers the following commands:

- Alter Vel(ocities): Randomly changes the velocity values of all steps while retaining the selected beats (the trigger row remains unchanged).
- Alter Gate (Time): Randomly changes the note lengths of all steps while retaining the selected beats (the trigger row remains unchanged).
- Randomize Vel(ocities): Same as Alter Velocities, but random parameter alteration is more pronounced.
- Randomize Gate (Time): Same as Alter Gate, but random parameter alteration is more pronounced.

Switching the Step Grid to Full View

Clicking the Full View button in Ultrabeat's lower right corner switches the synthesizer controls to a large grid filled with trigger buttons. The large grid displays the 32 trigger buttons for each of the 25 drum sounds simultaneously—independent of the currently selected sound.



As usual the selected sound is visible in the step sequencer area so that you can set the velocity and gate time for each step as well as offsets in Step mode. This means that the sequencer lane of the selected drum sound is represented in both the full view grid and the sequencer lane. Both the grid and the sequencer lane are active for the selected drum sound, so you can, for example, create trigger events quickly in the full view grid, then set accents in the sequencer lane.

Step Mode

Setting the Edit Mode switch to Step mode engages Ultrabeat's step automation feature. Step automation lets you program parameter changes per step for each drum sound. You can automate as many of the parameters available for automation per step as you want.

The sound parameters that can be edited per step include all functions of the Synthesizer section, except the menus (modulation routings and so on), buttons (oscillator type buttons, the buttons in the trigger/group section), and Pan/Spread parameters.

All parameters you automate will show up in the pop-up menu at the top of the Parameter offset row (see "The Parameter Offset Row" below).



When you engage Step mode, Ultrabeat's interface changes in the following ways:

- Yellow frames appear around all parameters in the Synthesizer section that are available for automation. Parameters not available for automation are still visible, but are disabled.
- The Velocity/Gate row in the step grid changes to show the Parameter offset row.

The Parameter Offset Row

In this row you can view and enter offset values per step for any parameter in the Synthesizer section framed in yellow. Editing parameters themselves is done, as before, using the controls in the Synthesizer section. In addition, it is however possible to edit offset values directly in the offset row.

Note: These have an effect only in relation to the actual parameter value. In other words: a value entered into the parameter offset row will be either added to the value set in the Synthesizer section (if the offset value is positive) or subtracted (if the offset value is negative).

Entering Offsets

By clicking in the parameter offset row you select the step into which you would like to enter a value. Every following parameter change made in the Synthesizer section will be recorded as an offset value for this step in the parameter offset row. To provide a better overview, the parameter offsets in the Synthesizer section are displayed by a yellow range of values. To enter an offset for a new parameter, click-drag on any yellow-outlined parameter in the Synthesizer section.



Once a parameter offset has been created for a given parameter on a given step, this will be represented in two ways. First, a yellow bar on the parameter will be drawn from the original parameter to the new parameter. In the Parameter offset row, the offset from the original parameter is represented as a yellow line starting from the zero point and going up for positive offsets or down for negative offsets.



The Parameter Menu of the Parameter Offset Row

All parameters that have been changed in Step mode are automatically added to the Parameter menu of the offset row. You can also select other parameters in the Parameter menu to display their recorded offset values:



You can change these values in two different ways:

- By simply drawing with the mouse you can change or add parameter offset values.
- By clicking on a step in the parameter offset row and grabbing and moving the corresponding control in the Synthesizer section, you can change the displayed parameter offset value.

Note: Moving a control element in the Synthesizer section whose value has not yet been changed in step edit mode adds an additional entry to the offset menu.

Parameter Offset Row Buttons

The Parameter offset row has three buttons: Mute, Solo, and Reset



The buttons functions are:

- *Mute*: Mutes the offsets of the selected parameter
- Solo: Solo the offsets of the selected parameter
- Reset: Returns all the offset values of the selected parameter to zero (no offset). A second click on the Reset button will remove the parameter from the Parameter menu.

Switching Between Step and Voice Mode

When creating offsets in Step mode, you may decide that you want to make a quick change in the original drum sound. Rather than requiring you to switch back and forth between edit modes for a simple change, you can press Option and Command to temporarily flip Ultrabeat back into Voice mode.

Copying and Reorganizing Patterns

You can reorganize the 24 patterns of a sound in the Pattern menu, using a copy and paste operation.

To copy a pattern using a shortcut menu:

- 1 Choose the desired pattern in the Pattern menu.
- 2 Control-click (or right-click) the Pattern menu, then choose Copy in the shortcut menu.
- 3 Choose the target pattern in the Pattern menu.
- 4 Control-click the Pattern menu, then choose Paste in the shortcut menu.

You can also use a key command short cut to copy patterns.

To copy a pattern using the key command short cut:

- 1 Select the desired pattern in the Pattern menu.
- 2 Press Option, open the Pattern menu and select another Ultrabeat pattern. This replaces the pattern in the target position.

All sequencer data that exists in the new pattern number is replaced. If you change your mind during the process, select the source pattern number.

To clear a pattern:

- 1 Select the desired pattern in the Pattern menu.
- 2 Control-click (or right-click) the Pattern menu, then choose Clear in the shortcut menu.

Exporting Patterns as MIDI Regions

Patterns programmed in Ultrabeat's internal step sequencer can be exported as MIDI regions into the Arrange area of Logic Pro.

To export an Ultrabeat pattern to the Arrange area:

- 1 Select the desired pattern in Ultrabeat's Pattern menu.
- 2 Click-hold the area to the left of the pattern menu.



3 Drag the pattern at the desired position on the respective Ultrabeat track.

A region is created, containing all MIDI events, including Swing and Accent parameter settings. Accents are interpreted as polyphonic pressure events.

Note: To avoid double triggering while playing back the exported MIDI region, you may want to turn off Ultrabeat's internal sequencer.

Note: Any step automation created in Step mode is also exported as part of the MIDI region.

Using MIDI to Control the Sequencer

Pattern performance can be influenced by incoming MIDI notes. This allows you to spontaneously interact with the step sequencer, making Ultrabeat an excellent live performance instrument. The manner in which Ultrabeat reacts to MIDI control is determined by the pattern, playback, and Voice Mute mode.



Pattern Mode

If activated, you can switch and start patterns via incoming MIDI note commands.

The Transport button turns blue to indicate that it is ready to receive incoming control commands.

MIDI notes C-1 to B0 switch between patterns: C-1 selects pattern 1, C#-1 pattern 2 and so on up to pattern 24, selected when MIDI note B0 is received.

Playback Mode

Pattern reactions to incoming MIDI notes is set in the Playback Mode menu. You will find the following options here:

- One Shot Trigger: The reception of a MIDI note starts the pattern, which plays once through its cycle, then stops. If the next note is received before the pattern has reached its final step, the new note stops playback of the first pattern and the next pattern begins playing immediately (this can be a different pattern or the same pattern, depending on the MIDI note received).
- Sustain: The reception of a MIDI note starts the pattern and it continues playing in an infinite loop until the corresponding MIDI note is released.
- Toggle: The reception of a MIDI note starts the pattern and it continues playing in an infinite loop until the next note is received. If it is the same note, the pattern stops immediately. If it is a different note, the sequencer immediately switches to the new pattern.

Note: In Toggle mode, you can confidently switch between patterns in the middle of a bar—the sequencer will stay in time and will automatically jump to the corresponding beat of the new pattern. This isn't the case in One Shot Trigger mode, however, because the moment you switch in this mode, the new pattern is started from the beginning.

• Toggle on Step 1: The behavior is as per Toggle mode except that the pattern change or stop occurs the next time beat 1 is reached—at the beginning of the next pattern cycle.

Voice Mute Mode

When you activate Voice Mute mode, playing a MIDI note starting at C1 and upwards mutes the corresponding sound in Ultrabeat's mixer. A subsequent MIDI note of the same pitch un-mutes it.

This is ideal for spontaneous arranging of pre-programmed patterns, and muting single elements of a pattern without deleting them. This is especially useful in a live performance—but not only there. Triggering the step sequencer via MIDI notes opens up a number of remixing possibilities. Ultimately, all of the creative pattern switching options discussed in this section is achieved through the use of MIDI note messages and can therefore be simply recorded, edited, arranged, and automated in Logic Pro.

Creating Drum Sounds in Ultrabeat

The following section covers a few specific sound creation tips. Please take the time to explore the vast and complex possibilities available to you in Ultrabeat, using the following programing tips as a starting point. You'll discover that there is hardly a category of electronic drum sound that Ultrabeat can't create easily.

Note: In Ultrabeat's Settings > Factory > Tutorial Settings folder, you will find a drum kit called Tutorial Kit. This drum kit contains all drum sounds discussed in this tutorial. It also includes a drum sound called Standard Tut(orial), which is a default set of neutral parameters that provide an excellent starting point for many of the following examples.

Creating Kick Drums

Electronically produced kick drum sounds are primarily based on the sound of a deeply tuned sine wave.

To program a kick drum in Ultrabeat:

- 1 Load the Standard Tutorial setting.
 - Note that Oscillator 1 is in Phase Oscillator mode.
- 2 Find a suitably tuned pitch in the lower octaves by soloing the bass drum along with other important tonal elements of the song (a bass or pad sound, for example). Use the Osc 1 Pitch slider to adjust the pitch until appropriate.
- 3 Use Env 4 to shape the volume of the bass drum.
 - For slower beats you'll want a longer decay phase, while at faster tempos you'll choose a shorter decay time. The attack time of Env 4 should be very short in any case (zero, in most cases) or the sound will lose its percussiveness, and its ability to be clearly heard in the mix.

The kick drum still sounds very soft and is somewhat reminiscent of the famous TR 808 bass drum. It's still missing a clearly defined attack.

To give the bass drum more kick by controlling the pitch with an envelope:

- 1 Ensure that Env 1 is chosen in the Mod menu of Oscillator 1's Pitch parameter.
- 2 Set the degree of modulation by moving the blue Mod slider approximately 3 to 4 octaves above the original pitch.



3 Set the attack time in Env 1 to zero by sliding the leftmost of the two junction points that sit on the x-axis all the way to the left.





- 4 Experiment with the decay time by moving the rightmost of the two junction points that sit on the x-axis; you'll discover that higher decay values (shifting the Bezier handle to the right) result in sounds similar to synth toms, while shorter decay values (shifting to the left) provide the kick character.
- 5 Change the Mod amount (the blue control) of Osc 1 Pitch again (see step 1).

 The interaction of this parameter with the envelope's decay time provides numerous possibilities for shaping the kick or punch of the bass drum sound.

Note: This simple bass drum sound is called Kick 1 in the Tutorial Kit, at a pitch of C1.

Removing Tonality

One advantage of bass drums based on sine waves is that their sound can be precisely tuned to match the song. The disadvantage: a recognizable pitch is not always desirable. Ultrabeat offers several methods to reduce the tonality of the sound. A very effective tool is the 2 Band EQ.

To reduce tonality using the 2 Band EQ:

- 1 For band 1, choose the Shelving mode at a frequency of about 80 Hz, a high Q value, and a negative Gain value.
- 2 For band 2, choose the Peak mode at a frequency of around 180 Hz, a medium Q value and also a negative Gain value.
 - On the EQ graph, you can see how the frequencies around 80 Hz are boosted, while the surrounding frequencies are reduced.



3 Vary the frequency of band 2 (easily recognizable in the blue part of the EQ graph) to influence the extent of bass drum tonality.

A further method for reducing the tonality of a drum sound that is rich with overtones is to use a lowpass filter. In the following example, you will control the cutoff frequency of the filter with an envelope.

To reduce tonality using a lowpass filter:

- 1 Reload the Standard Tutorial sound, choose A#0 as the basic pitch in Oscillator 1 and modulate it (as shown in the example page 603) using Env 1.
- 2 Increase the Saturation parameter value to enhance the overtones of the drum sound. Note, that the output of Osc 1 is directed to the filter, as the filter bypass button (the arrow between Osc 1 and the filter) is activated.
- 3 Make the settings shown in the following graphic to the Filter section:

Filter type: LP 24Cutoff value: 0.10

Mod Source for Cut: Env 3Mod Amount for Cut: 0.60

• Resonance: 0.30



- 4 Set the attack time of Env 3 to zero. Use the Decay time of Env 3 to shape the sound of the filtered bass drum.
- 5 You may also choose to control the filter resonance with an envelope. Make sure you dedicate a single envelope to this function (in this case, use Env 2 as a Mod source for Res). Choose a Mod amount for Res of about 0.80. Select a longer decay time in Env 2 than in Env 3 and listen carefully to the fatter and more atonal bass drum sound achieved through this Res modulation (due to the higher filter resonance).

Note: The bass drum described in the above example is called Kick 2 in the Tutorial Kit, at a pitch of C#1. It also features an interesting EQ setting (see paragraph below).

More Bass ...

Use the Kick 2 filtered bass drum sound as a starting point, and try out the remaining parameters in the phase oscillator. You will discover that high saturation values make the sound rounder and add more bass, for example. The character of the example is beginning to head in the direction of a TR-909.

More Kick ...

To get even closer to the 909, use an EQ setting as shown in the following graphic. Note the low frequency pressure point around 60 Hz (which can be seen in the red area on the EQ graph) as well as the assertive punch or kick (the blue area starting at 460 Hz and up) of a 909 bass drum are strengthened. (This EQ setting is already part of the Kick 2 setting.)



More Contour ...

In the example, all four envelopes are being used. Take some time to play with the shapes of the envelopes, while maintaining the attack and decay settings. Experiment with the junction points of the decay phase in the different envelopes to familiarize yourself with the sound shaping options available. Start with the decay phase of Env 4 (which controls the volume of Oscillator 1 as well as filter resonance) and observe how reshaping the belly of the envelope can change the character of the sound from crisp and short to round and voluminous.

The Ultrabeat Kick

You can use the extensive Ultrabeat features to create bass drums that are uniquely "Ultrabeat." Try modulating pitch with an LFO, rather than an envelope, for example.

To create an LFO modulated kick drum:

- 1 Start with the Standard Tutorial sound at a pitch of A#0 (Osc 1 Pitch), and choose LFO 1 as the Mod source in the Osc 1 Pitch section.
- 2 Set the degree of modulation by moving the blue Mod control to a value of A3.
- 3 Set LFO 1 to a low number of Cycles (25 to 35), a high Rate (starting with 70 Hz and higher) and a medium value for Decay (Ramp rotary knob at about –190).
- 4 Experiment with the LFO waveform and you'll discover that you can attain different nuances in the character of the bass drum attack.
- 5 Modulate the Asym (Asymmetry) parameter with the same LFO, and also vary the Slope and Saturation values.

This method enables you to create very different bass drum sounds with a single oscillator, one LFO, and one envelope (for volume). The character of the sounds can range from soft to punchy and the degree of tonality in the sound can be adjusted to taste.

Note: The bass drum sound described is called Kick 3 in the Tutorial Kit at a pitch of D1.

Use the second oscillator (with similar settings or with a sample) or use the filter and the ring modulator—the sky's the limit as far as your imagination is concerned, so get on with it, and create that next "gotta have it" drum sound.

Note: You can find an "emulation" of the legendary 808 bass drum under the name Kick 4 in the Tutorial Kit, at a pitch of D#1.

Creating Snare Drums

The sound of an acoustic snare drum primarily consists of two sound components: the sound of the drum itself and the buzzing of the snare springs. Try to approximate this combination in Ultrabeat with a single oscillator and the noise generator.

To create a basic snare drum:

- 1 Load the Standard Tutorial setting. Deactivate Oscillator 1, and switch Oscillator 2 on (in phase oscillator mode).
- 2 To get rid of the sine wave (which is not especially desirable for a snare sound, in contrast to the bass drum), modulate Osc 2 Pitch with a rapidly vibrating LFO with a medium Ramp Decay value. To accomplish this, choose LFO 1 in the Mod menu of Osc 2 Pitch. The pitch value for Osc 2 Pitch should be around G#2 and the Mod amount (the blue Mod control) should be about 3 to 4 octaves higher.
- 3 Set LFO 1 to a high Rate. Choose a value of 20 for Cycles and –20 for Ramp. The LFO Waveform parameter should be set to a value of about 0.58, which is a square wave.
- 4 Use Env 1 to control the volume of Oscillator 2 by setting Vol to the lowest possible value (–60 dB), choosing Env 1 in the Mod menu and adjusting the modulation intensity to a point just below its maximum value.

The screenshot shows the settings of Oscillator 2 and Env 1 described in steps 2 and 3.



- 5 Experiment with different Slope and Asym values to impart a more or less electronic character on the sound.
- 6 Turn on the noise generator and control its volume with the same quick envelope used in Osc 2 Volume.

7 Use the filter parameters of the noise generator to either roughen up, refine, or add bright frequencies to the noise component of the snare drum sound. Select a LP filter type, and try a filter frequency between 0.60 and 0.90. Modulate it with LFO 1 that you're already using to control the pitch of Oscillator 2.

Note: The snare drum sound is called Snare 1 in the Tutorial Kit, at a pitch of E1.

To refine the snare drum sound using FM synthesis:

- 1 Turn on Oscillator 1 in FM mode. Use Env 1 to control the volume of Osc 1 as well.
- 2 Choose a pitch for Oscillator 1 that's about an octave lower than Oscillator 2. Consciously avoid even intervals between the oscillators and detune them slightly from each other. As an example, try a pitch setting of F#2 in Osc 2 and E1 in Osc 1, then fine tune Osc 1 a few cents higher by holding Shift while adjusting its Pitch slider.
- 3 Experiment with FM Amount, and add more tone (low FM Amount value) or noise (more FM Amount) as desired. Also try modulating the FM Amount with a fast LFO.

Note: An exemplary snare drum sound that makes use of FM can be found in the Tutorial Kit at a pitch of F1. It is called Snare 2.

Higher FM Amount values lead to considerably more overtones and a very electronic sound character. If you want to make the sound more acoustic, feed Oscillator 1 (and possibly Oscillator 2 as well) into the main filter. Use these settings to start: LP 24 mode, a Cutoff value of about 0.60.

The 808 Snare

The famous 808 snare is based on two resonating filters and a noise generator, fed through a highpass filter. The mix ratio of the two filters and the volume of the noise generator can be adjusted. This structure cannot be 100% replicated in Ultrabeat.

To clone the 808 snare sound:

- 1 Load the Standard Tutorial setting.
- 2 Replicate the resonating filters of the 808 snare by two cleverly programmed phase oscillators:
 - Assign slightly different Slope values to them, and detune them by almost an octave.
 - The tonal relationship between the oscillators should also be uneven here—from E3 to F2, for example.
- 3 Control the volume of each oscillator with a different envelope. The envelope for the lower tuned oscillator should have a longer decay time than the very snappy envelope setting for the higher oscillator.
- 4 Feed the output of both oscillators into Ultrabeat's main filter, and hollow out the sound with a highpass filter. Activate the filter bypass button in both oscillators. Choose the HP 12 setting in the filter, a Cutoff value around 0.40 and a Resonance value of about 0.70.

You have just very cleverly emulated both of the 808's resonating filters. Shifting the pitch of both oscillators simulates the behavior of the 808's Tone control by the way.

To complete the 808 emulation by adding some noise:

- 1 Switch the noise generator on, and activate the highpass mode in its filter (HP). Set the Cutoff value to about 0.65, Resonance to 0.35 and add a little Dirt (around 0.06).
- 2 The noise generator provides the sustained snare sound. It should be shaped by its own envelope, independent of the decay phases of both oscillators, in order to get 808-like results. Changing the volume of the noise generator simulates the snap parameter of the 808.

Note: The 808 snare drum described is called Snare 3-808 in the Tutorial Kit, at a pitch of F#1. It also features an interesting EQ setting.

Dynamics Through Velocity

Use the 808 snare drum sounds in the Tutorial Kit to explore the possibilities Ultrabeat offers for implementing velocity.

To implement velocity:

- 1 Select the sound Snare 3-808.
- 2 Click on the word Off below the Volume knob in Oscillator 1.



- 3 In the Via menu that appears, choose Vel.
 A slider appears on the ring around the knob.
- 4 Move the slider in a clockwise direction. When you grab the slider, a help tag displays the value. Set it to –0 dB.



5 Repeat steps 1 to 4 in both Oscillator 2 and the noise generator.

You can now dynamically play the sound using velocity.

To increase the performance dynamics:

1 Reduce the values of the individual volumes by turning down the Volume knobs in both oscillators and the noise generator. Note how the Mod ring and its Via sliders also move back. Change the Via slider positions until all three Volume knobs look like this:



If you use differing intensities for each Volume knob when completing step1, you'll have the potential of individual velocity reactions for each sound component.

2 Increase the dynamics of the sound as a whole by assigning the following setting to the Voice Volume knob:



You now have an 808 snare that is exceptionally responsive to velocity. As you may know, this wasn't possible with the original—not even an 808 sample could offer the dynamic volume control of individual sound components that's demonstrated here. A sample only offers you the whole sound, not its constituent parts.

In the next step, you'll use velocity to control the character of the sound—individually for each component—plus volume, of course:

3 In the Saturation Mod menu of Oscillator 2, choose Max and then Velocity in the corresponding Via menu.





4 Set the additional control that appears as shown below, to control the character of the sound with velocity:



5 Repeat this with the other parameters of Oscillator 2, as well as pitch:



- 6 Modulate the noise generator as follows:
 - Cut parameter: Choose Max as modulation source, then set the modulation control as shown below.
 - *Dirt parameter:* Choose LFO 2 as modulation source, then set the modulation control as shown below.



The sound is now nothing like an 808 snare—and that's exactly what you wanted to achieve. Keep experimenting with velocity and figure out when it makes sense to use it as a direct or indirect modulation source, in either its positive or negative form.

Note: The dynamic 808 snare drum described above is called Snare 4—Vel in the Tutorial Kit, at a pitch of G1.

The Kraftwerk Snare

A further classic electronic snare drum sound is the highly resonant lowpass filter of an analog synthesizer that quickly closes with a snap. This sound was used extensively by Kraftwerk.

To recreate the Kraftwerk snare sound with Ultrabeat:

- 1 Select the Snare 1 sound.
- 2 Direct the signals of both oscillators and the noise generator to the main filter.
- 3 Modulate Cutoff with Env 1 (this is already modulating the volume of the noise generator).
- 4 Modulate the filter resonance with Env 2.
- 5 Experiment with the parameters described in steps 1 to 5 (especially the envelopes), introduce EQ into the sound, and discover how much "playing room" these basic settings allow you.

Note: An exemplary sound called Snare 5—KW is in the tutorial set, at a pitch of G#1. Analyze, and compare it to your own creation.

Creating Toms and Tonal Percussion

Tonal percussion sounds such as toms or congas are relatively easy to emulate electronically with sine or triangular wave oscillators. Ultrabeat's phase oscillator offers you a broad spectrum of suitable basic sounds with which to start. Control the pitch of the oscillators with envelopes, and use the programming techniques discussed in the bass and snare drum sections to alter tonality. You should find it easy to create a broad range of toms and similar sounds.

Note: At the pitches A1 to B0 in the Tutorial Kit you'll find typical 808 toms. Analyze these sounds and modify them as you see fit.

At this point, spend some time experimenting with the Model mode of Oscillator 2. Try to familiarize yourself with the effect of each parameter, and create some of your own tonal percussion sounds, ranging from small tabla drums to glass bowls.

Note: The Tabla and Glass sounds (at pitches C2 and C#2) of the tutorial set combine both Osc 2 Model and FM. They are also good examples of the complex use of velocity as a modulation source.

Creating Hi-Hats and Cymbals

Electronic hi-hat sounds are very easy to create in Ultrabeat.

To create a hi-hat in Ultrabeat:

- 1 Load the Standard Tutorial sound.
- 2 Switch off Oscillator 1 and turn on the noise generator.
- 3 Choose the following settings for the noise generator:



In the screenshot above, you can see, that the Cutoff parameter is modulated by Env 1. The modulation is negative, the position of the Mod slider is below that of the base parameter value.

- 4 Use rather short decay values for Env 1 and Env 4.
- 5 Set the attack time of Env 4 to a value of 0. The attack time of Env 1 should also be rather short, but not equal to zero.

Note: You'll find a similarly constructed sound called HiHat 1 at a pitch of F2 in the tutorial set. Also analyze the hi-hat sound Hihat 2 at a pitch F#2.

It's not far from the hi-hat to the crash cymbal: the main difference between a hihat and crash cymbal sound is the length of the decay time. Correct assignment of the envelopes is the key to producing different cymbal sounds.

Select the Cym 1 and Cym 2 sounds in the Tutorial Kit and try different envelope assignments and settings for Cutoff and Volume in the noise generator, Cutoff and Volume in the main filter, and so on.

Metallic Sounds

If you want to create metallic sounds with Ultrabeat, the ring modulator and the Model oscillator are the ideal tools.

To use the ring modulator:

- 1 Load the Standard Tutorial sound.
- 2 Activate a phase oscillator and the Model oscillator. Choose a pitch for each oscillator above C3 so that a slightly detuned interval is created.

3 In the Material Pad of the Model oscillator, choose a setting with plenty of overtones as in the graphic below.



4 Set the volume of each oscillator to a value of –60 dB and turn the ring modulator on by clicking on its name.

You've just created a bell-like sound that you can filter (with a high resonance value) if required.

Note: You can find a similar sound called Ring Bell at a pitch of A2 in the Tutorial Kit.

Creating Clicks and Cuts

Ultrabeat features extremely fast envelopes and uncommonly powerful LFOs. Use these modulation sources to perform extreme modulations of the oscillator and filter parameters. The key to creating "out of the ordinary" sounds is to try modulating as many targets as possible, and not to be afraid of using extreme settings: use a quick envelope to drive the filter to self oscillation for a fraction of a second, use a few LFO cycles at a much higher rate than other cycles, or experiment with the Dirt parameter or the bit crusher.

Programming in Building Blocks

As you become familiar with drum sound programming, you may begin thinking in building blocks, realizing that drum sounds usually consist of different components. Once you've mentally, or physically, written down your list of components, you should try to emulate each *component* that contributes to the sound's character—making use of the different sound generators available in Ultrabeat. Assigning dedicated (amplitude) envelopes to the different components allows you to control their temporal behavior individually. As an example: You can emulate the body of a drum with Oscillator 1, the sound of the stick hitting the skin (or first transient) with the noise generator, additional overtones and harmonics can be provided by Oscillator 2 or the ring modulator. Once you begin thinking that drum sounds consist of several building blocks or layers, the design of the Volume controls in the individual sound generators might make more sense to you, as this is the place where the blocks are combined, balanced, and controlled.

GarageBand Instruments are automatically installed with Logic Pro. You can insert them as per other software 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 Pro a trouble-free experience.

GarageBand Instruments are actually less CPU and memory-intensive versions of equivalent Logic Pro instrument plug-ins. In the case of synthesizer sounds, the ES2 is the "big brother" of the GarageBand Instruments. In the case of organ sounds, the EVB3, and in the case of electric piano sounds, it's the EVP88 and so on with Clavinet (EVD6) or other sounds (EXS24 mkll).

The interface of GarageBand Instruments consists of a simple brushed metal panel that contains a number of parameter sliders and associated value fields. As an example, the Digital Stepper instrument is shown below:



Many of the GarageBand parameters are macros, which address specific, useful parameters in the EXS24 mkll, ES1 (or other equivalent Logic Studio instrument).

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This has two main benefits:

- As the GarageBand instrument plug-ins are less CPU and memory-intensive, they load faster than the equivalent Logic Studio software instruments.
- Limitation to a few, powerful, parameters simplifies the use of the instruments. Play around with the parameters to see how easily spectacular sounds can be created!

The macro parameter sliders available to each GarageBand Instrument are different. This is because the equivalent Logic Studio Instrument parameters being addressed 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!

GarageBand Instrument Parameters

Many of the GarageBand Instruments share the same parameters, which function identically in each. The following covers these universal parameters, and the individually-named sections provide information and cover parameters or facilities that are unique to that specific GarageBand instrument. Note that not all GarageBand instruments contain all of the macro parameters described below.

Universal GarageBand Instrument Parameters

- Volume: Sets the overall level of the instrument.
- Tuning: Sets the overall pitch of the instrument.
- Mix: Mixes two tones together by modulating them.
- Cutoff: Allows more (high) or less (low) of the sound through, making it brighter or darker.
- Resonance: Determines where in the frequency range a sharp peak is.
- Attack: Makes the sound start more slowly or quickly. A fast setting will make the sound strike like a piano key, whereas a slow setting will make the sound bow like a violin string.
- Decay: Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values.
- Sustain: Determines how long a note will continue at its peak volume.
- Release: Determines how fast notes will fade out after you have let go of the keys on your keyboard.

Analog Basic

The Analog Basic sound is based on the ES2. This is a simple analog synthesizer tone that is useful for a range of musical styles.

Analog Mono

This is a monophonic (one note can be played at a time) analog synthesizer lead sound. Unique macro parameters are:

- Glide: Determines the time it takes a note to change (slide) to another.
- *Richness*: Determines the complexity of the sound texture, making the sound fuller.

Analog Pad

The Analog Pad is based on the ES2. This is a warm analog synthesizer pad that is useful for a range of musical styles. Unique parameters are:

- Modulation: Makes the sweeping movement of the pad faster or slower.
- Character: Determines if the sound is soft or sharp.
- *Cutoff Envelope*: Determines where in the frequency range the cutoff of the envelope is positioned.
- *Duration:* Determines the duration of the sweeping movement.
- Animation: Determines how much animation is added to the pad by the envelope.

Analog Swirl

The Analog Swirl instrument is based on the ES2. It is a warm analog synthesizer pad that features a chorused, swirling sound. It offers one unique parameter:

• Modulation: Makes the sweeping movement of the pad faster or slower.

Analog Sync

The Analog Sync instrument is based on the ES2. It emulates tones from analog synthesizers that would synchronize two oscillators to produce their sound. The Analog Sync instrument is most useful for hard-edged analog synthesizer lead sounds. Unique parameters are:

- *Sync:* Determines the synchronization (or lack of it) between the two oscillators, and therefore the harshness of the sound.
- *Sync Modulation:* Determines how much the synchronization of the two oscillators is modulated, resulting in more complex (and harder) tones.
- *Sync Envelope:* Determines the amount that envelope parameters impact on the sound.

Bass

The Bass sound is sample-based. It offers various bass sounds, accessible from the Settings menu, and can be used for a variety of musical styles. Included are Fingerstyle, Fretless, Muted, and Slapped Electric basses, and an Upright Jazz bass.

Digital Basic

The Digital Basic instrument is based on the ES2. This is a basic digital synthesizer sound that is useful for a range of musical styles. Unique parameters are:

- *Harmonics:* Changes the sound dramatically as *more* harmonics (overtones) are added. The impact of this parameter is difficult to describe, so please experiment with it.
- *Timbre*: Changes the color of the sound from dark to bright.

Digital Mono

The Digital Mono instrument is based on the ES2. This is a monophonic digital synthesizer lead sound. Unique parameters are:

- Harmonics: Makes the sound a little thicker (more) or thinner (less).
- *Timbre*: Changes the color of the sound from dark to bright.
- Timbre Envelope: Changes the color of the sound, dependent on how firmly you strike your keyboard. If low is selected, the color of the sound won't change much, no matter how hard you play the keys. If a high level is selected, the sound will change dynamically, in response to firmer or softer keyboard playing.
- *Richness:* Subtly detunes each played note from one another, making the sound a little thicker as you move towards *high* values.
- *Distortion:* Distorts the overall sound, making it quite nasty and aggressive. Watch out, though, this makes the sound much louder!

Digital Stepper

The Digital Stepper sound is based on the ES2. This is a digital synthesizer sound that steps through a number of tones, creating a rhythmic pattern. Unique parameters are:

- *Balance:* Sets the balance between a harder and more spiky sound (digital) and a warmer, softer sound (analog).
- *Modulation*: Makes the tonal steps faster (high) or slower (low).
- Harmonics: Makes the sound a little thicker (more) or thinner (less).
- Harmonic Steps: Makes the tonal steps more (large) or less (small) noticeable.
- *Cutoff Steps:* Sets the amount of cutoff applied to each step. A higher value (large) will make the cutoff effect more pronounced.
- Duration: Sets the length of the steps.

Drum Kits

The Drum Kits sounds are sample-based. The Settings menu offers various kits, including Dance, Hip-Hop, Jazz, Pop, Rock, and Techno.

Electric Clavinet

The Electric Clavinet sound is based on the EVD6. It emulates the Hohner D6 clavinet. It offers the following unique parameter:

• *Damper:* Changes the tone of the clavinet, making it less sustained, and more woody sounding as you move towards the *high* setting.

Electric Piano

The Electric Piano sound is based on the EVP88. It sounds like the Fender Rhodes electric piano. Macro parameters are:

- Model: A more bell like tone is achieved when the Tines button is selected.
- *Decay*: A short value will make the sound almost plucked, whereas a long setting will make the sound sustain while the keys are held.

Guitar

The Guitar sound is sample-based. It offers Classical Acoustic, Clean Electric, and Steel String guitar sounds, which are accessible from the Settings menu.

Horns

The Horns sound is sample-based. It emulates a Horn Section, Pop Horn Section, and a Trumpet Section, which can be accessed via the Settings menu.

Hybrid Basic and Hybrid Morph

The Hybrid Basic and Hybrid Morph are sample-based synthesizers that can create spectacular sounds. Unique parameters are:

- *Waveform:* Choose the sample set used to generate the basic synthesizer sound. Each wave in the Hybrid Morph synthesizer is based on two sample layers.
- Morph: Controls crossfades between the two sample layers.
- Morph Envelope: Allows you to control the morph over time. As an example, setting the Morph parameter to B and the Morph Envelope to From A To B, morphs the Wave from A to B in accordance with the ADSR envelope settings.
- *Cutoff Type*: This menu contains a number of preset filter curves. Try them out, and experiment with the Cutoff and Resonance parameters.

Note: It you set the Morph parameter to A and the Morph Envelope to From A to B, certain ADSR settings will result in no sound. In this context, you can achieve interesting results by using the modulation wheel to offset the Morph parameter during live performances.

In the Hybrid Basic synthesizer, you can use the Wheel to Vibrato and Wheel to Cutoff sliders to determine the parameters controlled by the modulation wheel.

Piano, Sound Effects, and Strings

The Piano, Sound Effects, and Strings sounds are sample-based. As with other GarageBand instruments, the Settings menu offers several variations.

Tonewheel Organ

The Tonewheel Organ sound is based on the EVB3. It emulates the Hammond B3 organ. Please try out the various Settings available, as this instrument is capable of generating a wide array of organ tones. Unique parameters are:

- Drawbars: Makes the sound a little thicker (more) or thinner (less).
- *Percussion Level*: Adds a 2nd or 3rd harmonic to the sound, changing both the color and timbre of the instrument.
- *Percussion Time:* Sustains the 2nd or 3rd harmonic when set to a long value. If a short value is selected, the harmonics will only be heard during the initial keystrike.
- *Click*: Introduces a click sound to the keystrike. Select a high level if you'd like this to be clearly heard.
- *Distortion:* Makes the sound rough, dirty, and noisy. This is great if you want to do Deep Purple covers!
- Rotary Speaker: These switches select from three different speaker effects. Chorale
 makes the sound swirl. Brake makes the sound swirl initially, and then slow down.
 Tremolo makes the sound wobble.

Tuned Percussion

The Tuned Percussion sound is sample-based. It emulates a vibraphone.

Voice

The Voice sound is sample-based. It emulates a mixed choir.

Woodwind

The Woodwind sound is sample-based. It offers you the sound of woodwind instruments, such as flutes and saxophones.

Synthesizer Basics

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, have a 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 Pro'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 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?

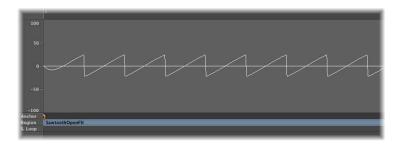
Synthesis in this context is the (re)production of a sound which emulates, or synthesizes the sound of another instrument, a voice, helicopter, car, dog bark—in fact, any sound you can think of!

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

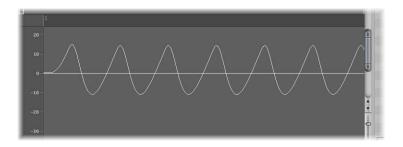


This picture shows an overview of a sawtooth wave (a = 220 Hz); The filter is open, with cutoff set to its maximum, and with no resonance applied. The screenshot shows the output signal of Logic Pro's ES1, routed to a monophonic Logic Pro output channel. The recording was performed with the Bounce function of this channel, and is displayed in the Sample Editor at a high zoom setting.

▶ Tip: Classic synthesizer literature indicates the use of the sawtooth wave to create a sound similar to that of a violin. The rich and full sound of the sawtooth wave is the most popular synthesizer waveform, and serves as a basis for synthetic string and brass sounds. It is also handy for synthesized bass sounds.

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.

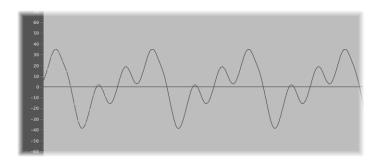


The wave length here is not really higher, but the zoom setting is.

Fourier Theorem and Harmonics

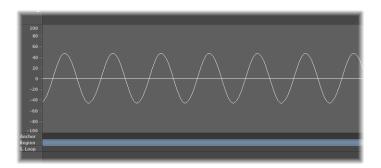
"Every periodic wave can be seen as the sum of sine waves with certain wave lengths and amplitudes, the wave lengths of which have harmonic relations (ratios of small numbers)." This is known as the Fourier theorem. Roughly translated into more musical terms, this means that any tone with a certain pitch 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 an ES1 sawtooth wave with a high resonance setting, and the cutoff frequency set to around 60%. 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:



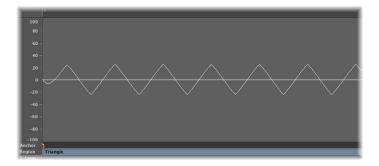
The effect of the resonating filter is comparable to a graphic equalizer with all faders higher than 660 Hz pulled all the way down, but with only 660 Hz (*Cutoff Frequency*) pushed to its maximum position (*resonance*). The faders for frequencies below 660 Hz remain in the middle (0 dB).

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.



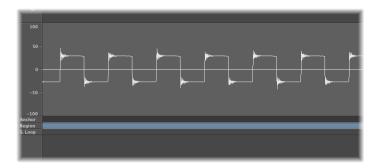
Other Oscillator Waveforms

Waveforms (waves) are named sawtooth, square, pulse, or triangular because of their shape when displayed as an oscillogram (as in the Sample Editor of Logic Pro). 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.

Classic synthesizer literature encourages the use of the triangular wave for the creation of flute-like sounds. In the age of sampling, however, it's pretty hard to sell a triangular wave as a flute sound to anyone.

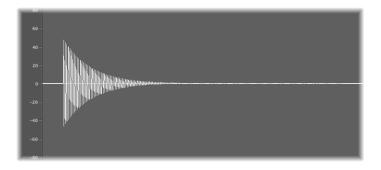


The screenshot above shows a rectangular wave. The rectangular wave contains all odd harmonics, the amplitudes of which decrease proportionately with their number. The pulse width can be set to any value and serves as a modulation address.

Classic synthesizer literature likens the rectangular wave to the sound of a clarinet, as the clarinet doesn't feature any even harmonics in a certain frequency range either.

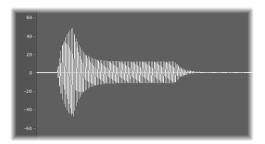
Envelopes

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 its maximum level in the *attack time*, falls from this maximum level to the *sustain level* in the *decay time*, and maintains the *sustain level* as long as you hold the key. When the key is released, the envelope falls from its *sustain level* to zero over the *release time*. The brass or string-like envelope of the following sound—the envelope itself is not shown in this graphic—has longer *attack* and *release times*, and a higher *sustain level*.



The envelope generator can also control the rise and fall of the cutoff frequency. You can also use envelope generators to modulate other parameters. In this context, modulation can be thought of as a remote control for a given parameter. There are more sources that can serve as a modulation source: the pitch (note number), velocity sensitivity or the modulation wheel, for example.

Glossary

AAC Abbreviation for Advanced Audio Codec. A compression and decompression algorithm and file format for audio data.

AAF Abbreviation for Advanced Authoring Format. A cross-platform project exchange file format that you can use to import multiple audio tracks, inclusive of references to tracks, time positions, and volume automation.

accelerando A gradual increase in tempo (see tempo).

AD converter or ADC Short for analog to digital converter; a device that converts an analog signal to a digital signal.

ADAT Abbreviation for Alesis Digital Audio Tape. The ADAT is an eight track digital multitrack cassette recorder that uses an S-VHS video tape to record audio at 16 or 20 bit depth.

ADAT optical Optical interface for parallel transmission of eight audio channels via fiber-optic cable. It is a well-established standard for digital multi-channel interfaces.

AES/EBU Short for Audio Engineering Society/European Broadcasting Union. This association has a standard transmission format for professional stereo digital audio signals called AES/EBU. The format is similar to S/P-DIF, but uses balanced line drivers at a higher voltage. Depending on the type of devices involved, AES/EBU and S/P-DIF coaxial interfaces can communicate directly.

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 key pressure or pressure sensitivity.

AIFF Abbreviation for Audio Interchange File Format. A cross-platform file format supported by a large number of digital audio and video editing applications. AIFF audio can use a variety of bit depths, most commonly 16 and 24 bit.

AKAI Common sample data format that the EXS24 mkll is compatible with.

ALAC Abbreviation for Apple Lossless Audio Codec, an encoding/decoding algorithm that delivers lossless audio compression.

alias A pointer to a MIDI region in the Arrange area. 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 artifact 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 increases 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 volume of the sound, measured in decibels (dB).

amplitude peak The loudest point of an audio signal.

analog signal A description of data that consists of an endlessly varying voltage level, that represents audio information. Analog signals must be digitized, or captured, for use in a Logic Studio application. Compare with digital.

anchor point The start point of the audio file that an audio region is based on. Also see Sample Editor.

Apple Loops An audio file format, commonly used for recurring rhythmic musical elements or elements suitable for repetition. Apple Loops contain tags and transients that are used by Logic Pro for time stretching and pitch shifting tasks. These tags also allow you to quickly locate files by instrument, genre, or mood in the Loop Browser.

Apple Loops Utility The Apple Loops Utility allows you to create your own Apple Loops. To use the Apple Loops Utility, select an audio region in the Arrange and choose Audio > Open in Apple Loops Utility.

ASCII Acronym for American Standard Code for Information Interchange. Standard computer character set, allowing computers to deal with text characters. When you type ASCII characters from the keyboard, the computer interprets them as binary so they can be read, manipulated, stored, and retrieved. Also see scan code.

attack Start phase of a sonic event. Also part of an envelope (see envelope).

attenuate The act of lowering the level of an audio signal (see boosting and cutting).

audio file Any digital recording of sound, stored on your hard drive. You can store audio files in the AIFF, WAV, Sound Designer II (SDII), and CAF formats in Logic Pro. All recorded and bounced WAV files are in Broadcast Wave format.

audio interface Device used to get sound into and out of your computer. An audio interface converts digital audio data, sent from your computer, into analog signals that speakers can broadcast. In the other direction, an audio interface converts analog signals (such as a vocal performance) into digital audio data that can be understood by your computer.

Audio MIDI Setup (AMS) The Audio MIDI Setup (AMS) utility is used to configure the audio and MIDI input and output devices connected to your computer.

Audio Units (AU) Audio Units is the standard Mac OS X format for real time plug-ins. It can be used for audio effects, software instruments, and Generators. The Audio Unit format is incorporated into the operating system, and installed Audio Unit plug-ins can be simultaneously accessed by all applicable programs.

automation Automation is the ability to record, edit, and play back the movements of all knobs, controls, and buttons, including: volume faders, pan, EQ, and aux send controls, plus most effect and instrument plug-in parameters.

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 a lot 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.

bass A musical instrument. The term also refers to low frequency sounds or components within a sound. See frequency.

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.

Bezier curve A curve created from a line that contains two points. These points affect the line, allowing it to be pulled into a curve. In a computer application, Bezier curves are created by moving handles at these two points, thus adjusting the curve shape. Named after Pierre Bezier, who discovered the mathematical formula for these curves.

binaural hearing A description of the way human beings process audio positioning information, allowing the direction of a signal source to be recognized (in front, behind, above, below, and to the left or right of the listening position).

binaural panning A process that emulates binaural hearing.

bit depth The number of bits used by a digital recording or digital device. The number of bits in each sample determines the (theoretical) maximum dynamic range of the audio data, regardless of sample rate.

bit rate Bit rate, when talking about MP3 files, refers to the transfer bit rate at which the files are encoded. Conversationally, the term is more often used to describe the relative quality of the file, with lower bit rates resulting in less defined audio.

bit resolution Alternative term for bit depth. See bit depth and sample rate entries.

blue noise Highpass-filtered white noise, which sounds like tape hiss.

boosting The act of raising an audio level (see cutting).

bounce To process MIDI or audio regions with any applied effects, such as delay or compression, and combining them into one audio file.

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).

Broadcast Wave See Wave.

bus Busses are used to send audio to aux channel strips for processing or submixing tasks.

bypass To deactivate a plug-in. Bypassed plug-ins do not drain system resources. In

CAF Abbreviation for Core Audio Format. This file format can be used as a container for compressed or uncompressed audio files of (almost) any size, sample rate, or bit depth. The CAF file format can handle audio recordings of around 3 hours in length (at a 44.1 kHz sample rate—shorter at higher sample rates).

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.

CD Audio Short for Compact Disc—Audio; a standard for stereo music CDs: 44.1 kHz sampling rate and 16 bit depth.

cent A tuning division of a semitone. There are one hundred cents in a semitone. Many of the Logic Studio software instruments contain a Fine parameter that allows sounds to be tuned in cent steps.

checkbox A small box. You click a checkbox to select or deselect (or turn on/off) 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 data, using the Edit menu. Data stored in the Clipboard can be pasted to different positions.

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.

clock Electrical synchronization impulse, transmitted every 1/96 note. Was used in older drum machines before the advent of MIDI (MIDI Clock is a modern implementation of simple clock signals. It runs at 24 ppqn—pulses per quarter note—or 96 pulses per note).

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 (also see expander).

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 being sent or received is encoded in the second data byte.

Controls view All Logic Pro 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 item in the plug-in header's View 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.

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.

count-in Beats heard prior to the start of a recording (or playback).

cueing Monitoring (hearing playback) while fast-forwarding or rewinding.

cutoff frequency Frequency at which the audio signal passing through a low or highpass filter is attenuated by 3 dB.

cutting The act of reducing a level, or frequency, when using EQ or other filters. Also used to describe physically dividing and removing sections of files, regions, and so on (see boosting and attenuation).

DA converter or DAC Short for digital/analog converter; a device that changes an analog signal into a digital signal.

data bytes These define the content of a MIDI message. The first data byte represents the note, or controller number; and the second the velocity, or controller value.

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.

DC offset An error that can result in direct current (DC) being layered over the audio signal, resulting in a vertical shift in the waveform position shown in the Sample Editor.

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 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.

DFS An abbreviation for Digital Full Scale. Sometimes expressed as 0 dB DFS. This is the maximum theoretical level that a digital signal can reach before clipping and other types of distortion may be introduced.

dialog A window containing a query or message. You must interact with it (by pressing a button) before you can 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 by electronic or electromagnetic signals. All files used in the Logic Studio applications are digital. Also see analog for comparison.

Digital Full Scale See DFS.

disclosure triangle A small triangle you click to show or hide details in the user interface.

distortion The effect that occurs 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 computer applications. If the proper driver is not correctly installed for your audio hardware, your computer may not recognize or work properly with it. See Devices tab.

DSP (digital signal processing) The mathematical processing of digital information to modify a signal. An example is the Insert slot of Logic Pro channel strips, which assigns DSP effects such as dynamic compression and delay to a channel signal. Even simple operations such as changing volume and pan are DSP calculations.

dynamics Refers to changes in volume, or other aspects of a piece of music, over time.

dynamic range The dynamic range is the difference in level between the highest signal peak that can be reproduced by an audio system (or device in the system) and the amplitude of the highest spectral component of the noise floor. Put another way, 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 view Almost all Logic Pro plug-ins (and Audio Units) offer a graphical view of effect and instrument parameters. The Editor view is the default, but can be accessed via the Editor item in the View menu at the top of each plug-in window, should the Controls view be visible.

effect A type of software algorithm that alters the sound of an audio signal in a variety of ways. Logic Studio includes a set of EQ, dynamics, time-based, modulation, and distortion effects.

envelope The envelope graphically represents the variation that a sound exhibits over time. An envelope, used as a controlling device, basically determines how a sound starts, continues, and ends. Synthesizer envelopes usually consist of attack, decay, sustain, and release phases.

EQ Shortened form of equalizer. Equalizers are used to boost or cut frequencies in an audio signal. There are several types of EQ available in the Logic Studio applications.

equalization See EQ.

event Individual MIDI command, such as a note on command. Continuous controller movements (modulation wheel, for example) produce a quick succession of individual events—each with an absolute value.

expander An effect process that increases the dynamic range of an audio signal. It is the antithesis of the compression effect (see compressor).

export To create a version of a file, such as a Logic Pro project, in a different format that can be distributed and used by other applications.

filter effect Filters are effects designed to reduce the energy of a specific frequency within a signal. The names of the individual filters illustrate their function. As an example: A lowpass filter allows frequencies that are lower than the cutoff frequency to pass (see cutoff frequency).

filter slope The filter slope is the steepness, or severity, of filter attenuation (level reduction). As examples, a filter slope of 6 dB per octave would sound much softer than a filter slope of 12 dB per octave.

FireWire Apple trademarked name for the IEEE 1394 standard. A fast and versatile serial interface, often used to connect audio interfaces, and audio processing units to computers. FireWire is well suited to applications that move large amounts of data, and can be used to connect hard disks, scanners, and other kinds of computer peripherals. There are two versions of FireWire: FireWire 400 and FireWire 800. The latter is a faster variant, which uses a different connector type. Use of FireWire 400 devices on a FireWire 800 port is possible with suitable cabling, but it effectively halves the bandwidth to all devices on the port (also see M-LAN).

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 See window type.

frame Unit of time. A second in the SMPTE standard is divided into frames that correspond to a single still image in a video file or on video tape.

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.

frequency The number of times a sound signal vibrates each second, measured in cycles per second, or Hertz (Hz).

GM Abbreviation for General MIDI. A standard for MIDI sound modules that specifies a uniform set of instrument sounds on the 128 program numbers, a standardized key assignment for drum and percussion sounds on MIDI channel 10, 16-part multi-timbral performance and at least 24 voice polyphony. The GM specification is designed to ensure compatibility between MIDI devices. A musical sequence generated by a GM instrument should play correctly on any other GM synthesizer or sound module.

grab (an object) Positioning the mouse cursor over an object, then pressing and holding the mouse button down.

GS Extended GM standard developed by Roland Inc.

headroom An alternate term for dynamic range. See dynamic range.

help tag A small text window which appears when the mouse cursor is placed over an interface element. It indicates the name or value of the element. When editing operations such as moving or cutting a region are performed, a larger help tag will display the current (and starting) position of the region or function—in real time.

hierarchical menu Structured menus that open cascading sub-menus when an individual entry is chosen at a higher level.

high cut filter A high cut filter is essentially a lowpass filter that offers no slope or resonance controls. As the name suggests, it attenuates frequencies above the cutoff.

highpass filter A highpass filter allows frequencies above the cutoff frequency to pass. A highpass filter that offers no slope or resonance controls is generally known as low cut filter.

icon Small graphic symbol. In Logic Pro, an icon may be assigned to each track.

importing The process of bringing files of various types into a project. Imported files can be created in another application, captured from another device, or taken from another project.

Insert slot A panel on Logic Pro channel strips where you can patch in (insert) an effect plug-in.

Inspector Area at the left edge of the Logic Pro Arrange and editors, containing the Parameter boxes and channel strips (Arrange) of the selected track. Also see Parameter boxes. The Inspector area updates to reflect the parameters relevant to the window with key focus.

instrument channel Logic Pro and MainStage support the use of software based instruments. Software instrument plug-ins are inserted into the Instrument slot of instrument channels.

instrument object An object in the Logic Pro Environment designed to communicate with a single-channel MIDI device. An instrument object represents a physical or virtual device which handles MIDI information. Also see multi instrument object.

interface 1) A hardware component such as a MIDI or audio device that allows your Logic Studio applications to interface (connect) with the outside world. You need an audio or MIDI interface to get sound or MIDI into and out of your computer. Also see audio interface. 2) A term that is used to describe graphical elements within your Logic Studio applications that you can interact with. An example would be the Logic Pro Arrange, where graphical interface elements such as regions are interacted with to create a project, within the overall Logic Pro 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. Can also refer to a black or white key on a musical (MIDI) keyboard.

key command Function which can be executed by pressing a specific key (or key combination) on your computer keyboard or MIDI controller.

key focus The selected, active window, is said to have key focus in Logic Pro. Many key commands will only function when a window has key focus. The Inspector also updates to reflect the parameters of a window with key focus.

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 the audio interface in use, audio and MIDI drivers. One factor under your control, however, is the I/O buffer size, which is set in the Devices preferences.

legato Method of musical performance that smoothly connects one note to the next.

level meter A meter that monitors audio input or output levels to or from your computer. You use the level meters in your Logic Studio applications when recording, arranging, and editing audio files, and when creating a mix.

Link button Button in the top left corner of most Logic Pro windows that features the chain link icon. It controls the linking between different windows.

local menu bar 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 for your Logic Studio applications.

loop An audio file that contains recurring rhythmic musical elements, or elements suitable for repetition.

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.

low cut filter A low cut filter is essentially a highpass filter that offers no slope or resonance controls. It attenuates all frequencies below the defined cutoff.

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 screen, offering global functions such as opening, saving, exporting, or importing projects. It does not offer access to local window functions.

master channel strip Channel strip in the Mixer that acts as a separate attenuator stage, changing the gain of all output channel strips without affecting the level relationships between them.

merge Mix, or combine, two or more MIDI events or regions into a single event or region.

metadata Metadata is additional descriptive information that is stored in the file header of a number of file types (AAF, for example). It is used to reference external media, to simplify searches, and more.

metronome A device that produces a sound that taps out the beat. In Logic Pro, it can configured in the Metronome project settings.

MIDI Abbreviation for Musical Instrument Digital Interface. Standardized, asynchronous, serial, event-oriented hardware and software 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 the pitch, length, and volume of a musical note event, among other characteristics.

MIDI channel A MIDI channel is a "tube" for MIDI data, which flows through MIDI ports. Up to 16 separate MIDI channels can pass through a port simultaneously. Tracks recorded in Logic Pro can be directed to different tubes (channels), which can contain different information, and play back through different sounds, assigned to each channel. As examples, channel 1: piano, channel 2: bass, channel 3: strings and so on. This presumes that the receiving devices are capable of receiving data on more than one channel, and that they are capable of playing back different sounds simultaneously (see multi timbral).

MIDI Clock Short MIDI message for clock signals. It is used to provide a timing pulse between MIDI devices. It is accurate to 24 ppqn (pulses per quarter note), although some devices interpolate these pulse values, resulting in a more precise clock signal if each device is capable of interpreting this additional information correctly. Also see: SPP.

MIDI message A message transmitted via MIDI that consists 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 Pro, multi mode sound modules are addressed via multi instrument objects. This MIDI mode and multi mode sound modules are generally referred to as multi timbral (see multi timbral).

MIDI Time Code (MTC) Translation of a SMPTE time code signal into a MIDI standard time code signal. MTC is used to synchronize Logic Pro with MIDI devices, other sequencers, video and audio tape or hard disk machines that support MIDI Time Code. MTC determines absolute time positions and supports start, stop, and continue messages.

mixdown A term commonly used to describe either bouncing (see bounce) or merging of tracks (see merge).

mixing The process of shaping the overall sound of a project by adjusting the volume levels, pan positions, adding EQ and other effects, and using automation to dynamically alter these and other aspects.

M-LAN A Yamaha designed variant of the FireWire interface. It enables Yamaha digital mixers and other devices to be directly connected to a Macintosh FireWire port (see FireWire).

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. Many Logic Studio effects and synthesizers contain modulators.

modulation amount The strength, or intensity, of modulation.

modulation matrix The EXS24 mkll (and some other Logic Studio instruments) contains a grid that allows you to vary a number of target parameters, such as pitch, with a number of modulators (modulation sources). This grid is referred to as the modulation matrix in the EXS24 mkll.

modulation path A modulation path determines which target parameter will be affected 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.

movie See video.

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 the Logic Pro 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-instrument, has a fixed MIDI channel. All sub-channels share the same MIDI port. All other parameters can be set individually. The purpose of the multi instrument object is to address multi-channel MIDI devices, which can receive MIDI data (and play different sounds) on separate MIDI channels.

multi-timbral This term describes an instrument or other device that can play several different sounds at the same time, using multiple MIDI channels. See MIDI multi mode.

multi trigger mode This term is associated with synthesizers such as the ES1. In this mode, a synthesizer envelope is usually retriggered by each incoming note event.

mute Switch off the audio output of a channel or track. You can mute a track or channel by clicking the Mute button in the track list or at the bottom of the channel strip).

native Native refers to host-based processing of effects and software instruments in Logic Pro. The computer CPU natively calculates the effects and instruments. Native also refers to the internal Logic Pro plug-in format, which differs from the Audio Unit format. Logic Pro native plug-ins only work in Logic Pro.

normalize This function applies the current Parameter box settings to the selected MIDI events (by altering the actual events themselves), and clears existing 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, allowing all other frequencies to pass.

note number Pitch of a MIDI note, controlled by the first data byte of a MIDI note event.

OpenTL Abbreviation for Open Track List. This file format, typically used for data exchange with Tascam hard disk recorders, such as the MX 2424, can be imported and exported by Logic Pro. The OpenTL file format only supports the exchange of audio data (audio regions, inclusive of track position information). MIDI and automation data are ignored when using the Logic Pro OpenTL export function.

option Alternative function, often in the form of a checkbox, sometimes also available as a menu entry.

Option Modifier key, also known as the Alt key in MS Windows.

oscillator A synthesizer oscillator generates an alternating current, using a selection of waveforms which contain different amounts of harmonics.

output channel strip Channel strip type in the Mixer that controls the output level and pan/balance for each physical output of your audio interface.

pan, pan position The placement of mono audio signals in the stereo field, by setting different levels on both sides (see Balance).

peak meter A digital audio meter that displays the absolute volume of an audio signal as it plays. So named because every peak in the signal can be accurately seen.

peak 1) The highest level in an audio signal. 2) Portions of a digital audio signal that exceed 0 dB, resulting in clipping.

pink noise A harmonic noise type that contains more energy in the lower frequency range.

pitch The perceived highness (treble) or lowness (bass) of a musical sound. Corresponds to the frequency of the sound wave.

pitch bend message MIDI message transmitted by the pitch bend wheel of a MIDI keyboard.

playback Playing an audio or MIDI region, an audio file, or an entire arrangement, allowing you to hear it.

plug-in Software application that enhances the functionality of the main program (in this case, one of your Logic Studio applications).

plug-in window A window that opens 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 volume 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 volume fader in the signal flow, so the level of a signal routed pre-fader to a send remains constant, regardless of any fader movements.

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

PWM Pulse Width Modulation. Synthesizers often feature this facility, where a square waveform is deformed by adjusting its pulse width. A square waveform usually sounds hollow, and woody, whereas a pulse width modulated square wave sounds more reedy and nasal.

Q factor A term generally associated with equalizers. The Q factor is the quality factor of the equalization, and is used to select a narrower or broader frequency range within the overall sonic spectrum of the incoming signal.

quantization Time-correction of note positions by moving them to the nearest point on a selectable grid.

QuickTime QuickTime is an Apple cross-platform standard for digitized, data-compressed video playback and encoding.

RAM Abbreviation for random-access memory. Computer memory capacity, measured in megabytes (MB), determines the amount of data the computer can process and temporarily store at any given moment.

recording The act of capturing a performance as audio or MIDI data. The term is also commonly used when referring to the actual data (in Logic Pro, this is delineated by the use of the words region or file to make things clearer when discussing recordings).

ReCycle ReCycle is the name of an application from software manufacturer Propellerhead, which mainly serves as an editing and production tool for loops (repeated audio samples). ReCycle uses specific file formats (.rex) which can be imported by Logic Pro.

region Regions can be found on Logic Pro Arrange tracks: 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 (usually referred to as folders). Also see: audio region, MIDI region, and folder.

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 physical 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.

ReWire An audio streaming and synchronization technology from Propellerhead software. The output of ReWired applications can be routed into (and processed with) the Logic Pro Mixer. Logic Pro can also control the transport operations of ReWired programs. Further to these functions, Logic Pro instrument tracks can drive the software instruments of ReWire applications.

ritardando A gradual slowdown of tempo (see tempo).

routing Generally refers to the way audio is sent through processing units. Also often used to describe specific input and output assignments.

rubber band selection Technique for selecting consecutive regions, objects, or events by click-holding and dragging the mouse cursor around the desired items. A rubber band selection envelope (an outline) will expand from the starting position of the mouse cursor. All objects touched or enclosed by the rubber band selection envelope will be selected.

sample A digital recording of a sound at a particular instant in time.

sampler Device used for sampling. In Logic Pro and MainStage, this generally refers to the EXS24 mkll 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.

sample rate converter Device or algorithm that translates one sample rate to another.

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.

scale A group of related musical notes (or pitches) that forms the basis of the melody and harmony in a piece of music. The most common scales are the major scale and minor scale.

scan code Each key on a computer keyboard has a scan code rather than an ASCII symbol associated with it. As an example: The plus and minus keys on the numeric keypad and the corresponding keys above the keyboard have a different scan code, but use the same ASCII symbol.

scroll bar and scroller Gray beam at the edge of a window. A movable box inside the beam is used to select the displayed project section in the window.

SDII Sound Designer II audio file format. Very similar in structure to the AIFF file format.

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. A semitone is also called a 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 often used to send several signals to the same effect, which is useful for computationally-intensive effects such as reverb.

sequencer A sequencer is these days considered to be a computer application that allows you to record both digital audio and MIDI data and blend the sounds together in a software mixing console. In earlier times, a sequencer controlled synthesizers via a series of control voltages and gates, or via MIDI only. No audio recording or control was available.

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 Found in the gray header at the top of all Logic Pro 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 the frequency range above or below the specified frequency to be increased or decreased.

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 ES1. In this mode, envelopes are not retriggered when tied (legato) notes are played.

SMF See Standard MIDI File.

SMPTE Abbreviation for Society of Motion Picture and Television Engineers. The organization responsible for establishing a synchronization system that divides time into hours, minutes, seconds, frames, and subframes (SMPTE time code). SMPTE time code is also used for synchronizing different devices. The MIDI equivalent of SMPTE time code is MIDI Time Code (MTC). See MTC.

software instrument Software counterpart to hardware samplers or synthesizer modules, or acoustic sound sources such as drumkits or guitars. The sounds generated by software instruments are calculated by the computer CPU, and played via the audio interface outputs. Often colloquially called softsynths or softsamplers.

solo A way to temporarily highlight one or more tracks or regions or events, allowing them to be heard in isolation.

SPP Abbreviation for Song Position Pointer, a MIDI clock timing pulse sub-message that indicates the current "song" (project) position. It is accurate to bars (and beats for some devices), but is not as accurate as MIDI Time Code (MTC).

S/P-DIF Short for Sony/Philips Digital Interface, a standard transmission format for professional stereo digital audio signals. The format is similar to AES/EBU, but uses 75 ohm coaxial or optical connectors and cabling. Depending on the type of devices involved, AES/EBU and S/P-DIF coaxial interfaces can communicate directly. Most digital audio interfaces available today will feature S/P-DIF connectors.

Standard MIDI file (SMF) Standard file format for exchanging songs between different sequencers or MIDI file players. Standard MIDI files are not specific to a particular sequencer program, type of computer, or device. Any sequencer should be able to interpret at least the type 0 MIDI file format. Standard MIDI files contain information about MIDI events, including time positions and channel assignments, names of individual tracks, instrument names, controller data, tempo changes, and more.

status byte First byte in a MIDI message, which determines the type of message.

step sequencer While all sequencers, including Logic Pro, step through a series of events, this term is used to describe a device from the seminal years of analog synthesizers. Essentially, two rows of knobs (usually 8) were individually adjusted to control the gate time (note length) and pitch of a connected synthesizer. The sequencer would step through these knob settings once, or repeatedly. Many modern software instruments, particularly drum synthesizers, include an integrated step sequencer that synchronizes with Logic Pro playback. Ultrabeat incorporates a step sequencer which is considerably more flexible than its ancient analog cousins.

stereo Short for stereophonic sound reproduction of two different audio channels. Compare with mono.

subframe A sub-division of a SMPTE frame, corresponding to the individual bits of a SMPTE frame. One frame consists of 80 bits.

surround Surround indicates playback systems that make use of multiple speakers. The most common surround format is 5.1 channels (front left, front center, front right, left surround, right surround, and an LFE, or subwoofer, channel), typically used in home theater systems and in cinemas.

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 Pro.

Swing parameter Alters the rigid timing of a quantization grid by delaying every other note of a specified sub-division by a definable amount.

synchronization Method of keeping several recording or playback devices time locked with one another. In virtually all synchronized setups, there will be one master device and one or more slave devices that derive their synchronization clock from the master.

synchronizer Central unit used to control the synchronization of several devices.

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 Studio features several software synthesizers, including the ES1, ES2, EFM1, ES E, ES P, and ES M.

SysEx Abbreviation for System Exclusive data. SysEx data forms the top tier in the hierarchy of MIDI commands. These messages are tagged with an identification number for each manufacturer (the SysEx manufacturer ID number). The actual contents of these MIDI commands is left to the manufacturer. SysEx data is often used to transfer individual (or banks of) sound programs or system settings, or to address individual sound generation or signal processing parameters.

takes A take, put simply, is a recording. Logic Pro allows you to create several takes, one after the other, without leaving record mode. These takes can then be compiled into a super take (see comping).

template A project that contains settings and preferences that you have defined. Templates serve as a starting point for new projects (scoring tasks, audio only projects, MIDI only projects, and so on, as your personal needs require). Any project can be used as a template, and you can create and save multiple templates.

tempo The playback speed of a piece of music, measured in beats per minute.

tick The smallest unit of timing resolution in a MIDI sequencer. In Logic Pro, this is 1/3840th of a note. Logic Pro can go down to single sample accuracy (at sufficient zoom levels) for edits and positioning, but the MIDI protocol is not fast enough to support this.

time code A format (and signal) for assigning a unique, sequential time unit to each frame of video or project position. The SMPTE time code format, for example, is measured in hours: minutes: seconds: frames and subframes.

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).

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.

transposition Transposition is the act of changing the pitch of an audio or MIDI region (or event) by a number of semitones.

treble Refers to high frequency sounds or components within a sound. See frequency.

Undo function Function which reverses the previous editing operation.

unicode Fundamentally, computers just deal with numbers. They store letters and other characters by assigning a number for each one. Unicode provides a unique number for every character, no matter what the platform, no matter what the program, no matter what the language.

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.

VU meter Abbreviation for Volume Unit meter. An analog meter used to monitor audio levels.

WAV, WAVE The primary audio file format used by Windows-compatible computers. In Logic Pro, all recorded and bounced WAV files are in Broadcast Wave format, which includes high-resolution timestamp information that stores positional information. This makes it easy to align these files in other audio and video applications.

waveform A visual representation of an audio signal. Waveform graphics run from left to right, and are centered on a horizontal line. Louder portions of the waveform (amplitude peaks) are indicated as taller spikes or higher curves in the waveform.

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 type 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 Clock Clock signal required by digital audio interfaces to ensure the sampling rates of connected devices run synchronously. When two devices are connected via a standard digital audio interface (such as S/P-DIF or ADAT optical), Word Clock is transmitted via the audio circuit. If you want more than two digital audio devices to communicate with each other, you will need to use separate Word Clock ports for synchronization, in most cases.

word length See bit depth.

XG Extended General MIDI standard from Yamaha, compatible with Roland GS.

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 window.

zoom level The amount that window contents 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 project and work on very large sections.

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