Logic Express 9

Effects

Note: Because Apple frequently releases new versions and updates to its system software, applications, and Internet sites, images shown in this manual may be slightly different from what you see on your screen.

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Contents

An Introduction to the Logic Express Effects

Logic Express has an extensive range of digital signal processing (DSP) effects and processors that are used to color or tonally shape existing audio recordings, software instruments, and external audio sources—in real time. These will cover almost every audio processing and manipulation need you will encounter in your day-to-day work.

The most common processing options include EQs, dynamic processors, modulations, distortions, reverbs, and delays.

Less common are simulations of amplifiers and speaker cabinets, which enable you to "play" your instruments or other signals through a range of vintage and modern sound reproduction systems. Guitarists will also benefit from a number of classic pedal effect emulations.

Further advanced features include precise signal meters and analyzers, a test tone generator, noise reduction, pitch correction, imaging, bass enhancement, and time-altering processors and utilities.

As you can see, many of the included processors and utilities don't really fall into the "effects" category, but they may prove to be invaluable in your studio.

All effects, processors, and utilities provide an intuitive interface that simplifies operation, enabling you to work quickly. Outstanding audio quality is assured when needed, or—at the other end of the spectrum—extreme processing is possible when you need to radically alter your audio. All effects and processors are highly optimized for efficient CPU usage.

This preface covers the following:

About the Logic Express Effects

Logic Express includes a comprehensive suite of effects processors and utilities that can be used to enhance your music projects. Effects are grouped in the following categories.

Effect category	Included effects
Amp Modeling	Amp Designer
	Bass Amp
	Guitar Amp Pro
	Pedalboard
Delay	Echo
	Sample Delay
	Stereo Delay
	Tape Delay
Distortion	Bitcrusher
	Clip Distortion
	Distortion Effect
	Distortion II
	Overdrive
	Phase Distortion
Dynamics	Compressor
	DeEsser
	Ducker
	Enveloper
	Expander
	Limiter
	Noise Gate
	Preset Multipressor
	Silver Compressor
	Silver Gate
EQ	Channel EQ
	DJ EQ
	Fat EQ
	Single-Band EQs
	Silver EQ
Filter	AutoFilter
	EVOC 20 Filterbank
	EVOC 20 TrackOscillator
	Fuzz-Wah
	Spectral Gate

Effect category	Included effects
Imaging	Direction Mixer
	Stereo Spread
Metering	BPM Counter
	Correlation Meter
	Level Meter Plug-in
	Tuner
Modulation	Chorus Effect
	Ensemble Effect
	Flanger Effect
	Microphaser
	Modulation Delay
	Phaser Effect
	Ringshifter
	Rotor Cabinet Effect
	Scanner Vibrato Effect
	Spreader
	Tremolo Effect
Pitch	Pitch Correction Effect
	Pitch Shifter II
	Vocal Transformer
Reverb	AVerb
	EnVerb
	GoldVerb
	PlatinumVerb
	SilverVerb
Specialized	Denoiser
	Enhance Timing
	Exciter
	Grooveshifter
	Speech Enhancer
	SubBass
Utility	Gain Plug-in
	I/O Utility
	Test Oscillator

About the Logic Express Documentation

Logic Express comes with various documents that will help you get started as well as provide detailed information about the included applications.

- *Logic Express User Manual:* This onscreen manual provides comprehensive instructions for using Logic Express to set up a recording system, compose music, edit audio and MIDI files, and output audio for CD productions.
- *Exploring Logic Express:* This booklet provides a fast-paced introduction to the main features and tasks in Logic Express, encouraging hands-on exploration for new users.
- *Logic Express Control Surfaces Support:* This onscreen manual describes the configuration and use of control surfaces with Logic Express.
- *Logic Express Instruments:* This onscreen manual provides comprehensive instructions for using the powerful collection of instruments included with Logic Express.
- *Logic Express Effects:* This onscreen manual provides comprehensive instructions for using the powerful collection of effects included with Logic Express.
- *Logic Express Working with Apogee Hardware:* This onscreen manual describes the use of Apogee hardware with Logic Express.

Additional Resources

In addition to the documentation that comes with Logic Express, there are a variety of other resources you can use to find out more.

Release Notes and New Features Documents

Each application offers detailed documentation that covers new or changed features and functions. This documentation can be accessed in the following way:

- Open the application Help menu and choose Release Notes or New Features.

Logic Express Website

For general information and updates, as well as the latest news on Logic Express, go to:

- http://www.apple.com/logicexpress

Apple Service and Support Websites

For software updates and answers to the most frequently asked questions for all Apple products, go to the general Apple Support webpage. You'll also have access to product specifications, reference documentation, and technical articles about Apple products and products from other companies.

- http://www.apple.com/support

For software updates, documentation, discussion forums, and answers to the most frequently asked questions for Logic Express, go to:

- http://www.apple.com/support/logicexpress

For discussion forums for all Apple products from around the world, where you can search for an answer, post your question, or answer other users' questions, go to:

- http://discussions.apple.com

Amps and Pedals

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Logic Express features an extensive collection of guitar and bass amplifiers and classic pedal effects. You can play live—or process recorded audio and software instrument parts—through these amps and effects.

The amplifier models re-create vintage and modern tube and solid-state amps. Built-in effect units, such as reverb, tremolo, or vibrato, are also reproduced. Accompanying the amplifiers are a variety of emulated speaker cabinets, which can be used as a matching set or combined in different ways to create interesting hybrids.

Also emulated are a number of "classic" foot pedal effects—or *stompboxes*—that were, and remain, popular with guitarists and keyboardists. As with their real-world counterparts, you can freely chain pedals in any order to create the perfect sound.

This chapter covers the following:

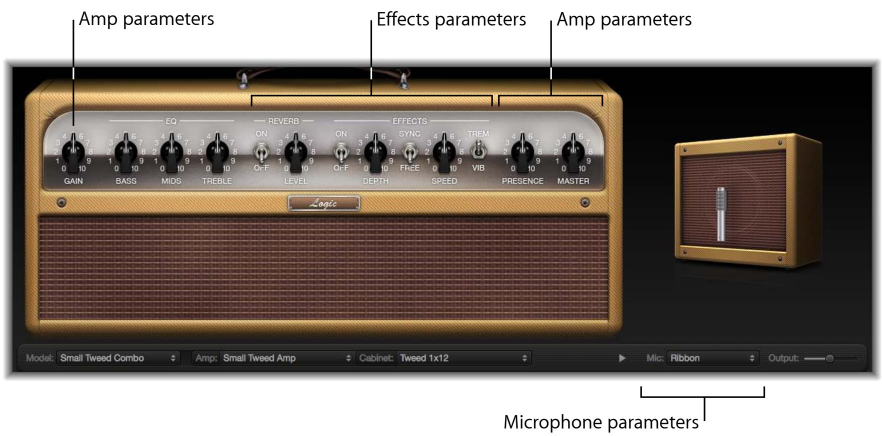
Amp Designer

Amp Designer emulates the sound of over 20 famous guitar amplifiers and the speaker cabinets used with them. Each preconfigured model combines an amp, cabinet, and EQ that re-creates a well-known guitar amplifier sound. You can process guitar signals directly, which allows you to reproduce the sound of your guitar played through these amplification systems. Amp Designer can also be used for experimental sound design and processing. You are free to use it with other instruments, applying the sonic character of a guitar amp to a trumpet or vocal part, for example.

The amplifiers, cabinets, and EQs emulated by Amp Designer can be combined in a number of ways to radically or subtly alter the tone. Virtual microphones are used to pick up the signal of the emulated amplifier and cabinet. You can choose from three different microphone types, and you can reposition them.

Amp Designer also emulates classic guitar amplifier effects, including spring reverb, vibrato, and tremolo.

The Amp Designer interface can be broken down into four general sections in terms of different kinds of parameters.



- *Model parameters:* The Model pop-up menu is found at the left of the black bar at the bottom. It is used to choose a preconfigured model, consisting of an amplifier, a cabinet, an EQ type, and a microphone type. See Choosing an Amp Designer Model. The model-customizing parameters on the black bar allow you to independently choose the type of amplifier and cabinet. See Building a Customized Amp Designer Combo. The EQ type is chosen from the EQ pop-up menu above the Bass, Mids, and Treble knobs in the knobs section. See Using Amp Designer's Equalizer.
- *Amp parameters:* Located at each end of the knobs section, these parameters are used to set an amp's input gain, presence, and output level. See Using Amp Designer's Gain, Presence, and Master Controls.
- *Effects parameters:* Located in the center of the knobs section, these parameters allow you to control the integrated guitar effects. See Getting to Know Amp Designer's Effects Parameters.
- *Microphone parameters:* Located slightly above the right end of the black bar at the bottom, these parameters are used to set the type and position of the microphone that captures the amplifier and cabinet sound. See Setting Amp Designer Microphone Parameters.

To switch between full and smaller versions of the interface

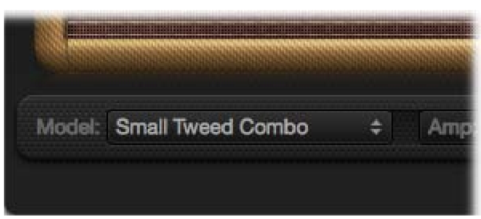
- Click the disclosure triangle between the Cabinet and Mic pop-up menus in the full interface to switch to the smaller version. To switch back to the full interface, click the disclosure triangle beside the Output field in the small interface. You can access all the parameters, with the exception of microphone selection and positioning, in the small interface.

Click here in full interface.

Click here in small interface.

Choosing an Amp Designer Model

You can choose a preconfigured model—consisting of an amplifier, a cabinet, an EQ type, and a microphone type—from the Model pop-up menu at the left end of the black bar at the bottom of the Amp Designer interface. Your choices include several combinations in each of the following categories:



- Tweed Combos
- Classic American Combos
- British Stacks
- British Combos
- British Alternatives
- Metal Stacks
- Additional Combos

Tweed Combos

The Tweed models are based on American combos from the 1950s and early 1960s that helped define the sounds of blues, rock, and country music. They have warm, complex, clean sounds that progress smoothly through gentle distortion to raucous overdrive as you increase the gain. Even after half a century, Tweeds can still sound contemporary. Many modern boutique amplifiers are based on Tweed-style circuitry.

Model	Description
Small Tweed Combo	A 1 x 12" combo that transitions smoothly from clean to crunchy, making it a great choice for blues and rock. For extra definition, set the Treble and Presence controls to a value around 7.
Large Tweed Combo	This 4 x 10" combo was originally intended for bassists, but was also used by blues and rock guitarists. More open and transparent-sounding than the Small Tweed Combo, but can deliver crunchy sounds.
Mini Tweed Combo	A small amp with a single 10" speaker, used by countless blues and rock artists. It is quite punchy-sounding, and can deliver the clean and crunch tones that the Tweed combos are known for.

Tip: Tweed combos respond beautifully to your playing dynamics. Adjust the knobs to create a distorted sound, then reduce the level of your guitar's volume knob to create a cleaner tone. Turn up your guitar's volume knob when the time comes for a scorching solo.

Classic American Combos

The Blackface, Brownface, and Silverface models are inspired by American combos of the mid 1960s. These tend to be loud and clean with tight lows and relatively restrained distortion. They are great for clean-toned rock, vintage R & B, surf music, twangy country, jazz, or any other style where strong note definition is essential.

Model	Description
Large Blackface Combo	A 4 x 10" combo with a sweet, well-balanced tone favored by rock, surf, and R & B players. Great for lush, reverb-drenched chords or strident solos.
Silverface Combo	A 2 x 12" combo with a loud, ultra-clean tone. Its percussive, articulate attack is great for funk, R & B, and intricate chord work. It can be crunchy when overdriven, but most players favor it for clean tones.
Mini Blackface Combo	A 1 x 10" combo that is bright and open-sounding, with a surprising amount of low-end impact. It excels at clean tones with just a hint of overdrive.
Small Brownface Combo	A 1 x 12" combo that is smooth and rich-sounding, but retains a nice level of detail.
Blues Blaster Combo	A 1 x 15" combo that has a clear top end with a tight, defined low end. This model is favored by blues and rock players.

Tip: While these amps tend toward a clean and tight sound, you can use a Pedalboard distortion stompbox to attain hard-edged crunch sounds with a biting treble and extended low-end definition. See Distortion Pedals and Pedalboard.

British Stacks

The British Stack models are based on the 50- and 100-watt amplifier heads that have largely defined the sound of heavy rock, especially when paired with their signature 4 x 12" cabinets. At medium gain settings, these amps are great for chunky chords and riffs. Raising the gain yields lyrical solo tones and powerful rhythm guitar parts. Complex peaks and dips across the tonal spectrum keep the tones clear and appealing, even when heavy distortion is used.

Model	Description
Vintage British Stack	Captures the sound of a late 1960s 50-watt amp famed for its powerful, smooth distortion. Notes retain clarity, even at maximum gain. After four decades this remains a definitive rock tone.
Modern British Stack	1980s and 1990s descendants of the Vintage British amplifier head, which were optimized for hard rock and metal styles of the time. The tones are deeper on the bottom, brighter on top, and more “scooped” in the middle than the Vintage British amp.
Brown Stack	Unique tones can be coaxed from a British head by running it at lower voltages than its designers intended. The resulting “brown” sound—often more distorted and loose than the standard tone—can add interesting thickness to a guitar sound.
British Blues Combo	This 2 x 12" combo has a loud, aggressive tone that is cleaner than the British heads, yet delivers fat distortion tones at high-gain settings.

Tip: You’ll rarely go wrong combining a British head, a 4 x 12" cabinet, and a great riff at high levels. But don’t hesitate to break that mold. These heads can sound stunning through small cabinets, or at clean, low-gain settings. If the British Blues Combo is too clean for your needs, combine it with Pedalboard’s Hi Drive stompbox for an aggressive blues tone, or the Candy Fuzz stompbox for an explosive rock tone. See Distortion Pedals and Pedalboard.

British Combos

The British Combos capture the brash, treble-rich sound that will forever be associated with 1960s British rock and pop. The sonic signature of these amps is characterized by their high-end response, yet they are rarely harsh-sounding due to a sweet distortion and smooth natural compression.

Model	Description
British Combo	A 2 x 12" combo based on the early 1960s amps that powered the British Invasion. Perfect for chiming chords and stabbing solos.

Model	Description
Small British Combo	A 1 x 12" combo with half the power of the British Combo, this amp offers a slightly darker, less open tone.
Boutique British Combo	A 2 x 12" combo that is a modern take on the original 1960s sound. The tone is thicker, with stronger lows and milder highs than the other British Combos.

Tip: Using high Treble and Presence knob settings that might become strident on other amp types can sound great with the British Combos.

British Alternatives

The late 1960s amplifier heads and combos that inspired the Sunshine models are loud and aggressive, with full-bodied mid frequencies. These amps are not just for single note solos and power chords, as they can sound great with big, open chords—one reason why they were embraced by the "Brit-pop" bands of the 1990s. The Stadium amps are famed for their ability to play ultra-loud without dissolving into mushy distortion. They retain crisp treble and superb note definition, even at maximum gain settings.

Model	Description
Sunshine Stack	A robust-sounding head paired with a 4 x 12" cabinet. It's a great choice for powerful pop-rock chords.
Small Sunshine Combo	A 1 x 12" combo based on a modern amp known for a "big amp" sound. It is brighter than the Sunshine Stack head, with a touch of 1960s British Combo flavor.
Stadium Stack	A classic head and 4 x 12" cabinet configuration popular with 1970s arena rock bands. Its tones are cleaner than other Amp Designer 4 x 12" stacks, while still retaining body and impact. A good choice if you need power *and* clarity.
Stadium Combo	A 2 x 12" combo based on a modern amp. The tone is a little smoother and rounder than that of the Stadium Stack.

Tip: The tone of the Sunshine Stack can seem dark at times, but a high Treble knob setting opens up the sound. While the Small Sunshine Combo sounds great with its default 1 x 12" cabinet, it also shines through a 4 x 12" cabinet. The Stadium amps can be slow to distort, so most famous users have paired them with aggressive fuzz pedals. Try combining it with Pedalboard's Candy Fuzz or Fuzz Machine stompboxes. See Distortion Pedals and Pedalboard.

Metal Stacks

The Metal Stack models are inspired by the powerful, ultra-high gain amplifier heads that put the "chunk" into modern hard rock and metal music. All are paired with 4 x 12" cabinets. Their signature tones range from heavy distortion to extremely heavy distortion. If you want powerful lows, razor-edged highs, and serious sustain, these are the models you should look to first.

Model	Description
Modern American Stack	A powerful, ultra-high gain amp that is ideal for heavy rock and metal. Use the Mids knob to set an ideal amount of scoop or boost.
High Octane Stack	Although a powerful, high-gain amp, this model offers a smooth transition between gain settings and excellent natural compression. It is a great choice for fast soloing and for two- and three-note chords.
Turbo Stack	An aggressive-sounding amp with spiky highs and noisy harmonics, especially at high gain settings. Try the Turbo Stack when you need to slice through a mix.

Tip: Combining the Turbo Stack with distortion and fuzz pedals may actually diminish the amp's edge. A dry sound is often the best choice for high-impact riffs.

Additional Combos

The combos and utility models in this category are versatile amps that can be used for a wide variety of musical styles.

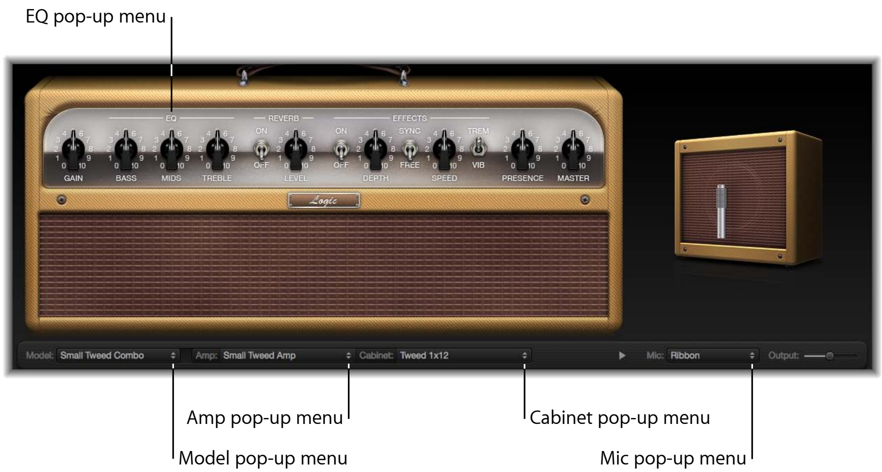
Model	Description
Studio Combo	A 1 x 12" combo based on boutique combos of the 1980s and 1990s that use multiple gain stages to generate smooth, sustain-heavy distortion without sacrificing bold, bright, clean sounds.
Boutique Retro Combo	A 2 x 12" combo inspired by high-end modern amps that combine the sounds of several great 1960s combos. It excels at shimmering clean tones and crunch tones, making it a good choice when you want an old-fashioned flavor, but with the crisp highs and defined lows of a modern amplifier.
Pawnshop Combo	A 1 x 8" combo based on the inexpensive amps sold in American department stores in the 1960s. Despite their limited features and budget workmanship, these amps are the secret behind the sound of many rock, blues, and punk players. The clean sounds are warm, and distorted sounds are thick and satisfying, despite the small speaker.
Transparent Preamp	As the name suggests, a preamp stage with no coloration. You should note that the Transparent Preamp is activated in the Amp pop-up menu, not in the Model pop-up menu.

Tip: Try pairing the Studio Combo amp with one of the 4 x 12" cabinets for a heavier sound. The Boutique Retro Amp has very sensitive tone controls, providing countless tonal shadings. Even extreme settings can yield great results. Combine the Pawnshop Combo amp with Pedalboard's Hi Drive or Candy Fuzz stompboxes to emulate hard rock tones of the late 1960s. See Distortion Pedals and Pedalboard.

Building a Customized Amp Designer Combo

You can use one of the default models or you can create your own hybrid of different amplifiers, cabinets, and so on, using the Amp, Cabinet, and Mic pop-up menus, located on the black bar at the bottom of the interface. The EQ pop-up menu is accessed by clicking the word *EQ* or *Custom EQ* toward the left of the knobs section.

Note: If you create your own hybrid amp combo, you can use the Settings menu to save it as a setting file, which also includes any parameter changes you may have made.



Building an Amp Designer model is described in the following sections:

- Choosing an Amp Designer Amplifier
- Choosing an Amp Designer Cabinet
- Using Amp Designer's Equalizer
- Setting Amp Designer Microphone Parameters

Choosing an Amp Designer Amplifier

You can choose an amplifier model from the Amp pop-up menu on the black bar at the bottom of the Amp Designer interface. See the following sections for details on the characteristics of each amplifier in these categories:

- Tweed Combos
- Classic American Combos
- British Stacks
- British Combos
- British Alternatives
- Metal Stacks

- Additional Combos

Choosing an Amp Designer Cabinet

Cabinets have a huge impact on the character of a guitar sound (see Amp Designer Cabinet Reference Table). While certain amplifier and cabinet pairings have been popular for decades, departing from them is an effective way to create fresh-sounding tones. For example, most players automatically associate British heads with 4 x 12" cabinets. Amp Designer allows you to drive a small speaker with a powerful head, or to pair a tiny amp with a 4 x 12" cabinet.

There's nothing wrong with trying random combinations. But if you consider the variables that determine a cabinet's sound, you'll be able to make educated guesses about non-traditional amplifier and cabinet combinations. Some factors to consider:

Combos or Stacks

Combo amps include both an amplifier and speakers in a single enclosure. These usually have an open back, so the sound resonates in multiple directions. The resulting sound is "open"—with bright, airy highs and a general feeling of spaciousness. Amplifier "stacks" consist of an amplifier head, with the speakers in a separate cabinet. These cabinets generally have a closed back, and project the sound forward in a tight, focused "beam." They tend to sound more powerful than open-back cabinets, and typically have a tighter low-end response at the expense of some high-end transparency.

Old or New Speakers

Amp Designer models that are based on vintage cabinets capture the character of aged speakers. These may be a bit looser and duller-sounding than new speakers, but many players prefer them for their smoothness and musicality. Sounds based on new cabinets tend to have more snap and bite.

Large Speakers or Small Speakers

A larger speaker doesn't guarantee a larger sound. In fact, the most popular bass guitar cabinet of all time uses only small 8" speakers. Don't be surprised if you get a deeper, richer tone from a 10" speaker than from a large 4 x 12" cabinet. Try several sizes and choose the one that works best for your music.

Single Speakers or Multiple Speakers

Guitarists sometimes use cabinets with multiple speakers, and not only for the larger sound they tend to provide. Phase cancellations occur between the speakers, adding texture and interest to the tone. Much of the "classic rock" sound, for example, has to do with the tonal peaks and dips caused by this interaction between the speakers in a 4 x 12" cabinet.

Amp Designer Cabinet Reference Table

You can choose a cabinet model from the Cabinet pop-up menu on the black bar at the bottom of Amp Designer's interface. The table below covers the properties of each cabinet model available in Amp Designer.

Cabinet	Description
Tweed 1 x 12	A 12" open-back cabinet from the 1950s with a warm and smooth tone.
Tweed 4 x 10	A 4 x 10" open-back cabinet that was originally conceived for bassists, but guitarists love its sparkling presence. An authentic late 1950s sound.
Tweed 1 x 10	A single 10" open-back combo amp cabinet from the 1950s with a smooth sound.
Blackface 4 x 10	Classic open-back cabinet with four 10" speakers. Its tone is deeper and darker than the Tweed 4 x 10.
Silverface 2 x 12	An open-back model from the 1960s that provides great low-end punch.
Blackface 1 x 10	An open-back 1960s cabinet with glistening highs and surprising low-mid body.
Brownface 1 x 12	A beautifully balanced 1960s open-back cabinet. It is smooth and rich-sounding, but with nice transparency.
Brownface 1 x 15	This early 1960s open-back cabinet houses the largest speaker emulated by Amp Designer. Its highs are clear and glassy, and its lows are tight and focused.
Vintage British 4 x 12	This late 1960s closed-back cabinet is synonymous with classic rock. The tone is big and thick, yet also bright and lively, thanks to the complex phase cancellations between the four 30-watt speakers.
Modern British 4 x 12	A closed-back 4 x 12" cabinet that is brighter, and has a better low-end than the Vintage British 4 x 12, with less mid-range emphasis.
Brown 4 x 12	A closed-back 4 x 12" cabinet with a great bottom end and complex mid-range.
British Blues 2 x 12	A bright-sounding open-back cabinet with solid lows, and highs that maintain their edge even at high gain settings.
Modern American 4 x 12	A closed-back 4 x 12" cabinet that has a full sound. The low-mids are denser than the British 4 x 12" cabinets.
Studio 1 x 12	A compact-sounding open-back cabinet with full mids and shimmering highs.
British 2 x 12	A mid 1960s open-back cabinet with an open, smooth tone.
British 1 x 12	A small open-back cabinet with crisp highs and nice low-mid transparency.
Boutique British 2 x 12	A 2 x 12" cabinet based on the British 2 x 12. It has a richer mid-range and is more assertive in the treble range.

Cabinet	Description
Sunshine 4 x 12	A 4 x 12" closed-back cabinet with a thick, rich mid-range.
Sunshine 1 x 12	A single 12" open-back combo amp cabinet with a bright, lively sound that has sweet highs, and transparent mids.
Stadium 4 x 12	A tight, bright, closed-back British cabinet with bold upper-mid peaks.
Stadium 2 x 12	A nicely balanced modern British open-back cabinet. Tonally, it is a compromise between the fatness of the Blackface 4 x 10 and the brilliance of the British 2 x 12.
Boutique Retro 2 x 12	A 2 x 12" cabinet based on the British 2 x 12. It has a rich, open mid-range and is more assertive in the treble range.
High Octane 4 x 12	A modern, closed-back European cabinet with strong lows and highs and scooped mids appropriate for metal and heavy rock.
Turbo 4 x 12	A modern, closed-back European cabinet with strong lows, very strong highs, and deeply scooped mids appropriate for metal and heavy rock.
Pawnshop 1 x 8	Single 8" speaker cabinet that has excellent low-end punch.
Direct	This option bypasses the speaker emulation section.

Using Amp Designer's Equalizer

Hardware amplifier tone controls vary between models and manufacturers. There's a good chance, for example, that the treble knobs on two different models target different frequencies, or provide different levels of cut or boost. Some equalizer (EQ) sections amplify the guitar signal more than others, affecting the way the amp distorts.

Amp Designer provides multiple EQ types to mirror these variations in hardware amplifiers. No matter which EQ type you choose, you'll see an identical set of controls: Bass, Mids, and Treble. Switching between EQ types can result in these controls behaving very differently.

Selecting an EQ type other than the one traditionally associated with a certain amplifier typically results in significant tonal changes, although these may not necessarily be for the better. As with hardware amplifiers, Amp Designer's EQs are calibrated to perform well with particular amplifier sounds. Choosing other EQ types can sometimes produce a thin, or unpleasantly distorted tone. See Amp Designer Equalizer Type Reference Table.

Despite these less pleasant-sounding possibilities, you should experiment with different amplifier and EQ combinations because many will sound great together.

EQ pop-up menu

Bass, Mids, and Treble knobs

The EQ parameters include the EQ pop-up menu and the Bass, Mids, and Treble knobs. These parameters are found toward the left-end of the knobs section.

- *EQ pop-up menu:* Click the word *EQ* or *CUSTOM EQ* above the Bass, Mids, and Treble knobs to open the EQ pop-up menu, which contains the following EQ models: British Bright, Vintage, U.S. Classic, Modern, and Boutique. Each EQ model has unique tonal qualities that affect the way the Bass, Mids, and Treble knobs respond. See Amp Designer Equalizer Type Reference Table.
- *Bass, Mids, and Treble knobs:* Adjust the frequency ranges of the EQ models, similar to the tone knobs on a hardware guitar amplifier. The behavior and response of these knobs changes when different EQ models are chosen.

Amp Designer Equalizer Type Reference Table

You can choose an Equalizer type by clicking the word *EQ* or *CUSTOM EQ* above the Bass, Mids, and Treble knobs in the knobs section. The table below covers the properties of each EQ type available in Amp Designer.

EQ type	Description
British Bright	Inspired by the EQ of British combo amps of the 1960s. It is loud and aggressive, with even bolder highs than the Vintage EQ. This EQ is useful if you want more treble definition without an overly clean sound.
Vintage	Emulates the EQ response of American Tweed-style amps and the vintage British stack amps that used a very similar circuit. It is loud and somewhat distortion-prone. This EQ is useful if you want to roughen the sound.
U.S. Classic	Derived from the EQ circuit of the American Blackface-style amps. The tone is of higher fidelity than the Vintage EQ, with tighter lows and crisper highs. This EQ is useful if you want to brighten your tone and reduce distortion.

EQ type	Description
Modern	Based on a digital EQ unit popular in the 1980s and 1990s. This EQ is useful for sculpting the hyped highs, booming lows, and scooped mids associated with the era’s rock and metal music styles.
Boutique	Replicates the tone section of a “retro modern” boutique amp. It excels at precise EQ adjustments, though its tone may be cleaner than desired when used with vintage amplifiers. This EQ is a good choice if you want a cleaner, brighter sound.

Using Amp Designer’s Gain, Presence, and Master Controls

The amp parameters include controls for the input gain, presence, and master output. The Gain knob is found to the left in the knobs section and the Presence and Master knobs are to the right.

- *Gain knob:* Sets the amount of pre-amplification applied to the input signal. This control affects various amp models differently. For example, when you are using the British Amp, the maximum gain setting produces a powerful crunch sound. When you are using the Vintage British Head or Modern British Head, the same gain setting produces heavy distortion, suitable for lead solos.
- *Presence knob:* Adjusts the high-frequency range—above the range of the Treble control. The Presence parameter affects only the output (Master) stage.
- *Master knob:* Sets the output volume of the amplifier going to the cabinet. For tube amplifiers, increasing the Master level typically produces a somewhat compressed and saturated sound, resulting in a more distorted and powerful—that is, louder—signal. High Master settings can produce an extremely loud output that can damage your speakers or hearing, so ramp this up slowly. The final output level of Amp Designer is set with the Output slider at the lower-right edge of the interface. See Setting Amp Designer’s Output Level.

Getting to Know Amp Designer's Effects Parameters

The effects parameters include Tremolo, Vibrato, and Reverb, which emulate the processors found on many amplifiers. these controls are found in the center of the knobs section.

You can use the switch toward the right to select either Tremolo (TREM), which modulates the amplitude or volume of the sound, or Vibrato (VIB), which modulates the pitch.

Reverb, which is controlled by a switch in the middle, can be added to either of these effects, or used independently.

Note: The Effects section is placed *before* the Presence and Master controls in the signal flow, and receives the pre-amplified, pre-Master signal.

Reverb, Tremolo, and Vibrato are described in the following sections:

- Using Amp Designer's Reverb Effect
- Using Amp Designer's Tremolo and Vibrato Effects

Using Amp Designer's Reverb Effect

Reverb is always available in Amp Designer, even when using a model that is based on an amplifier that provides no reverb function. Reverb is controlled by an On/Off switch and a Level knob in the middle, above which is the Reverb pop-up menu. Reverb can be added to either the Tremolo or Vibrato effect, or used independently.

- *On/Off switch:* Enables or disables the reverb effect.
- *Reverb pop-up menu:* Click the word *Reverb* to choose one of the following reverb types from the pop-up menu: Vintage Spring, Simple Spring, Mellow Spring, Bright Spring, Dark Spring, Resonant Spring, Boutique Spring, Sweet Reverb, Rich Reverb, and Warm Reverb. See Amp Designer Reverb Type Reference Table for information on these reverb types.

- *Level knob:* Sets the amount of reverb applied to the pre-amplified signal.

Amp Designer Reverb Type Reference Table

You can choose a reverb type by clicking the Reverb label in the center of the Amp section. The table below covers the properties of each reverb type available in Amp Designer.

Reverb type	Description
Vintage Spring	This bright, splashy sound has largely defined combo amp reverb since the early 1960s.
Simple Spring	A darker, subtler spring sound.
Mellow Spring	An even darker, somewhat low-fidelity spring sound.
Bright Spring	Has some of the brilliance of Vintage Spring, but with less surf-style splash.
Dark Spring	A moody-sounding spring. More restrained than Mellow Spring.
Resonant Spring	Another 1960s-style spring with a strong, slightly distorted mid-range emphasis.
Boutique Spring	A modernized version of the classic Vintage Spring with a richer tone in the bass and mids.
Sweet Reverb	A smooth modern reverb with rich lows and restrained highs.
Rich Reverb	A bold, well-balanced modern reverb.
Warm Reverb	A lush modern reverb with rich low-mids and understated highs.

Using Amp Designer's Tremolo and Vibrato Effects

Tremolo and vibrato are controlled by several switches and two knobs in the Effects section found toward the right of the knobs section. Tremolo modulates the amplitude or volume of the sound, and vibrato modulates the pitch.

- *On/Off switch:* Enables or disables the tremolo or vibrato effect.
- *Trem/Vib switch:* Choose either tremolo or vibrato.
- *Depth knob:* Sets the intensity of the modulation (tremolo or vibrato).
- *Speed knob:* Sets the speed of the modulation in Hertz. Lower settings produce a smooth, floating sound. Higher settings produce a rotor-like effect.

- *Sync/Free switch:* When the switch is set to Sync, the modulation speed is synchronized with the host application tempo. The Speed knob lets you select different bar, beat, and musical note values (1/8, 1/16, and so on, including triplet and dotted-note values). When the switch is set to Free, the modulation speed can be set to any available value with the Speed knob.

Setting Amp Designer Microphone Parameters

Amp Designer offers a choice between three different virtual microphones. As with every other component in the tone chain, different selections yield very different results. After choosing a cabinet, you can set the type of microphone you want to be emulated, and where the microphone is placed in relation to the cabinet. The Mic pop-up menu is available near the right end of the black bar at the bottom, and the speaker-adjustment graphic appears when you move your mouse to the area above the Mic pop-up menu.

Note: The parameters described in this section are accessible only in the full Amp Designer interface. If you are in the small interface, click the disclosure triangle to the right of the Output field at the bottom-right edge of the interface to switch back to the full interface.

Move your mouse above the Mic pop-up menu to display the speaker-adjustment graphic.

- *Cabinet and speaker-adjustment graphic:* By default, the microphone is placed in the center of the speaker cone (on-axis). This placement produces a fuller, more powerful sound, suitable for blues or jazz guitar tones. If you place the microphone on the rim of the speaker (off-axis), you obtain a brighter, thinner tone, making it suitable for cutting rock or R & B guitar parts. Moving the microphone closer to the speaker emphasizes bass response.

  The microphone position is shown on the cabinet and indicated by the white dot in the speaker-adjustment graphic. Drag the white dot to change the microphone position and distance, relative to the cabinet. Placement is limited to near-field positioning.

- *Mic pop-up menu:* You can choose one of the Microphone models from the pop-up menu:
  - *Condenser:* Emulates the sound of a high-end German studio condenser microphone. The sound of condenser microphones is fine, transparent, and well-balanced.
  - *Dynamic:* Emulates the sound of popular American dynamic cardioid microphones. This microphone type sounds brighter and more cutting than the Condenser model. The mid-range is boosted, with lower-mid frequencies being less pronounced, making it a good choice for miking rock guitar tones. It is especially useful if you want your guitar part to cut through other tracks in a mix.
  - *Ribbon:* Emulates the sound of a ribbon microphone. A ribbon microphone is a type of dynamic microphone that captures a sound often described as bright or brittle, yet still warm. It is useful for rock, crunch, and clean tones.

  Tip: Combining multiple microphone types can produce an interesting sound. Duplicate the guitar track, and insert Amp Designer on both tracks. Select different microphones in each Amp Designer instance while retaining identical settings for all other parameters, and set track signal levels to taste.

Setting Amp Designer's Output Level

The Output slider (or the Output field, in the small interface) is found at the lower-right corner of the Amp Designer interface. It serves as the final level control for Amp Designer and can be thought of as a "behind the speaker" volume control that sets the level of the output that is fed to the ensuing Insert slots in the channel strip, or directly to the channel strip output.

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

Bass Amp

Bass Amp simulates the sound of several famous bass amplifiers. You can route bass guitar and other signals directly through the Bass Amp, reproducing the sound of your musical part played through a number of high-quality bass guitar amplification systems.

Bass Amp offers the following parameters.

- *Model pop-up menu:* Includes the following amplifier models:
  - *American Basic:* 1970s-era American bass amp, equipped with eight 10" speakers. Well-suited for blues and rock recordings.
  - *American Deep:* Based on the American Basic amp, but with strong lower-mid frequency (from 500 Hz on) emphasis. Well-suited for reggae and pop recordings.
  - *American Scoop:* Based on the American Basic amp, but combines the frequency characteristics of the American Deep and American Bright, with both low-mid (from 500 Hz) and upper-mid (from 4.5 kHz) frequencies emphasized. Well-suited for funk and fusion recordings.
  - *American Bright:* Based on the American Basic amp, this model emphasizes the upper-mid frequencies (from 4.5 kHz upward).
  - *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 strongly emphasizes the frequency range above 2 kHz. Well-suited for rock and heavy metal.

- *Top Class DI Warm:* Famous DI box simulation, well-suited for reggae and pop recordings. Mid frequencies, in the range between 500 and 5000 Hz, are de-emphasized.
- *Top Class DI Deep:* Based on the Top Class DI Warm, this model is well-suited for funk and fusion. The mid frequency range is strongest around 700 Hz.
- *Top Class DI Mid:* Based on the Top Class DI Warm, this model features an almost linear frequency range, with no frequencies emphasized. It is suitable for 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 Freq slider:* Sets the center frequency of the mid band (between 200 Hz and 3000 Hz).
- *Output Level slider:* Sets the final output level for Bass Amp.

Guitar Amp Pro

Guitar Amp Pro can simulate the sound of popular guitar amplifiers and the speakers used with them. You can process guitar signals directly, which enables you to reproduce the sound of your guitar through a number of high-quality guitar amplification systems.

Guitar Amp Pro can also be used for experimental sound design and processing. You can freely use it with other instruments, applying the sonic character of a guitar amp to a trumpet or vocal part, for example.

The amplifier, speaker, and EQ models emulated by Guitar Amp Pro can be combined in a number of ways to radically or subtly alter the tone. Virtual microphones are used to pick up the signal of the emulated amplifier and cabinet. You can choose from two different microphone types, and you can reposition them. Guitar Amp Pro also emulates classic guitar amplifier effects, including reverb, vibrato, and tremolo.

The Guitar Amp Pro window is organized into sections according to different kinds of parameters.

- *Amp section:* The model parameters at the top are used to choose the type of amp, EQ model, and speaker. See Building Your Guitar Amp Pro Model.

  Farther down in the Amp section, the knobs in the V-shaped formation are used to set tone, gain, and level. See Using Guitar Amp Pro's Gain, Tone, Presence, and Master Controls.
- *Effects section:* Provides parameters to control the built-in tremolo, vibrato, and reverb effects. See Using Guitar Amp Pro's Reverb Effect and Using Guitar Amp Pro's Tremolo and Vibrato Effects.
- *Microphone Position and Type sections:* These sections enable you to set the position and type of the microphone. See Setting Guitar Amp Pro Microphone Parameters.

Building Your Guitar Amp Pro Model

An amplifier "model" consists of an amplifier, speaker cabinet, EQ type, and microphone type. You can create your own hybrids of different amplifiers, cabinets, and so on—using the pop-up menus at the top center of the interface. You choose the microphone position and type in the yellow areas to the left and right.

You can use the Settings menu to save your new hybrid amp combos as setting files, which also include any parameter changes you may have made.

How to build your amplifier model is described in the following sections:

- Choosing a Guitar Amp Pro Amplifier
- Choosing a Guitar Amp Pro Speaker Cabinet

- Choosing a Guitar Amp Pro Equalizer
- Setting Guitar Amp Pro Microphone Parameters

Choosing a Guitar Amp Pro Amplifier

You can choose an amplifier model from the Amp pop-up menu near the top of the interface.

- *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 mid-frequency range, making this model ideal for solo sounds.
- *US Hot Top 100W:* This amp produces very fat sounds, even at low Master settings, that 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 (GarageBand):* 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 (GarageBand):* 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 (GarageBand):* Emulates the traditional full tube combos used for clean and crunchy sounds.
- *American Gain (GarageBand):* 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).

Choosing a Guitar Amp Pro Speaker Cabinet

The speaker cabinet can have a huge bearing on the type of tones you can extract from your chosen amplifier. The speaker parameters are found near the top of the interface.

- *Speaker pop-up menu:* You can choose one of the 15 speaker models:
  - *UK 1 x 12 open back:* Classic open enclosure with one 12" speaker, neutral, well-balanced, multifunctional.
  - *UK 2 x 12 open back:* Classic open enclosure with two 12" speakers, neutral, well-balanced, multifunctional.
  - *UK 2 x 12 closed:* Loads of resonance in the low frequency range, therefore well-suited for Combos: crunchy sounds are also possible with low Bass control settings.

- *UK 4 x 12 closed slanted:* when used in combination with off-center miking, you will get an interesting mid frequency range; therefore, this model works well when combined with High Gain amps.
- *US 1 x 10 open back:* Not much resonance in the low frequency range. Suitable for use with blues harmonicas.
- *US 1 x 12 open back 1:* Open enclosure of an American lead combo with a single 12" speaker.
- *US 1 x 12 open back 2:* Open enclosure of an American clean/crunch combo with a single 12" speaker.
- *US 1 x 12 open back 3:* Open enclosure of another American clean/crunch combo with a single 12" speaker.
- *US broad range:* Simulation of a classic electric piano speaker.
- *Analog simulation:* Internal speaker simulation of a well-known British tube preamplifier.
- *UK 1 x 12 (GarageBand):* A British Class A tube open back with a single 12" speaker.
- *UK 4 x 12 (GarageBand):* Classic closed enclosure with four 12" speakers (black series), suitable for rock.
- *US 1 x 12 open back (GarageBand):* Open enclosure of an American lead combo with a single 12" speaker.
- *US 1 x 12 bass reflex (GarageBand):* Closed bass reflex cabinet with a single 12" speaker.
- *DI Box:* This option allows you to bypass the speaker simulation section.

- *Amp-Speaker Link button:* Located between the Amp and Speaker pop-up menus, links these pop-up menus so that when you change the amp model, the speaker associated with that amp is loaded automatically.

Choosing a Guitar Amp Pro Equalizer

The EQ pop-up menu and the Amp-EQ Link button are near the top of the interface.

- *EQ pop-up menu:* Contains the following EQ models: British1, British2, American, and Modern. Each EQ model has unique tonal qualities that affect the way the Bass, Mids, and Treble knobs in the Amp section respond.
- *Amp-EQ Link button:* Located between the Amp and EQ pop-up menus, links these pop-up 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. The default combinations of amp, speaker, and EQ settings recreate a well-known guitar sound. You are, of course, free to combine any speaker or EQ model with any amp by turning off the two Link buttons.

Using Guitar Amp Pro's Gain, Tone, Presence, and Master Controls

The Gain, Bass, Mids, Treble, Presence, and Master knobs run from left to right in the V-shaped formation in the upper half of the interface.

- *Gain knob:* Sets the amount of pre-amplification applied to the input signal. This control has different effects, depending on which Amp model is chosen. For example, when you are using the British Clean amp model, the maximum Gain setting produces a powerful crunch sound. If you use the British Gain or Modern Gain amps, the same Gain setting produces heavy distortion, suitable for lead solos.
- *Bass, Mids, and Treble knobs:* Adjust the frequency range levels of the EQ models, similar to the tone knobs on a hardware guitar amplifier.
- *Presence knob:* Adjusts the high frequency range level. 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. For tube amplifiers, increasing the Master level typically produces a more compressed and saturated sound, resulting in a more distorted and powerful—that is, louder—signal. High Master settings can produce an extremely loud output that can damage your speakers or hearing, so ramp this up slowly. In Guitar Amp Pro, the Master parameter modifies the sonic character, and the final output level is set using the Output parameter at the bottom of the interface. See Setting the Guitar Amp Pro Output Level.

Getting to Know Guitar Amp Pro's Effects Section

The effects parameters include Tremolo, Vibrato, and Reverb, which emulate the processors found on many amplifiers.

You can use the pop-up menu to choose either Tremolo, which modulates the amplitude or volume of the sound, or Vibrato, which modulates the pitch.

Reverb can be added to either of these effects, or used independently.

To use or adjust an effect, you must first enable it by clicking the corresponding On button to the left. The On button is red when active.

Note: The Effects section is placed *before* the Presence and Master controls in the signal flow, and receives the preamplified, pre-Master signal.

Tremolo, Vibrato, and Reverb are described in the following sections:

- Using Guitar Amp Pro's Tremolo and Vibrato Effects
- Using Guitar Amp Pro's Reverb Effect

Using Guitar Amp Pro's Tremolo and Vibrato Effects

Tremolo and vibrato are controlled by an On button, the FX pop-up menu, the Depth and Speed knobs, and the Sync button in the Effects section. Tremolo modulates the amplitude or volume of the sound, and vibrato modulates the pitch.

- *FX pop-up menu:* You can choose either Tremolo or Vibrato.
- *Depth knob:* Sets the intensity of the modulation.
- *Speed knob:* Sets the speed of the modulation in Hertz. Lower settings produce a smooth and floating sound, while higher settings produce a rotor-like effect.
- *Sync button:* When the Sync button is turned on, the modulation speed is synchronized to the project tempo. You can adjust the Speed knob to select bar, beat, and musical note values (including triplet and dotted notes). When the Sync button is turned off, the modulation speed can be set to any available value with the Speed knob.

Using Guitar Amp Pro's Reverb Effect

Reverb is controlled by an On button, the Reverb pop-up menu, and a Level knob in the Reverb section near the bottom. Reverb can be added to either the Tremolo or Vibrato effect, or used independently.

- *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.

Setting Guitar Amp Pro Microphone Parameters

After choosing a speaker cabinet from the Speaker menu, you can set the type of microphone you want to be emulated, and where the microphone is placed in relation to the speaker. The Microphone Position parameters are available in the yellow area to the left, and the Microphone Type parameters in the yellow area to the right.

Microphone Position Parameters

- *Centered button:* 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:* Places the microphone on the edge of the speaker, also referred to as *off-axis*. This placement produces a tone that is brighter and sharper, but also thinner—suitable for cutting rock or R & B guitar parts.

  When you select either button, the graphic speaker display reflects your choice.

Microphone Type Parameters

- *Condenser button:* Emulates the sound of a studio condenser microphone. The sound of condenser microphones is fine, transparent, and well-balanced.

- *Dynamic button:* Emulates the sound of a dynamic cardioid microphone. This microphone type sounds brighter and more cutting than the Condenser model. At the same time, the lower-mid frequency range is less pronounced, making this model more suitable for miking rock guitar tones.

  Tip: Combining both microphone types can sound quite interesting. Duplicate the guitar track, and insert Guitar Amp Pro as an insert effect on both tracks. Select different microphone types in each Guitar Amp Pro instance, while retaining identical settings for all other parameters, and mix the track signal levels. You can, of course, choose to vary any other parameters.

Setting the Guitar Amp Pro Output Level

The Output slider is found at the bottom, below the Effects section. It serves as the final level control for Guitar Amp Pro and can be thought of as a "behind the speaker" volume control that is used to set the level fed to the ensuing plug-in slots on the channel strip or to Output channel strips.

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

Pedalboard

The Pedalboard simulates the sound of a number of well-loved and famous "stompbox" pedal effects. You can process any audio signal with a combination of stompboxes.

You can add, remove, and reorder pedals. The signal flow runs from left to right in the Pedal area. The addition of two discrete busses, coupled with splitter and mixer units, enables you to experiment with sound design and precisely control the signal at any point in the signal chain.

All stompbox knobs, switches, and sliders can be automated. Eight Macro controls enable real time changes to any pedal parameter with a MIDI controller.

- The Pedal Browser shows all pedal effects and utilities. These can be dragged into the Pedal area as part of the signal chain. See Using Pedalboard's Pedal Browser. This interface area is also used for the alternative import mode. See Using Pedalboard's Import Mode.
- The Pedal area is where you determine the order of effects and set effect parameters. You can add, replace, and remove stompboxes here. See Using Pedalboard's Pedal Area.
- The Routing area is used to control signal flow in the two effects busses (Bus A and Bus B) available in Pedalboard. See Using Pedalboard's Routing Area.
- The Macro Controls area is used to assign eight MIDI controllers, which can be used to control any stompbox parameter in real time. See Using Pedalboard's Macro Controls Area.
- The effect and utility pedals are described in the following sections:
  - Distortion Pedals
  - Modulation Pedals
  - Delay Pedals
  - Filter Pedals
  - Dynamics Pedals
  - Utility Pedals

Using Pedalboard's Pedal Browser

Pedalboard offers dozens of pedal effects and utilities in the *Pedal Browser* on the right side of the interface. Each effect and utility is grouped into a category, such as distortion, modulation, and so on. For information about these types of stompboxes, see Distortion Pedals, Modulation Pedals, Delay Pedals, Filter Pedals, Dynamics Pedals, and Utility Pedals.

To hide or show the Pedal Browser

- Click the disclosure triangle in the lower-right corner of the Pedal area.

To show only specific pedal groups in the Pedal Browser

- Open the View pop-up menu and choose Distortion, Modulation, Delay, Filter, Dynamics, or Utility. The Pedal Browser shows only the stompboxes within the category you choose.

  To show all the pedal groups, choose Show All from the View pop-up menu.

To add a stompbox to the Pedal area

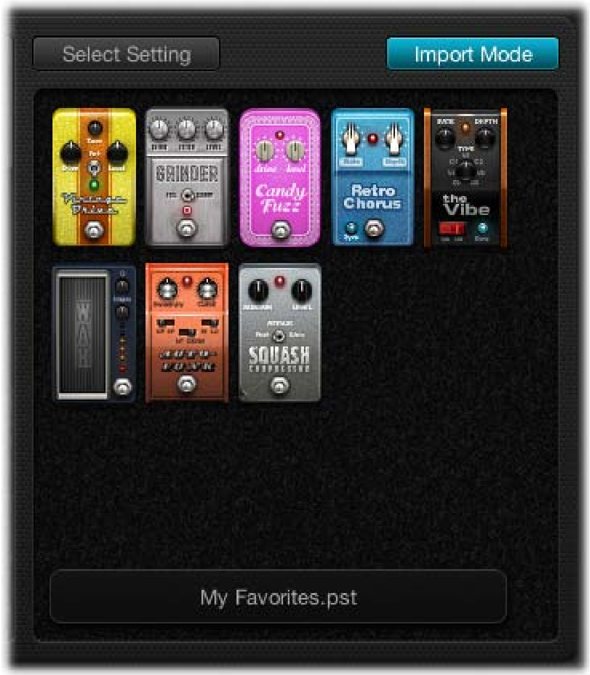
Do one of the following:

- Drag the effect that you want to insert from the Pedal Browser to the appropriate Pedal area position. This can be to the left, to the right, or in-between existing pedals.
- Double-click an effect in the Pedal Browser to add it to the right of all existing stompboxes in the Pedal area.

Note: Double-clicking a stompbox in the Pedal Browser when a stompbox is selected in the Pedal area will replace the selected pedal.

Using Pedalboard's Import Mode

Pedalboard has a feature you can use to import parameter settings for each type of pedal. In contrast to the plug-in window Settings menu, which you use to load a setting for the entire Pedalboard plug-in, this feature can be used to load a setting for a specific stompbox type.



To activate or deactivate import mode

- Click the Import Mode button to show all pedals used in the most recent Pedalboard setting. When the Import Mode button is active, the Pedal Browser switches to an alternate view mode that displays imported settings. When import mode is inactive, the normal Pedal Browser view is shown.

To import pedal settings into the Pedal Browser

1. Click the Import Mode button to activate import mode. Note that the View menu changes to the Select Setting button.

   Note: If this is your first attempt to import settings, a dialog opens where you can select a setting to import.

2 Click the Select Setting button and select a setting, then click Open. Dependent on the chosen setting, one or more stompboxes appear in the Pedal Browser. The name of the imported setting is shown at the bottom of the Pedal Browser.

To add an imported pedal to the Pedal area

Do one of the following:

- Drag the stompbox that you want to add from the Pedal Browser to the appropriate Pedal area position. This can be to the left, to the right, or in-between existing pedals.
- Ensure that no pedal is selected in the Pedal area, then double-click a stompbox in the Pedal Browser to add it to the right of all existing effects in the Pedal area.

Note: The parameter settings of pedals added in import mode are also imported.

To replace a pedal setting in the Pedal area with an imported pedal setting

1 Click the pedal you want to replace in the Pedal area. It becomes highlighted with a blue outline.

2 Click the stompbox in the Pedal Browser to replace the selected pedal (or pedal setting) in the Pedal area. The blue outlines of the selected pedal in the Pedal area and Pedal Browser blink on and off to indicate an imported setting. The setting name area at the bottom of the Pedal Browser displays "Click selected item again to revert."

Note: If you want to make your replacement permanent, click the background in the Pedal Browser, or click the Import Mode button.

3 To restore the selected pedal's previous setting, click the highlighted stompbox in the Pedal Browser. The Import Mode button and the outline of the selected pedal (in the Pedal area) become solidly highlighted, indicating that the original setting has been restored.

Using Pedalboard's Pedal Area

Pedalboard's stompbox effect pedals not only resemble their physical counterparts; they are also used in much the same way—without the inconvenience of patch cords, power supplies, and screws or locking mechanisms. The Pedal area layout mirrors a traditional pedalboard, with signals running from left to right.

To add a pedal to the Pedal area

Do one of the following:

- Drag the stompbox that you want to insert from the Pedal Browser to the appropriate Pedal area position. This can be to the left, to the right, or in-between existing pedals.
- Ensure that no pedal is selected in the Pedal area, then double-click a stompbox in the Pedal Browser to add it to the right of all existing effects in the Pedal area.

Note: You insert Mixer and Splitter utility pedals in a different way. See Using Pedalboard's Routing Area.

To change an effect pedal position in the Pedal area

- Drag the stompbox to a new position, either to the right or the left. Automation and bus routings, if active, are moved with the effect pedal. For information about automation and bus routings, see Using Pedalboard's Routing Area.

Note: There are two exceptions to the bus routing rule: If the dragged pedal is the only pedal between a Splitter and Mixer utility, both utility pedals are automatically removed. If the second Bus ("B") is not active at the destination, the pedal is inserted into Bus A.

To change a Mixer utility position in the Pedal area

- Drag the Mixer utility to a new position, either to the left or the right.

When moved to the left: The "downmix" of Bus A and B will occur at the earlier insertion point. Relevant effect pedals are moved to the right and are inserted into Bus A.

When moved to the right: The "downmix" of Bus A and B will occur at the later insertion point. Relevant effect pedals are moved to the left and are inserted into Bus A.

Note: A Mixer pedal cannot be moved to a position directly after (or to the left of) a corresponding split point or Splitter utility.

To change a Splitter utility position in the Pedal area

- Drag the Splitter utility to a new position, either to the right or the left.

When moved to the left: The split between Bus A and B will occur at the earlier insertion point. Relevant effect pedals are moved to the right and are inserted into Bus A.

When moved to the right: The split between Bus A and B will occur at the later insertion point. Relevant effect pedals are moved to the left and are inserted into Bus A.

Note: A Splitter pedal cannot be moved to a position directly preceding (or to the right of) a corresponding Mixer utility.

To replace a pedal in the Pedal area

Do one of the following:

- Drag the stompbox from the Pedal Browser *directly over* the pedal you want to replace in the Pedal area.

- Click to select the stompbox you want to replace in the Pedal area, then double-click the appropriate pedal in the Pedal Browser.

  Note: You can only replace "effect" pedals, not the Mixer or Splitter utilities. Bus routings, if active, are not changed when an effect pedal is replaced.

To remove a pedal from the Pedal area

Do one of the following:

- Drag the pedal out of the Pedal area.
- Click the pedal to select it and press the Delete key.

Using Pedalboard's Routing Area

Pedalboard has two discrete signal buses—Bus A and Bus B—that are found in the Routing area above the Pedal area. These busses provide a great deal of flexibility when you are setting up signal processing chains. All stompboxes that you drag into the Pedal area are inserted into Bus A, by default.

Note: The Routing area appears when you move your pointer to a position immediately above the Pedal area, and it disappears when you move the pointer away. When you create a second bus routing, the Routing area remains open even when your pointer is not over it. You can close the Routing area by clicking the small latch button at the top, and then the Routing area will open or close automatically when you move your pointer over it.

To create a second bus routing

Do one of the following:

- Move your pointer immediately above the Pedal area to open the Routing area, and click the name of a stompbox in the Routing area. The pedal name moves upward, and the chosen stompbox is routed to Bus B. Two gray lines appear in the Routing area, which represent Bus A and Bus B. A Mixer utility pedal is automatically added to the end of the signal chain.
- Drag a Splitter utility pedal into the Pedal area when more than one pedal is inserted. This also inserts a Mixer at the end of the signal chain if one doesn't already exist.

To remove the second bus routing

Do one of the following:

- Remove the Mixer and Splitter utility pedals from the Pedal area.

- Remove all stompboxes from the Pedal area. This automatically removes an existing Mixer utility.

To remove an effect from the second bus

- Click the name of the pedal (or on either of the gray lines) in the Routing area.

Note: The removal of all effects from Bus B does not remove the second bus. The Mixer utility pedal remains in the Pedal area, even when a single stompbox (effect) is in the Pedal area. This allows parallel routing of wet and dry signals. Only when all pedal effects are removed from the Pedal area is the Mixer utility (and second bus) removed.

To determine the split point between busses

- When more than one bus is active, a number of dots appear along the "cables" (gray lines) in the Routing area. These represent the output (the *socket*) of the pedal to the lower left of the dot. Click the appropriate dot to determine where the split point—where the signal is routed between busses. A cable appears between the busses when you click a dot.

Note: You can not create a split point directly before, or after, the Mixer utility.

To switch between a Splitter utility and bus split point

- Double-click a bus split point dot in the Routing area to replace it with a Splitter utility. The Splitter utility is shown in the Pedal area.
- Double-click the Splitter label in the Routing area to replace the Splitter utility with a bus split point dot. The Splitter utility is removed from the Pedal area.

Notes on Splitter and Mixer Utility Use

Dragging a Splitter utility into the Pedal area automatically inserts a Mixer utility to the far right of all inserted pedals.

You cannot drag a Splitter utility to the far right of all inserted pedals, to directly after an inserted Splitter utility, to directly in front of an inserted Mixer utility, or to an empty space in the Pedal area.

Dragging a Mixer utility into the Pedal area automatically creates a split point at the earliest possible (the leftmost) point within the signal chain.

You cannot drag a Mixer utility to the first slot in the Pedal area, to between an inserted Splitter and Mixer utility combo, or directly to the right of an inserted Mixer utility.

Using Pedalboard's Macro Controls Area

Pedalboard provides eight Macro Targets—A through H—which are found in the Macro Controls area below the Pedal area. These enable you to map any parameter of an inserted stompbox as a Macro A–H target. You can save different mappings with each Pedalboard setting.

In Logic Express, you use a controller assignment or create a Workspace knob for "Macro A–H Value." MIDI hardware switches, sliders, or knobs can then be used to control the mapped Pedalboard Macro A–H target parameters in real time. See the *Logic Express User Manual* for details.

Click the triangle at the bottom left to hide or show the Macro Controls area.

- *Macro A–H Target pop-up menus:* Determine the parameter that you want to control with a MIDI controller.
- *Macro A–H Value sliders and fields:* Set, and display, the current value for the parameter chosen in the corresponding Macro Target pop-up menu.

To assign a Macro A–H Target

Do one of the following:

- Click any of the Macro A–H Target pop-up menus, and choose the parameter that you want to control.

  Each stompbox parameter is shown in the following way: *"Slot number—Pedal Name—Parameter"*. As examples: *"Slot 1—Blue Echo—Time"*, or *"Slot 2—Roswell Ringer—Feedback"*. The "slot" number refers to the pedal position, as they appear from left to right in the Pedal area.

- Choose the "-Auto assign-" item in any Macro A–H Target pop-up menu, then click the appropriate parameter in any inserted pedal.

  Note: The chosen parameter is displayed in the Macro A–H Target pop-up menu.

Distortion Pedals

This section describes the distortion effects pedals.

Stompbox	Description
Candy Fuzz	A bright, "nasty" distortion effect. Drive controls the input signal gain. Level sets the effect volume.
Double Dragon	A deluxe distortion effect. It offers independent level controls for input (Input) and output (Level). Drive controls the amount of saturation applied to the input signal. The Tone knob sets the cutoff frequency. The Squash knob sets the threshold for the internal compression circuit. Contour sets the amount of nonlinear distortion applied to the signal. Mix sets the ratio between the source and distorted signals. The Bright/Fat switch changes between two fixed high shelving filter frequencies. Blue and red LEDs indicate each switch position, respectively.

Stompbox	Description
Fuzz Machine	An American "fuzz" distortion effect. Fuzz controls the input gain. Overall output gain is set with Level. The Tone knob increases treble, while simultaneously rolling-off low frequencies, as you move it to higher values.
Grinder	Grinder is a lo-fi, dirty "metal" distortion. Grind sets the amount of drive applied to the input signal. Tone is controlled with the Filter knob, making the sound harsher and more crunchy at higher values. The Full/Scoop switch alternates between two fixed Gain/Q filter settings. At the Full position, filtering is less pronounced than at the Scoop position. Overall output level is controlled with the Level knob.
Happy Face Fuzz	A softer, full-sounding distortion effect. Fuzz sets the amount of saturation applied to the input signal. Volume sets the output level.
Hi-Drive	An overdrive effect that can emphasize high frequency content in the signal. Level controls the effect output. The Treble/Full switch sets a fixed shelving frequency, allowing either the treble portion or the full range input signal to be processed.
Monster Fuzz	A saturated, somewhat harsh distortion. Roar sets the amount of gain applied to the input signal. Growl sets the amount of saturation. Tone sets the overall color of the distortion. Higher Tone values increase the treble content of the signal, but there is a corresponding decrease in overall volume. Texture can smooth out or roughen up the distortion. Grain sets the amount of nonlinear distortion applied to the signal. The effect output is controlled with the Level knob.
Octafuzz	A fat fuzz effect, that can deliver a soft, saturated distortion. Fuzz controls the input gain. Level sets the ratio between the distorted and source signals. The Tone knob sets the cutoff frequency of the highpass filter.
Rawk! Distortion	A metal/hard rock distortion effect. Crunch sets the amount of saturation applied to the input signal. Output gain is set with Level. Tonal color is set with the Tone knob, making the sound brighter at higher values.
Vintage Drive	Overdrive effect that emulates the distortion produced by a field effect transistor (FET), which is commonly used in solid-state amplifiers. When saturated, FETs generate a warmer sounding distortion than bipolar transistors (such as those emulated by Grinder). Drive sets the saturation amount for the input signal. Tone sets the frequency for the high cut filter, resulting in a softer or harsher tone. The Fat switch, when at the top position, enhances lower frequency content in the signal. Level sets the overall output level of the effect.

Modulation Pedals

This section describes the modulation effects pedals.

Stompbox	Description
Heavenly Chorus	A rich, sweet-sounding chorus effect that can significantly thicken the sound. Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). Depth sets the strength of the effect. Feedback sends the output of the effect back in to the input, further thickening the sound, or leading to intermodulations. Delay sets the ratio between the original and effect signals. The upper Bright switch position applies a fixed frequency internal EQ to the signal. At the bottom position, the EQ is bypassed.
Phase Tripper	A simple phasing effect. Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). Depth sets the strength of the effect. Feedback determines the amount of the effect signal that is routed back into the input. This can change the tonal color, can make the sweeping effect more pronounced, or can do both.
Phaze 2	A very flexible dual-phaser effect. LFO 1 and LFO2 Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. Ceiling and Floor determine the frequency range that is swept. Order switches between different algorithms, with higher (even) numbers resulting in a heavier phasing effect. Odd order numbers result in more subtle comb-filtering effects. Feedback determines the amount of the effect signal that is routed back into the input. This can change the tonal color, can make the phasing effect more pronounced, or can do both. Tone works from the center position; turn it to the left to increase the amount of lowpass filtering, or turn it to the right to increase the amount of highpass filtering. Mix sets the level ratio between each phaser.
Retro Chorus	A subtle, vintage chorus effect. Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). Depth sets the strength of the effect.
Robo Flanger	Flexible flanging effect. Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). Depth sets the strength of the effect. Feedback determines the amount of the effect signal that is routed back into the input. This can change the tonal color, can make the flanging effect more pronounced, or can do both. The Manual knob sets a delay time between the source and effect signals. This can result in flanger-chorus effects, or in metallic-sounding modulations, particularly when used with high Feedback values.

Stompbox	Description
Roswell Ringer	A ring modulation effect that can make incoming audio sound metallic (or unrecognizable), can deliver tremolos, brighten up signals and more. The Freq knob sets the core filter cutoff frequency. Fine is a fine tuning knob for the filter frequency. The Lin/Exp switch determines if the frequency curve is linear (12 notes per octave) or exponential. FB (feedback) determines the amount of the effect signal that is routed back into the input. This can change the tonal color, can make the effect more pronounced, or can do both. Balance between the original and effect signals is set with the Mix knob. See Ringshifter for background information on ring modulation.
Roto Phase	A phaser effect that adds movement to, and alters the phase of, the signal. Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes) with the Rate knob. Intensity sets the strength of the effect. The Vintage/Modern switch activates a fixed-frequency internal EQ when switched to Vintage, and deactivates it when switched to Modern.
Spin Box	Emulation of a Leslie rotor speaker cabinet, commonly used with the Hammond B3 organ. Cabinet sets the type of speaker box. Fast Rate sets the maximum modulation speed (only applies when Fast button is active). Response determines the amount of time required for the rotor to reach its maximum and minimum speed. Drive increases the input gain, introducing distortion to the signal. The Bright switch activates a high shelving filter when turned on. The Slow, Brake and Fast buttons determine how the "speaker" behaves: Slow rotates the speaker slowly. Fast rotates the speaker quickly (up to the maximum speed determined by the Fast Rate knob). Brake stops the speaker rotation. See Rotor Cabinet Effect for background information on the Leslie effect.
Total Tremolo	A flexible tremolo effect (modulation of the signal level). Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). Depth sets the strength of the effect. Wave and Smooth work in combination to alter the waveform shape of the LFO. This enables you to create floating changes in level, or abrupt steps. Volume determines the output level of the effect. The 1/2 and 2 x Speed buttons immediately halve or double the current Rate value. *Hold down* the Speed Up and Slow Down buttons to gradually accelerate or reduce the current Rate value to the maximum or minimum possible values.
Trem-o-Tone	A tremolo effect (modulation of the signal level). Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). Depth sets the strength of the effect. Level sets the post-tremolo gain.

Stompbox	Description
the Vibe	A vibrato/chorus effect based on the Scanner Vibrato unit found in the Hammond B3 organ. You can choose from three vibrato (V1–3) or chorus (C1–3) variations with the Type knob. Rate sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). Depth sets the strength of the effect. See Scanner Vibrato Effect for background information on this effect.

Delay Pedals

This section describes the Delay effects pedals.

Stompbox	Description
Blue Echo	A delay effect. Time sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). The Repeats knob determines the number of delay repeats. Mix sets the balance between the delayed and source signals. The Tone Cut switch controls a fixed frequency internal filter circuit that allows more low (Lo) or high (Hi) frequency content to be heard. You can also disable this filter circuit by choosing Off.
Spring Box	A spring reverb pedal. Time sets the length of the reverberation to short, medium, or long values. Tone controls the cutoff frequency, making the effect brighter or darker. Style switches between algorithms, each with different characteristics. You can choose from: Boutique, Simple, Vintage, Bright, and Resonant. Mix sets the ratio between the source and effect signals.
Tru-Tape Delay	A vintage tape delay effect. The Norm/Reverse switch changes the delay playback direction. Reverse mode is indicated by a blue LED and Normal mode is indicated by a red LED. Hi Cut and Lo Cut activate a fixed frequency filter. Dirt sets the amount of input signal gain, which can introduce an overdriven, saturated quality. Flutter emulates speed fluctuations in the tape transport mechanism. Time sets the modulation speed and can run freely, or be synchronized with the host application tempo by enabling the Sync button. When synchronized, you can specify bar, beat and note values (including triplets and dotted notes). Feedback determines the amount of the effect signal that is routed back into the input. The buildup of repeating signals can be used creatively for dub-delay and other effects by adjusting Feedback in real time. Mix sets the balance between the source and effect signals.

Filter Pedals

This section describes the filter effects pedals.

Stompbox	Description
Auto-Funk	An auto-wah (filter) effect. Sensitivity sets a threshold which determines how the filter responds to incoming signal levels. Cutoff sets the center frequency for the filter. The BP/LP switch enables either a bandpass or lowpass filter circuit. Signal frequencies just above and below the cutoff point are filtered when the BP switch position is chosen. When the LP switch position is active, only signals below the cutoff point are allowed through the filter. The Hi/Lo switch chooses one of two preset (filter) resonance settings. The Up/Down switch activates a positive or negative modulation direction (the "wah" filtering occurs above or below the source signal frequency).
Classic Wah	A funky wah effect, straight from 1970's TV police show soundtracks. You control it by dragging the pedal.
Modern Wah	A more aggressive wah effect. You control it by dragging the pedal. Mode enables you to choose from the following: Retro Wah, Modern Wah, Opto Wah 1, Opto Wah 2, Volume. Each has a different tonal quality. The Q knob determines the resonant characteristics. Low Q values affect a wider frequency range, resulting in softer resonances. High Q values affect a narrower frequency range, resulting in more pronounced emphasis.

Dynamics Pedals

This section describes the dynamics pedals.

Stompbox	Description
Squash Compressor	A simple compressor. Sustain sets the threshold level. Signals above this are reduced in level. Level determines the output gain. The Attack switch can be set to Fast for signals with fast attack transients, such as drums, or to Slow for signals with slow attack phases, such as strings.

Utility Pedals

This section describes the parameters of the Mixer and Splitter pedals.

Stompbox	Description
Mixer	A utility that is used to control the level relationship between Bus A and Bus B signals. It can be inserted anywhere in the signal chain, but is typically used at the end of the chain (at the extreme right of the Pedal area). See Using Pedalboard's Routing Area for details on use. The A/Mix/B switch solos the "A" signal, mixes the "A" and "B" signals, or solos the "B" signal. The level setting of the Mix fader is relevant for all A/Mix/B switch positions. In stereo instances, the Mixer utility also provides discrete Pan controls for each bus.

Stompbox	Description
Splitter	A utility that can be inserted anywhere in the signal chain. Splitter can be used in two ways; When set to Freq, it works as a frequency-dependent signal splitter that divides the incoming signal. Signals *above* the frequency set with the Frequency knob are sent to Bus B. Signals *below* this frequency are sent to Bus A. When set to Split, the incoming signal is routed equally to both buses. The Frequency knob has no impact in this mode. See Using Pedalboard's Routing Area for details on use.

Delay Effects

2

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

The held, and delayed, signal is repeated after a given time period, creating a repeating echo effect. Each subsequent repeat is a little quieter than the previous one. Most delays also allow you to feed a percentage of the delayed signal back to the input. This can result in a subtle, chorus-like effect or cascading, chaotic audio output.

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 to double individual sounds to resemble a group of instruments playing the same melody, to create echo effects, to place the sound in a large "space", to generate rhythmic effects, or to enhance 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 an unusual effect.

This chapter covers the following:

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.

- *Time pop-up menu:* 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 slider and field:* Determines how often the delay effect is repeated.
- *Color slider and field:* Sets the harmonic content (color) of the delay signal.
- *Dry and Wet sliders and fields:* Control the amount of original and effect signal.

Sample Delay

Sample Delay is more a utility than an effect—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, Sample Delay is useful for correction of timing problems that may occur with multichannel microphones. It can also be used creatively, to emulate stereo microphone channel separation.

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

- *Delay slider and field (L and R in stereo version):* Determines the number of samples that the incoming signal will be delayed by.
- *Link L & R button (only in stereo version):* Ensures that the number of samples is identical for both channels. Adjusting one channel value will adjust the other.

Stereo Delay

The Stereo Delay works much like the Tape Delay (see Tape Delay), but allows you to set the Delay, Feedback, and Mix parameters separately for the left and right channels. The Crossfeed knob for each stereo side 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 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.

As the parameters for the left and right delays are identical, the descriptions below only cover the left channel—the right channel information is provided in brackets, if named differently. Parameters that are common to both channels are shown separately.

Channel Parameters

- *Left (Right) Input pop-up menu:* Choose the input signal for the two stereo sides. Options include OFF, Left, Right, L + R, L – R.
- *Left (Right) Delay field:* Sets the current delay time in milliseconds (this parameter is dimmed when you synchronize the delay time to the project tempo).
- *Groove slider and field:* Determines the proximity of every second delay repeat to the absolute grid position—in other words, how close every second delay repeat is.
- *Note buttons:* Set the grid resolution for the delay time. These are shown as note durations (these are dimmed when the delay time is not synchronized with the project tempo).
- *Left (Right) Feedback knob and field:* Set the amount of feedback for the left and right delay signals.
- *Crossfeed Left to Right (Crossfeed Right to Left) knob and field:* Transfer the feedback signal of the left channel to the right channel, and vice versa.
- *Feedback Phase button:* Use to invert the phase of the corresponding channel's feedback signal.
- *Crossfeed Phase button:* Use to invert the phase of the crossfed feedback signals.

Common Parameters

- *Beat Sync button:* Synchronizes delay repeats to the project tempo, including tempo changes.
- *Output Mix (Left and Right) sliders and fields:* Independently control the left and right channel signals.
- *Low Cut and High Cut sliders and fields:* Frequencies below the Low Cut value and above the High Cut value are filtered out of the source signal.

Tape Delay

Tape Delay simulates the warm sound of vintage tape echo machines, with the convenience of easy delay time synchronization to your project tempo. The effect is equipped with a highpass and lowpass filter in the feedback loop, simplifying the creation of authentic dub echo effects. Tape Delay also includes an LFO for delay time modulation, which can be used to produce pleasant or unusual chorus effects, even on long delays.

- *Feedback slider:* Determines the amount of delayed and filtered signal that is routed back to the input of the Tape Delay. Set the Feedback slider to the lowest possible value to generate a single echo. Turn Feedback all the way up to endlessly repeat the signal. The levels of the original signal and its taps (echo repeats) tend to accumulate, and may cause distortion. You can use the internal tape saturation circuit to ensure that these overdriven signals continue to sound good.
- *Freeze button:* Captures the current delay repeats and sustains them until the Freeze button is turned off.
- *Delay field:* Sets the current delay time in milliseconds (this parameter is dimmed when you synchronize the delay time to the project tempo).
- *Sync button:* Synchronizes delay repeats to the project tempo (including tempo changes).
- *Tempo field:* Sets the current delay time in beats per minute (this parameter is dimmed when you synchronize the delay time to the project tempo).

- *Groove slider and field:* Determines the proximity of every second delay repeat to the absolute grid position—in other words, how close every second delay repeat is. A Groove setting of 50% means that every delay has 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.
- *Note buttons:* Set the grid resolution for the delay time. These are shown as note durations.
- *Low Cut and High Cut sliders and fields:* Frequencies below the Low Cut value and above the High Cut value are filtered out of the source signal. You can shape the sound of the echoes with the highpass and lowpass filters. The filters are located in the feedback circuit, which means that the filtering effect increases in intensity with each delay repeat. If you want an increasingly muddy and confused tone, move the High Cut slider towards the left. For ever thinner echoes, move the Low Cut slider towards the right. 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 and Wet controls *and* the filter settings—move the High Cut slider to the far right, and the Low Cut slider to the far left.
- *Smooth slider and field:* Evens out the LFO and flutter effect.
- *LFO Rate knob and field:* Sets the frequency of the LFO.
- *LFO Depth knob and field:* Sets the amount of LFO modulation. A value of 0 turns delay modulation off.
- *Flutter Rate and Intensity sliders and fields:* Simulate the speed irregularities of the tape transports used in analog tape delay units.
  - *Flutter Rate:* Sets the speed variation.
  - *Flutter Intensity:* Determines how pronounced the effect is.
- *Dry and Wet sliders and fields:* Independently control the amount of original and effect signal.
- *Distortion Level slider and field (Extended Parameters area):* Determines the level of the distorted (tape saturation) signal.

Distortion Effects

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 they are still used in musical instrument amplifiers today. When overdriven, they produce a type of distortion that 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 that 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 to create sound effects.

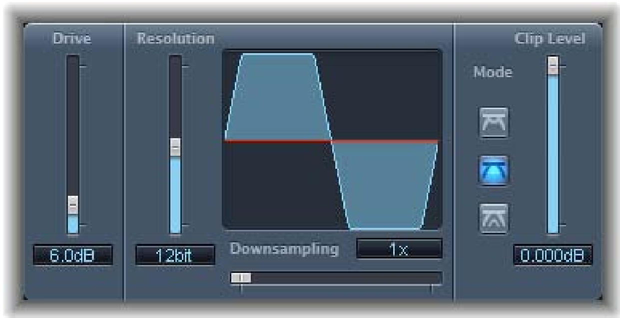
Distortion effects include parameters for *tone,* which let you shape the way the distortion alters the signal (often as a frequency-based filter), and for *gain,* which let you control how much the distortion alters the output level of the signal.

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

This chapter covers the following:

Bitcrusher

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



- *Drive slider and field:* Sets the amount of gain in decibels applied to the input signal.

  Note: Raising the Drive level tends to increase the amount of clipping at the output of the Bitcrusher as well.
- *Resolution slider and field:* Sets the bit rate (between 1 and 24 bits). This alters the calculation precision of the process. 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.
- *Waveform display:* Shows the impact of parameters on the distortion process.
- *Downsampling slider and field:* Reduces the sample rate. A value of 1 x leaves the signal unchanged, a value of 2 x halves the sample rate, and a value of 10 x reduces the sample rate to one-tenth of the original signal. (For example, if you set Downsampling to 10 x , a 44.1 kHz signal is sampled at just 4.41 kHz.)

  Note: Downsampling has no impact on the playback speed or pitch of the signal.
- *Mode buttons:* Set the distortion mode to Folded, Cut, or Displaced. Signal peaks that exceed the clip level are processed.

  Note: The Clip Level parameter has a significant impact on the behavior of all three modes. This is reflected in the Waveform display, so try each mode button and adjust the Clip Level slider to get a feel for how this works.
  - *Folded:* The start and end levels of the clipped signal are unchanged, but the center portion is effectively folded in half (halved in the level above the threshold), resulting in a softer distortion.
  - *Cut:* The signal is abruptly distorted when the clipping threshold is exceeded. Clipping that occurs in most digital systems is closest to Cut mode.

- *Displaced:* The start, center and end levels of the signal (above the threshold) are offset, resulting in a distortion which is less severe as signal levels cross the threshold. The center portion of the clipped signal is also softer than in Cut mode.
- *Clip Level slider and field:* Sets the point (below the clipping threshold of the channel strip) at which the signal starts clipping.
- *Mix slider and field (Extended Parameters area):* Sets the balance between dry (original) and wet (effect) signals.

Clip Distortion

Clip Distortion is a nonlinear distortion effect that produces unpredictable spectra. It can simulate warm, overdriven tube sounds and can also generate drastic distortions.

Clip Distortion features an unusual combination of serially connected filters. The incoming signal is amplified by the Drive value, passes through a highpass filter, and is then subjected to nonlinear distortion. Following the distortion, the signal passes through a lowpass filter. The effect signal is then recombined with the original signal and this mixed signal is sent through a further 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.

- *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.
- *Clip Circuit display:* Shows the impact of all parameters, with the exception of the High Shelving filter parameters.
- *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.

- *Mix slider and field:* Sets the ratio between the effect (wet) signal and original (dry) signals, following the Clip Filter.
- *Sum LPF knob and field:* Sets the cutoff frequency (in Hertz) of the lowpass filter. This processes the mixed signal.
- *(High Shelving) Frequency knob and field:* Sets the frequency (in Hertz) of the high shelving filter. 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 these types of treble controls, however, you can boost or cut the signal by up to ±30 dB with the Gain parameter.
- *(High Shelving) Gain knob and field:* Sets the amount of gain applied to the output signal.
- *Input Gain field and slider (Extended Parameters area):* Sets the amount of gain applied to the input signal.
- *Output Gain field and slider (Extended Parameters area):* Sets the amount of gain applied to the output signal.

Distortion Effect

The 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.

- *Drive slider and field:* Sets the amount of saturation applied to the signal.
- *Display:* Shows the impact of parameters on the signal.
- *Tone knob and field:* Sets the frequency for the high cut filter. Filtering the harmonically rich distorted signal produces a softer tone.
- *Output slider and field:* Sets the output level. This allows you to compensate for increases in loudness caused by adding distortion.

Distortion II

Distortion II emulates the distortion circuit 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.

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

Overdrive

Overdrive 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, such as those emulated by the Distortion effect.

- *Drive slider and field:* Sets the saturation amount for the simulated transistor.

- *Display:* Shows the impact of parameters on the signal.
- *Tone knob and field:* Sets the frequency for the high cut filter. Filtering the harmonically rich distorted signal produces a softer tone.
- *Output slider and field:* Sets the output level. This allows you to compensate for increases in loudness caused by using Overdrive.

Phase Distortion

The Phase Distortion effect is based on a modulated delay line, similar to a chorus or flanger effect (see Modulation Effects). Unlike these effects, however, the delay time is not modulated by a low frequency oscillator (LFO), but rather by a lowpass-filtered version of the input signal itself, using an internal sidechain. This means that the incoming signal modulates its own phase position.

The input signal only passes the delay line and is not affected by any other process. The Mix parameter blends the effect signal with the original signal.

- *Monitor button:* Enable to hear the input signal in isolation. Disable to hear the mixed signal.
- *Cutoff knob and field:* Sets the (center) cutoff frequency of the lowpass filter.
- *Resonance knob and field:* Emphasizes frequencies surrounding the cutoff frequency.
- *Display:* Shows the impact of parameters on the signal.
- *Mix slider and field:* Adjusts the percentage of the effect 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 checkbox (Extended Parameters area):* Enable to reduce the delay time on the right channel when input signals that exceed the cutoff frequency are received. Available only for stereo instances of the Phase Distortion effect.

Dynamics Processors

4

The Dynamics processors 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 highest amplitudes. Dynamics processors enable you to adjust the dynamic range of individual audio files, tracks, or an overall project. This can be to increase the perceived loudness and/or to highlight the most important sounds, while ensuring that softer sounds are not lost in the mix.

This chapter covers the following:

Types of Dynamics Processors

There are four types of dynamics processors included in Logic Express. These are each used for different audio processing tasks.

- *Compressors:* Logic Express features a number of downward compressors. These behave like an automatic volume control, lowering the volume whenever it rises above a certain level, called the *threshold*. So, why would you want to reduce the dynamic level?

By reducing the highest parts of the signal, called *peaks*, a compressor raises the overall level of the signal, increasing the perceived volume. This gives the signal 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 make a project sound better when played back in different audio environments. For example, the speakers of a television set or in a car typically have a narrower dynamic range than the sound system in a cinema. 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 singing prominent in an overall mix. They are also commonly used on music and sound effect tracks, but they are rarely used on ambience tracks.

Some compressors—*multiband compressors*—can divide the incoming signal into different frequency bands and apply different compression settings to each band. This helps to achieve the maximum level without introducing compression artifacts. Multiband compression is typically used on an overall 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 add life to audio signals.
- *Limiters:* Limiters—also called *peak limiters*—work in a similar way to compressors in that they reduce the audio signal when it exceeds a set threshold. The difference is that whereas a compressor gradually lowers signal levels that exceed the threshold, a limiter quickly reduces any signal that is 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 a way that is opposite to that used by compressors or limiters. Whereas a compressor lowers the level when the signal is louder than the threshold, a noise gate lowers the signal level whenever it falls 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 are often used to eliminate low-level noise or hum from an audio signal.

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 mix into focus, while preventing softer parts from becoming inaudible. It is probably the most versatile and widely used sound-shaping tool 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. Usually you insert the Compressor directly into a channel strip.

Compressor Parameters

The Compressor offers the following parameters:

- *Circuit Type pop-up menu:* Choose the type of circuit emulated by the Compressor. The choices are Platinum, Class(ic) A_R, Class(ic) A_U, VCA, FET, and Opto (optical).
- *Side Chain Detection pop-up menu:* Determines if the Compressor uses the maximum level of each side-chained signal (Max) or the summed level of all side-chained signals (Sum) to exceed or fall below the threshold.
  - If either of the stereo channels exceeds or falls below the Threshold, both channels are compressed.
  - If Sum is chosen, the combined level of both channels must exceed the Threshold before compression occurs.
- *Gain Reduction meter:* Shows the amount of compression in real time.
- *Attack knob and field:* Determines the amount of time it takes for the compressor to react when the signal exceeds the threshold.

- *Compression curve display:* Shows the compression curve created by the combination of Ratio and Knee parameter values. Input (level) is shown on the x-axis and output (level) on the y-axis.
- *Release knob and field:* Determines the amount of time it takes for the compressor to stop reducing the signal after the signal level falls below the threshold.
- *Auto button:* When the Auto button is active, the release time dynamically adjusts to the audio material.
- *Ratio slider and field:* Sets the compression ratio—the ratio of signal reduction when the threshold is exceeded.
- *Knee slider and field:* Determines the strength of compression at levels close to the threshold. Lower values result in more severe/immediate compression (hard knee). Higher values result in gentler compression (soft knee).
- *Compressor Threshold slider and field:* Sets the threshold level—signals above this threshold value are reduced in level.
- *Peak/RMS buttons:* Determines whether signal analysis is with the Peak or RMS method, when using the Platinum circuit type.
- *Gain slider and field:* Sets the amount of gain applied to the output signal.
- *Auto Gain pop-up menu:* Choose a value to compensate for volume reductions 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.
- *Output Distortion pop-up menu (Extended Parameters area):* Choose whether to apply clipping above 0 dB, and the type of clipping. Choices are: Off, Soft, Hard, and Clip.
- *Activity pop-up menu (Extended Parameters area):* Enables or disables the side chain. Choices are: Off, Listen, and On.
- *Mode pop-up menu (Extended Parameters area):* Choose the type of filter used for the side chain. Choices are: LP (lowpass), BP (bandpass), HP (highpass), ParEQ (parametric), and HS (high shelving).
- *Frequency slider and field (Extended Parameters area):* Sets the center frequency for the side-chain filter.
- *Q slider and field (Extended Parameters area):* Sets the width of the frequency band affected by the side-chain filter.
- *Gain slider and field (Extended Parameters area):* Sets the amount of gain applied to the side-chain signal.
- *Mix slider and field (Extended Parameters area):* Determines the balance between dry (source) and wet (effect) signals.

Using the Compressor

The following section explains how to use the main Compressor parameters.

Setting the Compressor Threshold and Ratio

The most important Compressor parameters are Threshold and Ratio. The *Threshold* sets the floor level in decibels. Signals that exceed this level are reduced by the amount set as the Ratio.

The *Ratio* is a percentage of the overall level; the more the signal exceeds the threshold, the more it is reduced. A ratio of 4:1 means that increasing the input by 4 dB results in an increase of the output by 1 dB, if above the threshold.

As an example, with the Threshold set at −20 dB and the Ratio set to 4:1, a −16 dB peak in the signal (4 dB louder than the threshold) is reduced by 3 dB, resulting in an output level of −19 dB.

Setting Suitable Compressor Envelope Times

The Attack and Release parameters shape the dynamic response of the Compressor. The Attack parameter determines the time it takes after the signal exceeds the threshold level before the Compressor starts reducing the signal.

Many sounds, including voices and musical instruments, rely on the initial attack phase to define the core timbre and characteristic of the sound. When compressing these types of sounds, you should set higher Attack values to ensure that the attack transients of the source signal aren't lost or altered.

When attempting to maximize the level of an overall mix, it is best to set the Attack parameter to a lower value, because higher values often result in no, or minimal, compression.

The Release parameter determines how quickly the signal is restored to its original level after it falls below the threshold level. Set a higher Release value to smooth out dynamic differences in the signal. Set lower Release values if you want to emphasize dynamic differences.

Important: The discussion above is highly reliant on not only the type of source material, but also the compression ratio and threshold settings.

Setting the Compressor Knee

The Knee parameter determines whether the signal is slightly, or severely, compressed as it approaches the threshold level.

Setting a Knee value close to 0 (zero) results in no compression of signal levels that fall just below the threshold, while levels at the threshold are compressed by the full Ratio amount. This is known as *hard knee compression*, which can cause abrupt and often unwanted transitions as the signal reaches the threshold.

Increasing the Knee parameter value increases the amount of compression as the signal approaches the threshold, creating a smoother transition. This is called *soft knee compression*.

Setting Other Compressor Parameters

As the compressor reduces levels, the overall volume at its output is typically lower than the input signal. You can adjust the output level with the Gain slider.

You can also use the Auto Gain parameter to compensate for the level reduction caused by compression (choose either −12 dB or 0 dB).

When you use the Platinum circuit type, the Compressor can analyze the signal using one of two methods: Peak or root mean square (RMS). While Peak is more technically accurate, RMS provides a better indication of how people perceive the signal loudness.

Note: If you activate Auto Gain and RMS simultaneously, the signal may become over-saturated. If you hear any distortion, switch Auto Gain off and adjust the Gain slider until the distortion is inaudible.

Using a Side Chain with the Compressor

Use of a side chain with a compressor is common. This allows you to use the dynamics (level changes) of another channel strip as a control source for compression. For example, the dynamics of a drum groove can be used to rhythmically change the compression, and therefore dynamics, of a guitar part.

Important: The side-chain signal is used only as a detector or trigger in this situation. The side-chain source is used to control the Compressor, but the audio of the side-chain signal is not actually routed through the Compressor.

To use a side chain with the Compressor

1. Insert the Compressor into a channel strip.
2. Select the channel strip that carries the desired signal (side-chain source) in the Side Chain menu of the Compressor plug-in.
3. Choose the desired analysis method (Max or Sum) from the Side Chain Detection pop-up menu.
4. Adjust the Compressor parameters.

DeEsser

The DeEsser is a frequency-specific compressor, designed to compress 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 rather than an EQ 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 has 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 can be easily compared in the DeEsser's Detector and Suppressor frequency range displays

The Suppressor frequency range is reduced in level for as long as the Detector frequency threshold is exceeded.

The DeEsser does not use a frequency-dividing network—a crossover utilizing lowpass and highpass filters. Rather, it isolates and subtracts the frequency band, resulting in no alteration of the phase curve.

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.

DeEsser Detector Section

- *Detector Frequency knob and field:* Sets the frequency range for analysis.
- *Detector Sensitivity knob and field:* Sets the degree of responsiveness to the input signal.
- *Monitor pop-up menu:* Choose Det(ector) to monitor the isolated Detector signal, Sup(pressor) to monitor the filtered Suppressor signal, Sens(itivity) to remove the sound from the input signal in response to the Sensitivity parameter, or Off to hear the DeEsser output.

DeEsser Suppressor Section

- *Suppressor Frequency knob and field:* Sets the frequency band that is reduced when the Detector sensitivity threshold is exceeded.
- *Strength knob and field:* Sets the amount of gain reduction for signals that surround the Suppressor frequency.
- *Activity LED:* Indicates active suppression in real time.

DeEsser Center Section

- *Detector and Suppressor frequency displays:* The upper display shows the Detector frequency range. 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 times, as they are used by compressors.

Ducker

Ducking is a common technique used in radio and television broadcasting: When the DJ or 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.

Ducker provides a simple means of achieving this result with existing recordings. It does not work in real time.

Note: For technical reasons, Ducker can only be inserted in output and aux channel strips.

Ducker Parameters

The Ducker has the following parameters:

- *Ducking On and Off buttons:* Enable or disable ducking.
- *Lookahead On and Off buttons:* Enable to ensure that the Ducker reads the incoming signal before processing. This results in no latency—it is primarily intended for slower computers.
- *Amount slider and field:* Defines the amount of volume reduction of the music mix channel strip, which is, in effect, the output signal.
- *Threshold slider and field:* 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 channel strip volume is not affected.
- *Attack slider and field:* 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 signal level is reduced before the threshold is reached. The earlier this occurs, the more latency is introduced.

  Note: This only works if the ducking signal is not live—the ducking signal must be an existing recording. The host application needs to analyze the signal level before it is played back in order to predefine the point where ducking begins.

- *Hold slider and field:* Determines the duration for which the music mix channel strip volume is reduced. This control prevents a chattering effect that can be caused by a rapidly changing side-chain level. If the side-chain 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 any rapid volume reductions.
- *Release slider and field:* Controls how quickly the volume returns to the original level. Set it to a high value if you want the music mix to slowly fade up after the announcement.

Using the Ducker

The steps below show how to use the Ducker on existing recordings.

Note: For technical reasons, the Ducker plug-in can be inserted only in output and aux channel strips.

To use the Ducker plug-in

1. Insert the plug-in into an aux channel strip.
2. Assign all channel strip outputs that are supposed to "duck" (dynamically lower the volume of the mix) to a bus—the aux channel strip chosen in step 1.
3. Choose the bus 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 parameters.

Enveloper

The Enveloper is an unusual processor that lets you shape the attack and release phases of a signal—the signal's *transients*, in other words. This makes it a unique tool that can be used to achieve results that differ from other dynamic processors.

- *Threshold slider and field:* Sets the threshold level. Signals that exceed the threshold have their attack and release phase levels altered.
- *(Attack) Gain slider and field:* Boosts or attenuates the attack phase of the signal. When the Gain slider is set to the center position—0%—the signal is unaffected.
- *Lookahead slider and field:* Sets the pre-read analysis time for the incoming signal. This enables the Enveloper to know in advance what signals are coming, enabling accurate and fast processing.
- *(Attack) Time knob and field:* Determines the amount of time it takes for the signal to increase from the threshold level to the maximum Gain level.
- *Display:* Shows the attack and release curves applied to the signal.
- *(Release) Time knob and field:* Determines the amount of time it takes for the signal to fall from the maximum gain level to the threshold level.
- *(Release) Gain slider and field:* Boosts or attenuates the release phase of the signal. When the Gain slider is set to the center position—0%—the signal is unaffected.
- *Out Level slider and field:* Sets the level of the output signal.

Using the Enveloper

The most important parameters of the Enveloper are the two Gain sliders, one on each side of the central display. These govern the Attack and Release levels of each respective phase.

Boosting the attack phase can add snap to a drum sound, or it can amplify the initial pluck or pick sound of a stringed instrument. Attenuating the attack causes percussive signals to fade in more softly. You can also mute the attack, making it virtually inaudible. A creative use for this effect is alteration of the attack transients to mask poor timing of recorded instrument parts.

Boosting the release phase also accentuates any reverb applied to the affected channel strip. Conversely, attenuating 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 adjusting the Out Level slider.

Generally, you'll find that Attack Time values of around 20 ms and Release Time values of 1500 ms are good to start with. Then adjust them for the type of signal that you're processing.

The Lookahead slider defines how far into the future of the incoming signal the Enveloper looks, in order to anticipate future events. You generally won't need to use this feature, except when processing signals with extremely sensitive transients. If you do raise the Lookahead value, you may need to adjust the Attack Time to compensate.

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

Expander

The Expander is similar in concept to a compressor, but increases, rather than reduces, the dynamic range above the threshold level. You can use the Expander to add liveliness and freshness to your audio signals.

- *Threshold slider and field:* Sets the threshold level. Signals above this level are expanded.
- *Peak/RMS buttons:* Determine whether the Peak or RMS method is used to analyze the signal.
- *Attack knob and field:* Determines the time it takes for the Expander to respond to signals that exceed the threshold level.
- *Expansion display:* Shows the expansion curve applied to the signal.
- *Release knob and field:* Sets the time it takes for the Expander to stop processing the signal after it falls below the threshold level.
- *Ratio slider and field:* Sets the expansion ratio—the ratio of signal expansion when the threshold is exceeded.

  Note: As the Expander is a genuine upward expander—in contrast to a downward expander, which increases the dynamic range below the Threshold—the *Ratio* slider features a value range of 1:1 to 0.5:1.
- *Knee slider and field:* Determines the strength of expansion at levels close to the threshold. Lower values result in more severe or immediate expansion—hard knee. Higher values result in a gentler expansion—soft knee.
- *Gain slider and field:* Sets the amount of output gain.
- *Auto Gain button:* Compensates for the level increase caused by expansion. When Auto Gain is active, the signal sounds softer, even when the peak level remains the same.

  Note: If you dramatically change the dynamics of a signal (with extreme Threshold and Ratio values), you may need to reduce the Gain slider level to avoid distortion. In most cases, turning on Auto Gain will adjust the signal appropriately.

Limiter

The Limiter works much like a compressor but 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 when mastering. Typically, you apply the Limiter as the very last process in the mastering signal chain, where it raises 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, the Limiter reduces the level before clipping can occur. The Limiter cannot, however, fix audio that is clipped during recording.

- *Gain reduction meter:* Shows the amount of limiting in real time.
- *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. This enables it to react earlier to peak volumes by adjusting the amount of reduction.

  Note: Lookahead causes latency, but this has no perceptible effect when you use the Limiter as a mastering effect on prerecorded material. Set it to higher values if you want the limiting effect to occur before the maximum level is reached, thus creating a smoother transition.
- *Release slider and field:* Sets the amount of time, after the signal falls below the threshold level, before the Limiter stops processing.
- *Output Level knob and field:* Sets the output level of the signal.
- *Softknee button:* When active, the signal is limited only when it reaches the threshold. The transition to full limiting is nonlinear, producing a softer, less abrupt effect, and reducing distortion artifacts that can be produced by hard limiting.

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 effectively removes lower-level parts of the signal, while allowing the desired parts of the audio to pass.

Noise Gate Parameters

The Noise Gate has the following parameters.

- *Threshold slider and field:* Sets the threshold level. Signals that fall below the threshold will be reduced in level.
- *Reduction slider and field:* Sets the amount of signal reduction.
- *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 reach maximum attenuation 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. This prevents the gate from rapidly opening and closing when the input signal is close to the threshold.
- *Lookahead slider and field:* Sets how far ahead the Noise Gate analyzes the incoming signal, allowing the effect to respond more quickly to peak levels.

- *Monitor button:* Enable to hear the side-chain signal, including the effect of the High Cut and Low Cut filters.
- *High Cut slider and field:* Sets the upper cutoff frequency for the side-chain signal.
- *Low Cut slider and field:* Sets the lower cutoff frequency for the side-chain signal.

  Note: When no external side chain is selected, the input signal is used as the side chain.

Using the Noise Gate

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

The Attack, Hold, and Release knobs modify the dynamic response of the Noise Gate. If you want the gate to open extremely quickly, for percussive signals such as drums, set the Attack knob to a lower value. For sounds with a slow attack phase, such as string pads, set Attack to a higher value. Similarly, when working with signals that fade out gradually or that have longer reverb tails, set a higher Release knob value that allows the signal to fade out naturally.

The Hold knob determines the minimum amount of time that the gate stays open. You can use the Hold knob to prevent abrupt level changes—known as *chattering*—caused by rapid opening or closing of the gate.

The Hysteresis slider provides another option for preventing chattering, without needing to define a minimum Hold time. Use it to set the range between the threshold values that open and close the Noise Gate. This is useful when the signal level hovers around the Threshold level, causing the Noise Gate to switch on and off repeatedly, producing the undesirable chattering effect. The Hysteresis slider essentially sets 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 accommodate 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 level of the signal you want to keep and the level of the noise signal are close, making it difficult to separate them. For example, when 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, use the side-chain controls to isolate the desired trigger signal with the High Cut and Low Cut filters.

Important: The side-chain signal is used only as a detector/trigger in this situation. The filters are used to isolate particular trigger signals in the side-chain source, but they have no influence on the actual gated signal—the audio being routed through the Noise Gate.

To use the side-chain filters

1. Click the Monitor button to hear how the High Cut and Low Cut filters will affect the incoming trigger signal.
2. Drag the High Cut slider to set the upper frequency. Trigger signals above this are filtered.
3. Drag the Low Cut slider to set the lower frequency. Trigger signals below this are filtered.

The filters allow only very high (loud) signal peaks to pass. In the drum kit example, you could remove the hi-hat signal, which is higher in frequency, with the High Cut filter and allow the snare signal to pass. Turn monitoring off to set a suitable Threshold level more easily.

Preset Multipressor

The Preset Multipressor is a simplified version of the Logic Pro Multipressor plug-in. A multi-band compressor splits the incoming signal into different frequency bands before applying compression. These frequency bands are then compressed independently. Following compression, the frequency bands are mixed back together, and sent to the output of the plug-in. The aim of independent compression on different frequency bands is to strongly compress the bands that need it, without introducing the pumping effect (on other bands) that would normally be heard when using high compression levels.

The Preset Multipressor features a single pop-up menu that allows you to choose settings optimized for various genres. The names of these settings are self-explanatory. As always, use your ears to determine which one best fits your needs.

Note: The Preset Multipressor is automatically inserted in place of the Multipressor when you load Logic Pro projects (that contain Multipressor instances) into Logic Express.

Silver Compressor

The Silver Compressor is a simplified version of the Compressor (for usage tips, see Using the Compressor).

- *Gain Reduction meter:* Shows the amount of compression in real time.
- *Threshold slider and field:* Sets the threshold level. Signals that exceed the threshold are reduced in level.
- *Attack knob and field:* Sets the amount of time it takes for the compressor to react when the signal exceeds the threshold.
- *Release knob and field:* Sets the amount of time it takes for the compressor to stop reducing the signal, after the signal falls below the threshold.
- *Ratio slider and field:* Sets the ratio by which the signal is reduced, when it exceeds the threshold.

Silver Gate

The Silver Gate is a simplified version of the Noise Gate (for usage tips, see Using the Noise Gate).

- *Lookahead slider and field:* Sets how far ahead the noise gate analyzes the incoming signal, allowing the Silver Gate to respond more quickly to peak levels.
- *Threshold slider and field:* Sets the threshold level. Signals that fall below the threshold will be reduced in level.
- *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.

Equalizers

5

An equalizer (commonly abbreviated as *EQ*) shapes the sound of incoming audio by changing the level of specific frequency bands.

Equalization is one of the most commonly used audio processes, both for music projects and in post-production work for video. You can use EQ to subtly or significantly shape the sound of an audio file, instrument, or project by adjusting specific frequencies or frequency ranges.

All EQs are specialized filters that allow certain frequencies to pass through unchanged while raising (boosting) or lowering (cutting) the level of other frequencies. Some EQs can be used in a "broad-brush" fashion, to boost or cut a large range of frequencies. Other EQs, particularly parametric and multiband EQs, can be used for more precise control.

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

Multiband EQs (such as the Channel EQ or Fat EQ) combine several filters in one unit, enabling you to control a large part of the frequency spectrum. Multiband EQs allow you to independently set the frequency, bandwidth, and Q factor of each frequency spectrum band. This provides extensive, and precise, tone-shaping on any audio source, be it an individual audio signal or an overall mix.

Logic Express includes a variety of single band and multiband EQs.

This chapter covers the following:

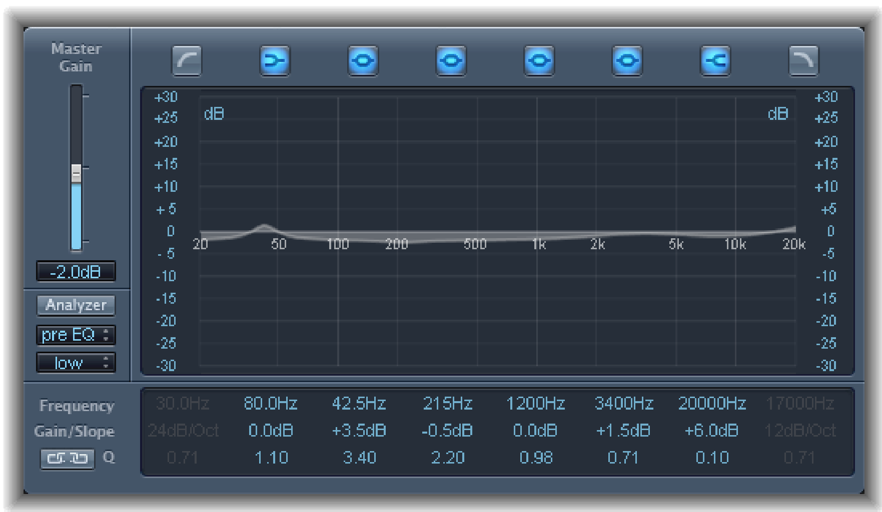
Channel EQ

The Channel EQ is a highly versatile multiband EQ. It provides eight frequency bands, including lowpass 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 may need adjustment.

You can use the Channel EQ to shape the sound of individual tracks or audio files, or for tone-shaping on an overall project mix. The Analyzer and graphic controls make it easy to view and change the audio signal in real time.

Channel EQ Parameters

The left side of the Channel EQ window features the Gain and Analyzer controls. The central area of the window includes the graphic display and parameters for shaping each EQ band.



Channel EQ Gain and Analyzer Controls

- *Master Gain slider and field:* Sets the overall output level of the signal. Use it after boosting or cutting individual frequency bands.
- *Analyzer button:* Turns the Analyzer on or off.
- *Pre/Post EQ button:* Determines whether the Analyzer shows the frequency curve before or after EQ is applied, when Analyzer mode is active.
- *Resolution pop-up menu:* Sets the sample resolution for the Analyzer, with the following menu items: low (1024 points), medium (2048 points), and high (4096 points).

Channel EQ Graphic Display Section

- *Band On/Off buttons:* Click to turn the corresponding band on or off. Each button icon indicates the filter type:

  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.
  - Drag horizontally in the section of the display that encompasses each band to adjust the frequency of the band.
  - Drag vertically in the section of the display that encompasses each band to adjust the gain of each band (except bands 1 and 8). The display reflects your changes immediately.
  - Drag the pivot point in each band to adjust the Q factor. Q is shown beside the cursor when it is moved over a pivot point.

Channel EQ Parameter Section

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

  Note: 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 is set to an extremely high value, such as 100, these filters affect only 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.
- *Analyzer Mode buttons (Extended Parameters area):* Choose Peak or RMS.
- *Analyzer Decay slider and field (Extended Parameters area):* 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 Parameters area):* Choose the amount of Gain-Q coupling.
  - Choose “strong” to preserve most of the perceived bandwidth.
  - Choose “light” or “medium” to 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, rather than 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.

Using the Channel EQ

The way you use the Channel EQ is obviously dependent on the audio material and what you intend to do with it, but a useful workflow for many situations is as follows: Set the Channel EQ to a flat response (no frequencies boosted or cut), turn on the Analyzer and play the audio signal. Keep an eye on the graphic display to see which parts of the frequency spectrum have frequent peaks and which parts of the spectrum stay at a low level. Pay particular attention to sections where the signal distorts or clips. Use the graphic display or parameter controls to adjust the frequency bands as desired.

You can reduce or eliminate unwanted frequencies, and you can raise quieter frequencies to make them more pronounced. You can adjust the center frequencies of 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. While doing so, change the Q parameter(s) so that only a narrow range of frequencies are affected, or widen it to alter a broad area.

Each EQ band has a different color in the graphic display. You can graphically adjust the frequency of a band by dragging horizontally. Drag vertically to adjust the amount of gain for the band. For bands 1 and 8, the slope values can be changed only in the parameter area below the graphic display. Each band has a pivot point (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 scale is shown, 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.

Using the Channel EQ Analyzer

The Analyzer, when active, makes uses of a mathematical process called a Fast Fourier Transform (FFT) to provide a real-time curve of all frequency components in the incoming signal. This is superimposed over any EQ curves you have set. The Analyzer curve uses the same scale as the EQ curves, making it easy to recognize important frequencies in the incoming audio. This also simplifies the task of setting EQ curves to raise or lower the levels of frequencies/frequency ranges.

The bands derived from FFT analysis are divided in a logarithmic scale—there are more bands in higher octaves than in lower ones.

As soon as the Analyzer is activated, you can change the scaling with the Analyzer Top parameter, on the right side of the graphic display. The visible area represents a dynamic range of 60 dB. Drag vertically to set the maximum value to anywhere between +20 dB and –80 dB. The Analyzer display is always dB-linear.

Note: When choosing a resolution, be aware that higher resolutions require significantly more processing power. High resolution is necessary when trying to obtain accurate analysis of very low bass frequencies, for example. It is recommended that you disable the Analyzer or close the Channel EQ window after setting the appropriate 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. You can adjust the Frequency, Gain, and Q-Factor of the latter. The DJ EQ allows the filter gain to be reduced by as much as –30 dB.

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

- *Low Shelf slider and field:* Sets the amount of gain for the low shelving filter.

Fat EQ

The Fat EQ is a versatile multiband EQ which can be used on individual sources or overall mixes. The Fat EQ provides up to five individual frequency bands, graphically displays EQ curves, and includes a set of parameters for each band.

The Fat EQ offers the following parameters.

- *Band Type buttons:* Located above the graphic display. For bands 1–2 and 4–5, click one of the paired buttons to select the EQ type for the corresponding band.
  - *Band 1:* Click the highpass or low shelving button.
  - *Band 2:* Click the low shelving or parametric button.
  - *Band 3:* Always acts as a parametric EQ band.
  - *Band 4:* Click the parametric or high shelving button.
  - *Band 5:* Click the high shelving or lowpass button.
- *Graphic display:* Shows the EQ curve of each frequency band.
- *Frequency fields:* Sets the frequency for each band.
- *Gain knobs:* Set the amount of gain for each band.

- *Q fields:* Sets the Q or bandwidth of each band—the range of frequencies around the center frequency that are altered. At low Q factor values, the EQ covers a wider frequency range. At high *Q* values, the effect of the EQ band is limited to a narrow frequency range. 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 more drastically to notice the difference.

  Note: For bands 1 and 5, this changes the slope of the filter.
- *Band On/Off buttons:* Enables/disables the corresponding band.
- *Master Gain slider and field:* Sets the overall output level of the signal. Use it after boosting or cutting individual frequency bands.

Single-Band EQs

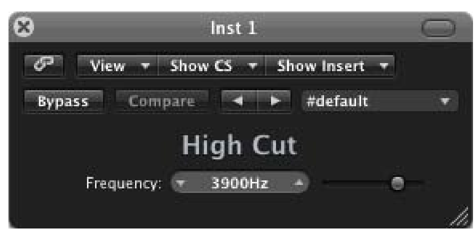
The sections below provide descriptions for the following Logic Express single-band EQ effects:

- Low Cut and High Cut Filter
- High Pass and Low Pass Filter
- High Shelving and Low Shelving EQ
- Parametric EQ

You can find these effects by opening the plug-in menu and choosing EQ > Single Band.

Low Cut and High Cut Filter

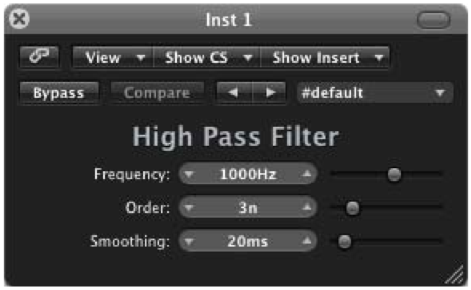
The Low Cut Filter attenuates the frequency range that falls below the selected frequency. The High Cut Filter attenuates the frequency range above the selected frequency. Use the Frequency slider and field to set the cutoff frequency.

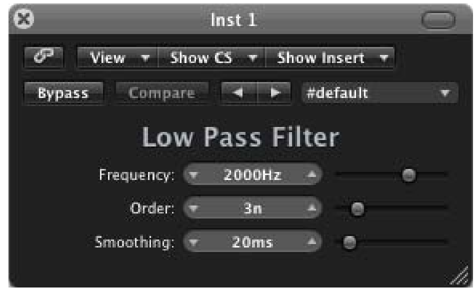


High Pass and Low Pass Filter

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.





- *Frequency slider and field:* Sets the cutoff frequency.
- *Order slider and field:* Sets the filter order. The more orders used, the stronger the filtering effect.
- *Smoothing slider and field:* Adjusts the amount of smoothing, in milliseconds.

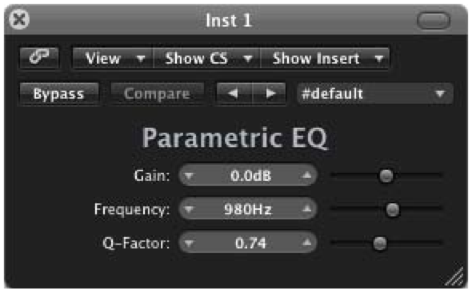
High Shelving and Low Shelving EQ

The Low Shelving EQ affects only the frequency range that falls below the selected frequency. The High Shelving EQ affects only the frequency range above the selected frequency.

- *Gain slider and field:* Sets the amount of cut or boost.
- *Frequency slider and field:* Sets 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.



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

Silver EQ

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

- *High Shelf slider and field:* Sets the level of the high shelving EQ.
- *High Frequency slider and field:* Sets the cutoff frequency for the high shelving EQ.
- *Frequency slider and field:* Sets the center frequency of the parametric EQ.
- *Q-Factor slider and field:* Adjusts the range (bandwidth) of the parametric EQ.

- *Gain slider and field:* Sets the amount of cut or boost for the parametric EQ.
- *Low Shelf slider and field:* Sets the level of the low shelving EQ.
- *Low Frequency slider and field:* Sets the cutoff frequency for the low shelving EQ.

Filter Effects

6

Filters are used to emphasize or suppress frequencies in an audio signal, resulting in a change to the tonal color of the audio.

Logic Express contains a variety of advanced filter-based effects that you can use to creatively modify your audio. These effects are most often used to radically alter the frequency spectrum of a sound or mix.

Note: Equalizers (EQs) are special types of filters. Typically, they are not used as "effects" per-se, but as tools to refine the frequency spectrum of a sound or mix. See Equalizers.

This chapter covers the following:

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 effect works by analyzing incoming signal levels through use of a threshold parameter. Any signal level that exceeds the threshold is used as a trigger for a synthesizer-style ADSR envelope or an LFO (low frequency oscillator). These control sources are used to dynamically modulate the filter cutoff.

The AutoFilter allows you to choose between different filter types and slopes, control the amount of resonance, add distortion for more aggressive sounds, and mix the original, dry signal with the processed signal.

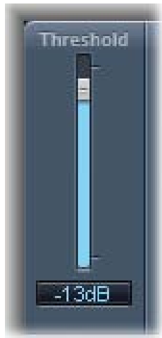
Getting to Know the AutoFilter Interface

The main areas of the AutoFilter window are the Threshold, Envelope, LFO, Filter, Distortion, and Output parameter sections.

- *Threshold parameter:* Sets an input level that—if exceeded—triggers the envelope or LFO, which are used to dynamically modulate the filter cutoff frequency. See AutoFilter Threshold Parameter.
- *Envelope parameters:* Define how the filter cutoff frequency is modulated over time. See AutoFilter Envelope Parameters.
- *LFO parameters:* Define how the filter cutoff frequency is modulated by the LFO. See AutoFilter LFO Parameters.
- *Filter parameters:* Control the tonal color of the filtered sound. See AutoFilter Filter Parameters.
- *Distortion parameters:* Distort the signal both before and after the filter. See AutoFilter Distortion Parameters.
- *Output parameters:* Set the level of both the dry and effect signal. See AutoFilter Output Parameters.

AutoFilter Threshold Parameter

The Threshold parameter analyzes the level of the input signal. If the input signal level exceeds the set threshold level, the envelope and LFO are retriggered—this applies only if the Retrigger button is active.



The envelope and LFO can be used to modulate the filter cutoff frequency.

AutoFilter Envelope Parameters

The envelope is used to shape the filter cutoff over time. When the input signal exceeds the set threshold level, the envelope is triggered.

- *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. If the input signal falls below the threshold level before the envelope sustain phase, the release phase is triggered.
- *Release knob and field:* Sets the release time for the envelope (this is triggered as soon as the input signal falls below the threshold).
- *Dynamic knob and field:* Determines the input signal modulation amount. You can modulate the peak value of the envelope section by varying this control.
- *Cutoff Mod. slider and field:* Determines the impact of the envelope on the cutoff frequency.

AutoFilter LFO Parameters

The LFO is used as a modulation source for filter cutoff.

- *Coarse Rate knob, Fine Rate slider and field:* Used to set the speed of LFO modulation. Drag the Coarse Rate knob to set the LFO frequency in Hertz. Drag the Fine Rate slider (the semicircular slider above the Coarse Rate knob) to fine-tune the frequency.

  Note: The labels shown for the Rate knob, slider, and field change when you activate Beat Sync. Only the Rate knob (and field) is available.
- *Beat Sync button:* Activate to synchronize the LFO to the host application tempo. You can choose from bar values, triplet values, and more. These are determined by the Rate knob or field.
- *Phase knob and field:* Shifts the phase relationship between the LFO rate and the host application tempo—when Beat Sync is active. This parameter is grayed out when Beat Sync is disabled.
- *Decay/Delay knob and field:* Sets the amount of time it takes for the LFO 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. Typically, when the input signal exceeds the threshold, the modulation width of the LFO increases from 0 to the Rate Mod. value. This parameter allows you to override this behavior.
- *Stereo Phase knob and field:* In stereo instances of the AutoFilter, sets the phase relationship of the LFO modulations between the two channels.
- *Cutoff Mod. slider and field:* Determines the impact of the LFO on the cutoff frequency.
- *Retrigger button:* When the Retrigger button is active, the waveform starts at 0 each time the threshold is exceeded.
- *Waveform buttons:* Click one of the following buttons to set the shape of the LFO waveform: descending sawtooth, ascending sawtooth, triangle, pulse wave, or random.
- *Pulse Width slider and field:* Shapes the curve of the selected waveform.

AutoFilter Filter Parameters

The Filter parameters allow you to precisely tailor the tonal color.

- *Cutoff knob and field:* Sets the cutoff frequency for the filter. Higher frequencies are attenuated, whereas lower frequencies are allowed to pass through in a lowpass filter. The reverse is true in a highpass filter. When the State Variable Filter is set to bandpass (BP) mode, the filter cutoff determines the center frequency of the frequency band that is allowed to pass.
- *Resonance knob and field:* Boosts or cuts the signals in the frequency band that surrounds the cutoff frequency. Use of very high Resonance values causes the filter to begin oscillating at the cutoff frequency. This self-oscillation occurs before you reach the maximum Resonance value.
- *Fatness slider and field:* Boosts the level of low frequency content. When you set Fatness to its maximum value, adjusting Resonance has no effect on frequencies below the cutoff frequency. This parameter is used to compensate for a weak or "brittle" sound caused by high resonance values, when in the lowpass filter mode.
- *State Variable Filter buttons:* Switch the filter between highpass (HP), bandpass (BP), or lowpass (LP) modes.
- *4-Pole Lowpass Filter buttons:* Set the slope of the filter to 6, 12, 18, or 24 dB per octave—when the lowpass (LP) filter is chosen as the State Variable Filter.

AutoFilter Distortion Parameters

The Distortion parameters can be used to overdrive the filter input or filter output. The distortion input and output modules are identical, but their respective positions in the signal chain—before and after the filter, respectively—result in remarkably different sounds.

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

AutoFilter Output Parameters

The Output parameters are used to set the wet/dry balance and overall level.

- *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 overall output level of the AutoFilter, allowing you to compensate for higher levels caused by adding distortion—or by the filtering process itself.

EVOC 20 Filterbank

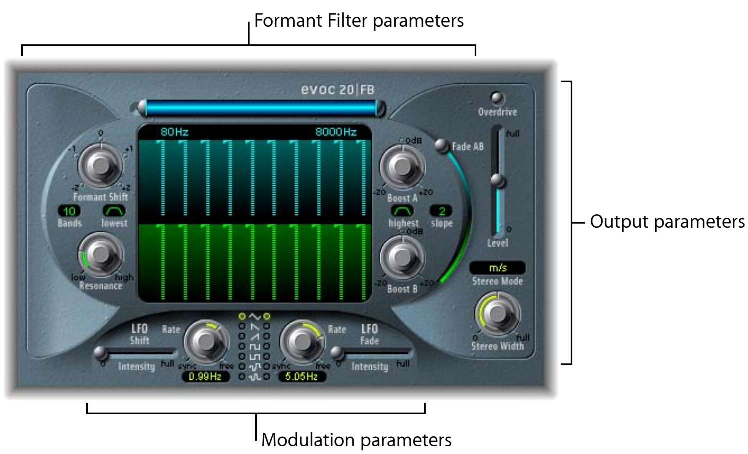
The EVOC 20 Filterbank consists of two formant filter banks. The input signal passes through the two filter banks in parallel. Each bank features level faders for up to 20 frequency bands, allowing independent level control of each band. Setting a level fader to its minimum value completely suppresses the formants in that band. You can control the position of the filter bands with the Formant Shift parameter. You can also crossfade between the two filter banks.

A Short Primer on Formants

A *formant* is a peak in the frequency spectrum of a sound. When the term is used in relation to human voices, formants are the key component that enables humans to distinguish between different vowel sounds—based purely on the frequency of these sounds. Formants in human speech and singing are produced by the vocal tract, with most vowel sounds containing four or more formants.

Getting to Know the EVOC 20 Filterbank Interface

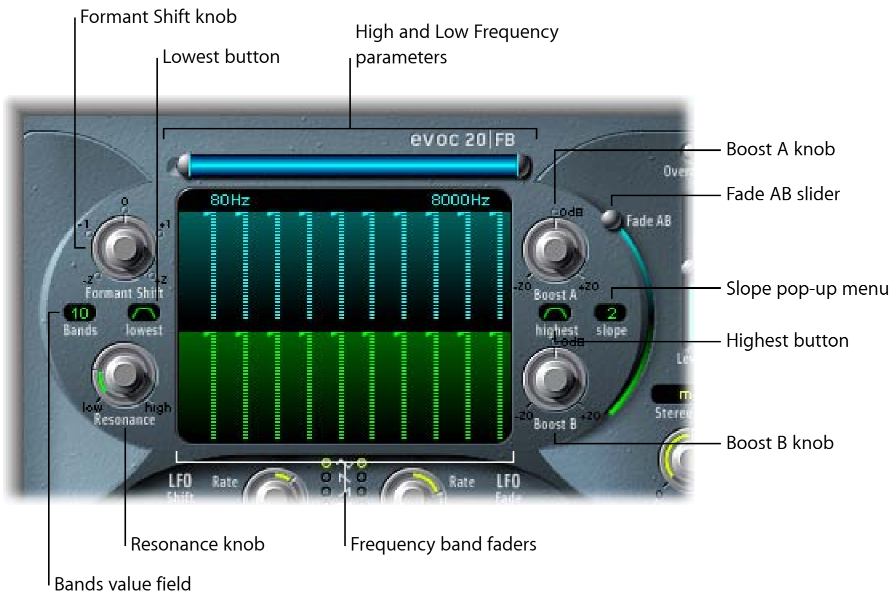
The EVOC 20 Filterbank interface is divided into three main sections: the Formant Filter parameters section in the center of the window, the Modulation parameters section at the bottom center, and the Output parameters section along the right side.



- *Formant Filter parameters:* Control the frequency bands in the two filter banks: Filter Bank A (top, blue) and Filter Bank B (bottom, green). See EVOC 20 Filterbank Formant Filter Parameters.
- *Modulation parameters:* Control how Formant Filter parameters are modulated. See EVOC 20 Filterbank Modulation Parameters.
- *Output parameters:* Control the overall output level and panning of the EVOC 20 Filterbank. See EVOC 20 Filterbank Output Parameters.

EVOC 20 Filterbank Formant Filter Parameters

The parameters in this section provide precise level and frequency control of the filters.



- *High and Low Frequency parameters:* Determine the lowest and highest frequencies allowed to pass by the filter banks. Frequencies that fall outside these boundaries will be cut.
  - The length of the horizontal blue bar at the top represents the frequency range. You can move the entire frequency range by dragging the blue bar. The silver handles on either end of the blue bar set the Low Frequency and High Frequency values, respectively.
  - You can also use the numeric fields to adjust the frequency values separately.
- *Frequency band faders:* Set the level of each frequency band in Filter Bank A (upper blue faders) or Filter Bank B (lower green faders). You can quickly create complex level curves by dragging horizontally ("drawing") across either row of faders.
- *Formant Shift knob:* Moves all bands in both filter banks up or down the frequency spectrum.

  Note: The use of Formant Shift can result in the generation of unusual resonant frequencies—when high Resonance settings are used.
- *Bands value field:* Sets the number of frequency bands—up to 20—in each filter bank.
- *Lowest button:* Click to determine whether the lowest filter band acts as bandpass or highpass filter. In the Bandpass setting, the frequencies below the lowest bands and above the highest bands are ignored. In the Highpass setting, all frequencies below the lowest bands are filtered.

- *Highest button:* Click to determine whether the highest filter band acts as bandpass or lowpass filter. In the Bandpass setting, the frequencies below the lowest bands and above the highest bands are ignored. In the Lowpass setting, all frequencies above the highest bands are filtered.
- *Resonance knob:* Determines the basic sonic character of both filter banks. Increasing Resonance emphasizes the center frequency of each band. Low settings result in a softer character; high settings result in a sharper, brighter character.
- *Boost A and B knobs:* Set the amount of boost—or cut—applied to the frequency bands in Filter Bank A or B. This allows you to compensate for the reduction in volume caused by lowering the level of one or more bands. If you use Boost to set the (level) mix relationship between the filter banks, you can use Fade A/B (see "Fade AB slider" below) to alter the tonal color, but not the levels.
- *Slope pop-up menu:* Sets the amount of filter attenuation applied to all filters in both filter banks. Choices are 1 (6 dB/Oct.) and 2 (12 dB/Oct.). 1 sounds softer, 2 sounds tighter.
- *Fade AB slider:* Crossfades between Filter Bank A and Filter Bank B. At the top position, only Bank A is audible. At the bottom position, only Bank B is audible. In the middle position, the signals passing through both banks are evenly mixed.

EVOC 20 Filterbank Modulation Parameters

The Modulation section offers two LFOs. The LFO Shift parameters on the left side control the Formant Shift parameter. The LFO Fade parameters on the right side control the Fade AB parameter.

- *LFO Shift Intensity slider:* Controls the amount of Formant Shift modulation by the Shift LFO.
- *Rate knobs and fields:* Determine the speed of modulation. Values to the left of the center positions are synchronized with the host application tempo and include bar values, triplet values, and more. Values to the right of the center positions are non synchronized 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, for example. 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 add new life to old audio material.

- *Waveform buttons:* Set the waveform type used by the Shift LFO on the left side or Fade LFO on the right side. You can choose between 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 for each LFO.
- *LFO Fade Intensity slider:* Controls the amount of Fade AB modulation by the Fade LFO.

  Tip: LFO modulations are the key to some extraordinary effects that can be obtained with the EVOC 20 Filterbank. 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. Feel free to try a tempo-synchronized delay effect—such as Tape Delay—after the EVOC 20 Filterbank to produce unique polyrhythms.

EVOC 20 Filterbank Output Parameters

The Output parameters provide control over the level and stereo width. The Output section also incorporates an integrated overdrive (distortion) circuit.

- *Overdrive button:* Click to turn 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:* Controls the volume of the EVOC 20 Filterbank output signal.
- *Stereo Mode pop-up menu:* Sets the input/output mode of the EVOC 20 Filterbank. The choices are m/s (mono input/stereo output) and s/s (stereo input/stereo output).
  - In s/s mode, the left and right channels are processed by separate filter banks.
  - In m/s mode, a stereo input signal is first summed to mono before being routed to the filter banks.

- *Stereo Width knob:* Distributes the output signals of the filter bands in the stereo field.
  - At the left position, the outputs of all bands are centered.
  - At the centered position, the outputs of all bands ascend from left to right.
  - At the right position, the bands are output—alternately—to the left and right channels.

EVOC 20 TrackOscillator

The EVOC 20 TrackOscillator is a vocoder with a monophonic pitch tracking oscillator. The tracking oscillator tracks, or follows, the pitch of a monophonic input signal. If the input signal is a sung vocal melody, the individual note pitches are tracked and mirrored, or played, by the synthesis engine.

The EVOC 20 TrackOscillator features two formant filter banks, an analysis bank, and a synthesis filter bank. Each offers multiple input options.

You can capture an analysis signal source by using the audio arriving at the input of the channel strip that the EVOC 20 TrackOscillator is inserted in, or by using a side chained signal from another channel strip.

The synthesis source can be derived from the audio input of the channel strip that the EVOC 20 TrackOscillator is inserted in, a side chain signal, or the tracking oscillator.

As you can freely select both the analysis and synthesis input signals, the EVOC 20 TrackOscillator is not limited to pitch tracking effects. It is extremely useful for unusual filter effects. For example, you could filter an orchestral recording on one channel strip with train noises side chained from another channel strip. Another great use is for processing drum loops with side chained signals, such as other drum loops or rhythmic guitar, clavinet and piano parts.

What Is a Vocoder?

The word *vocoder* is an abbreviation for *VOice enCODER*. A vocoder analyzes the sonic character of the audio signal arriving at its analysis input and transfers it to the synthesizer's sound generators. 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 sound was popularized in the late 1970s and early 1980s. You'll probably know it from tracks such as "O Superman" by Laurie Anderson, "Funky Town" by Lipps Inc., and numerous Kraftwerk pieces—such as "Autobahn," "Europe Endless," "The Robots," and "Computer World."

In addition to these "singing robot" sounds, vocoding has also been used in many films—such as with the Cylons in Battlestar Galactica, and most famously, with 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 features of a vocoder are actually two bandpass filter banks. Bandpass filters allow a frequency band—a slice—in the overall frequency spectrum to pass through unchanged, and cut the frequencies that fall outside the band's range.

In the EVOC 20 plug-ins, these filter banks are named the Analysis and Synthesis sections. Each filter bank has 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 bands.

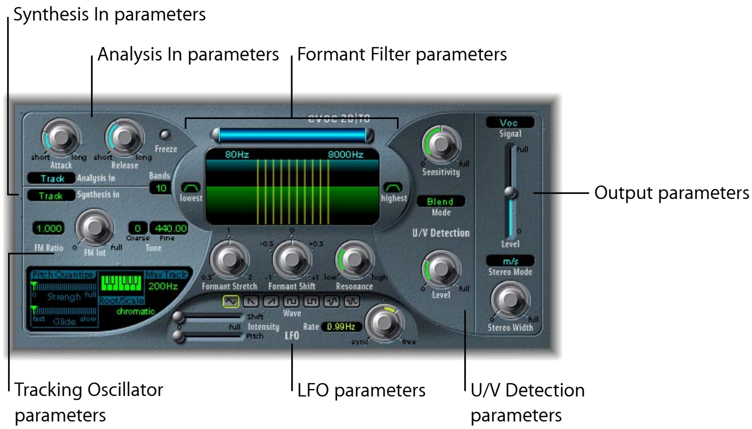
An envelope follower is coupled to each filter band. The envelope follower of each band tracks, or *follows*, any volume changes in the audio source—or, more specifically, the portion of the audio that has been allowed to pass by the associated bandpass filter. In this way, the envelope follower of each 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 voltage-controlled amplifiers (VCAs) in analog vocoders. This allows any volume changes to the bands in the analysis filter bank to be imposed on the matching bands in the synthesis filter bank. These filter changes are heard as a synthetic reproduction of the original input signal—or a mix of the two filter bank signals.

The more bands a vocoder offers, the more precisely the original sound's character will be remodeled. The EVOC plug-ins offer up to 20 bands per bank. To ensure their musical usefulness, you have full control over the output level of each bandpass filter, facilitating unique and dramatic changes to the frequency spectrum.

Getting to Know the EVOC 20 TrackOscillator Interface

The EVOC 20 TrackOscillator window is divided into several parameter sections.



- *Analysis In parameters:* Determine how the input signal is analyzed and used by the analysis filter bank. See EVOC 20 TrackOscillator Analysis In Parameters.
- *U/V Detection parameters:* Detect the unvoiced portions of the sound in the analysis signal, improving speech intelligibility. See EVOC 20 TrackOscillator U/V Detection Parameters.
- *Synthesis In parameters:* Determine how the input signal is used by the synthesis filter bank. See EVOC 20 TrackOscillator Synthesis In Parameters.
- *Tracking Oscillator parameters:* Determine how the analysis input signal is used by the oscillator. See Basic Tracking Oscillator Parameters.
- *Formant Filter parameters:* Configure the analysis and synthesis filter banks. See EVOC 20 TrackOscillator Formant Filter Parameters.
- *Modulation parameters:* Modulate either the oscillator pitch or the Formant Shift parameter. See EVOC 20 TrackOscillator Modulation Parameters.
- *Output parameters:* Configure the output signal of the EVOC 20 TrackOscillator. See EVOC 20 TrackOscillator Output Parameters.

EVOC 20 TrackOscillator Analysis In Parameters

The parameters in the Analysis In section determine how the input signal is analyzed and used by the EVOC 20 TrackOscillator. You should be as precise as possible with these parameters, to ensure the best possible speech intelligibility and accurate tracking.

- *Attack knob:* Determines how quickly each envelope follower—coupled to each analysis filter band—reacts to rising signals.
- *Release knob:* Determines how quickly each envelope follower—coupled to each analysis filter band—reacts to falling signals.
- *Freeze button:* When enabled, holds—or *freezes*—the current analysis sound spectrum indefinitely. While Freeze is enabled, the analysis filter bank ignores the input source, and the Attack and Release knobs have no effect.
- *Bands field:* Determines the number—up to 20—of frequency bands used by the EVOC 20 TrackOscillator.
- *Analysis In pop-up menu:* Sets the analysis signal source. The choices are:
  - *Track:* Uses the input audio signal of the channel strip in which the EVOC 20 TrackOscillator is inserted as the analysis signal.
  - *Side Chain:* Uses a side chain as the analysis signal. You choose the side chain source channel strip from the Side Chain pop-up menu at the top of the plug-in window.

  Note: If Side Chain is chosen and no Side Chain channel strip is assigned, the EVOC 20 TrackOscillator reverts to Track mode operation.

Using EVOC 20 TrackOscillator Analysis In Parameters

This section outlines some settings and approaches for the parameters of the Analysis In section.

Setting the Attack Time

Longer attack times result in a slower tracking response to transients—level spikes—of the analysis input signal. A long attack time on percussive input signals, such as a spoken word or hi-hat part, will translate into a less articulated vocoder effect. Therefore, you should set the Attack parameter to the lowest possible value to enhance articulation.

Setting the Release Time

Longer release times cause the analysis input signal transients to sustain for a longer period, at the vocoder's output. A long release time on percussive input signals, such as a spoken word or hi-hat part, will translate into a less articulated vocoder effect. Use of extremely short release times results in rough, grainy vocoder sounds. Release values of around 8 to 10 ms are useful starting points.

Using Freeze

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. The following are examples of when this could be useful:

If you are using a spoken word pattern as a source, the Freeze button could capture the attack or tail phase of an individual word within the pattern—the vowel *a*, for example.

If you want to compensate for people's inability to sustain sung notes for a long period, without taking a breath, you can use the Freeze button: If the synthesis signal needs to be sustained but the analysis source signal—a vocal part—is not sustained, use the Freeze button to lock the current formant levels of a sung note, even during gaps in the vocal part, between words in a vocal phrase. The Freeze parameter can be automated, which may be useful in this situation.

Setting the Number of Bands

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. 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.

Tip: To ensure the best possible pitch tracking, it is essential to use a mono signal with no overlapping pitches. Ideally, the signal should be unprocessed and free of background noises. Using a signal processed with even a slight amount of reverb, for example, will produce strange and probably 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 perfect for your project.

EVOC 20 TrackOscillator U/V Detection Parameters

Human speech consists of a series of voiced sounds—tonal sounds or formants—and unvoiced sounds—the nonformant nasal continuants, fricatives, and plosives, mentioned in A Short Primer on Formants. The main distinction between voiced and unvoiced sounds is that voiced sounds are produced by an oscillation of the vocal cords, whereas unvoiced sounds are produced by blocking and restricting the air flow with lips, tongue, palate, throat, and larynx.

If speech containing voiced and unvoiced sounds is used as a vocoder's analysis signal, but the synthesis engine doesn't differentiate between voiced and unvoiced sounds, the result will sound rather weak. To avoid this problem, the synthesis section of the vocoder must produce different sounds for the voiced and unvoiced parts of the signal.

The EVOC 20 TrackOscillator includes an Unvoiced/Voiced detector for this specific purpose. 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, with a mixture of noise and synthesizer signal, or with the original signal. 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.

- *Sensitivity knob:* Determines how responsive U/V detection is. When this knob is turned 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 sound source—determined by the Mode menu, as described in "Mode menu" below—being used on the majority of the input signal, including voiced signals. Sonically, this results in a sound that resembles a radio signal that is breaking up and contains a lot of static, or noise.
- *Mode menu:* Sets the sound sources that can be used to replace the unvoiced content of the input signal. You can choose between the following:
  - *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. The Sensitivity parameter has no effect when this setting is used.
- *Level knob:* Controls the volume of the signal used to replace the unvoiced content of the input signal.

Important: Take care with the Level knob, particularly when a high Sensitivity value is used, to avoid internally overloading the EVOC 20 TrackOscillator.

EVOC 20 TrackOscillator Synthesis In Parameters

The Synthesis In section controls various aspects of the tracking signal for the synthesizer. The tracking signal is used to trigger the internal synthesizer.

- *Synthesis In pop-up menu:* Sets the tracking signal source. The choices are:
  - *Oscillator (Osc.):* Sets the tracking oscillator as the synthesis source. The oscillator mirrors, or 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:* Uses the input audio signal of the channel strip, in which the EVOC 20 TrackOscillator is inserted, as the synthesis signal, which drives the internal synthesizer.
  - *Side Chain:* Uses a side chain as the synthesis signal. You choose the side chain source channel from the Side Chain pop-up menu at the top of the EVOC 20 TrackOscillator window.

  Note: If you choose Side Chain and no Side Chain channel is assigned, the EVOC 20 TrackOscillator reverts to Track mode operation.
- *Bands field:* Determines the number of frequency bands used by the Synthesis In section.

Basic Tracking Oscillator Parameters

The tracking oscillator follows the pitch of incoming monophonic audio signals and mirrors these pitches with a synthesized sound. 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 modulated by Oscillator 2 (the modulator), which deforms the sine wave of Oscillator 1. This results in a waveform with rich harmonic content.

Important: The parameters discussed in this section are available only if the Synthesis In menu is set to Osc.

- *FM Ratio field:* 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, whereas 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:* Determines the intensity of modulation. Higher values result in a more complex waveform with more overtones.
  - At a value of 0, the FM tone generator is disabled, and a sawtooth wave is generated.
  - At values above 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.
- *Fine Tune value field:* Sets the pitch offset in cents.

Tracking Oscillator Pitch Correction Parameters

The tracking oscillator pitch parameters control the automatic pitch correction feature of the tracking oscillator. They 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.

- *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 quantized pitches.
- *Root/Scale keyboard and pop-up menu:* Define the pitch or pitches that the tracking oscillator is quantized to.
- *Max Track value field:* Sets the highest frequency. All frequencies above this threshold are cut, making pitch detection more robust. If pitch detection produces unstable results, reduce this parameter to the lowest possible setting that allows all appropriate input signals to be heard or processed.

Quantizing the Pitch of the Tracking Oscillator

You can use the Root/Scale keyboard and pop-up menu to define the pitch or pitches that the tracking oscillator is quantized to.

To choose a root or scale

1. Click the green value field below the Root/Scale label to open the pop-up menu.
2. Choose the scale or chord that you want to use as the basis for pitch correction.

Note: You can also set the root key of the respective scale or chord by vertically dragging the Root value field, or by double-clicking it and entering a root between C and B. The Root parameter is not available when the Root/Scale value is set to "chromatic" or "user."

To add notes to, or remove notes from, the chosen scale or chord

- Click unused keys on the small keyboard to add them to the scale or chord.
- Click selected notes (illuminated) to remove them.

Tip: Your last edit is remembered. If you choose a new scale or chord but do not make any changes, you can revert to the previously set scale by choosing "user" in the pop-up menu.

EVOC 20 TrackOscillator Formant Filter Parameters

The EVOC 20 TrackOscillator features two formant filter banks—one for the Analysis In section and one for the Synthesis In section. In essence, the entire frequency spectrum of an incoming signal is analyzed (Analysis section), and equally divided into a number of frequency bands. Each filter bank can control up to 20 of these frequency bands. For more information, see How Does a Vocoder Work?.

The Formant Filter display is divided in two by a horizontal line. The upper half applies to the Analysis section and the lower half to the Synthesis section. Parameter changes are instantly reflected in the Formant Filter display, providing invaluable feedback on what is happening to the signal as it is routed through the two formant filter banks.

- *High and Low Frequency parameters:* Determine the lowest and highest frequencies allowed to pass by the filter section. Frequencies that fall outside these boundaries will be cut.
  - The length of the blue bar represents the frequency range for both analysis and synthesis—unless Formant Stretch or Formant Shift are used, as discussed in "Formant Stretch knob" and "Formant Shift knob" below. You can move the entire frequency range by dragging the horizontal blue bar at the top. The silver handles on either end of the blue bar set the Low Frequency and High Frequency values, respectively.
  - You can also use the numeric fields to adjust the frequency values separately.
- *Lowest button:* Click to determine whether the lowest filter band acts as bandpass or highpass filter. In the Bandpass setting, the frequencies below the lowest bands and above the highest bands are ignored. In the Highpass setting, all frequencies below the lowest bands are filtered.
- *Highest button:* Click to determine whether the highest filter band acts as bandpass or lowpass filter. In the Bandpass setting, the frequencies below the lowest bands and above the highest bands are ignored. In the Lowpass setting, all frequencies above the highest bands are filtered.
- *Formant Stretch knob:* Alters the width and distribution of all bands in the synthesis filter bank. This can be a broader or narrower frequency range than that defined by the blue bar (see "High and Low Frequency parameters" above).
- *Formant Shift knob:* Moves all bands in the synthesis filter bank up and down the frequency spectrum.

- *Resonance knob:* Resonance is responsible for the basic sonic character of the vocoder—low settings result in a soft character, whereas high settings lead to a more snarling, sharp character. Technically, increasing the Resonance value emphasizes the middle frequency of each frequency band.

Using Formant Stretch and Formant Shift

Formant Stretch and Formant Shift are significant Formant Filter parameters that you can use separately or in combination (see EVOC 20 TrackOscillator Formant Filter Parameters).

When Formant Stretch is set to 0, the width and distribution of the bands in the Synthesis filter bank at the bottom matches the width of the bands in the Analysis filter bank at the top. Low values narrow the width of each band in the Synthesis bank, whereas high values widen the bands. The control range is expressed as a ratio of the overall bandwidth.

When Formant Shift is set to 0, the positions of the bands in the Synthesis filter bank match the positions of the bands in the Analysis filter bank. Positive values move the Synthesis filter bank bands up in frequency, whereas negative values move them down—in respect to the Analysis filter bank band positions.

When combined, Formant Stretch and Formant Shift alter the formant structure of the resulting vocoder sound, which can lead to some interesting timbre changes. For 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.

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

EVOC 20 TrackOscillator Modulation Parameters

The parameters in this section control the LFO, which can be used to modulate either the frequency—the pitch—of the tracking oscillator, thus creating a vibrato, or the Formant Shift parameter of the synthesis filter bank.

- *Shift Intensity slider:* Controls the amount of formant shift modulation by the LFO.
- *Pitch Intensity slider:* Controls the amount of pitch modulation—vibrato—by the LFO.

- *Waveform buttons:* Set the waveform type used by the LFO. You can choose between 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 for each LFO.
- *LFO Rate knob and field:* Determines the speed of modulation. Values to the left of the center positions are synchronized with the host application tempo and include bar values, triplet values, and more. Values to the right of the center positions are non synchronized 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, for example. Alternatively, 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 add new life to old audio material.

EVOC 20 TrackOscillator Output Parameters

The Output section provides control over the type, stereo width, and level of signal that is sent from the EVOC 20 TrackOscillator.

- *Signal menu:* Determines the signal that is sent to the EVOC 20 TrackOscillator main outputs. You can choose among the following settings:
  - *Voc(oder):* Choose to hear the vocoder effect.
  - *Syn(thesis):* Choose to hear only the synthesizer signal.
  - *Ana(lysis):* Choose to hear only the analysis signal.

  Note: The last two settings are mainly useful for monitoring purposes.
- *Level slider:* Controls the volume of the EVOC 20 TrackOscillator 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).

  Note: 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 you use m/s mode on a stereo input signal, the signal is first summed to mono before it is passed to the filter banks.

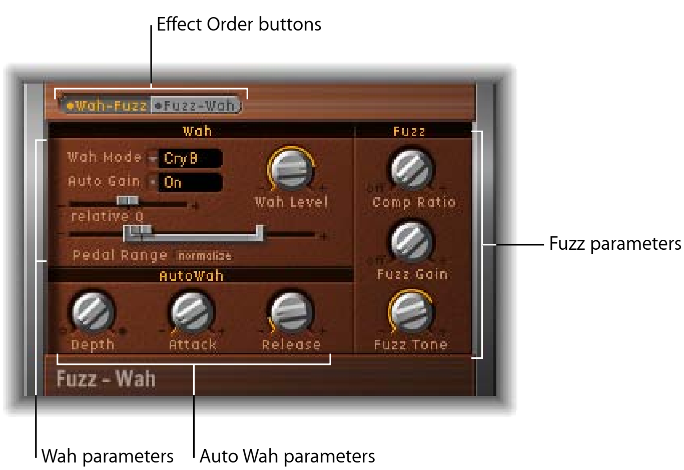
- *Stereo Width knob:* Distributes the output signals of the Synthesis section's filter bands in the stereo field.
  - At the left position, the outputs of all bands are centered.
  - At the centered position, the outputs of all bands ascend from left to right.
  - At the right position, the bands are output—alternately—to the left and right channels.

Fuzz-Wah

The Fuzz-Wah plug-in emulates classic wah wah effects often used with a clavinet, and it adds compression and fuzz distortion effects as well. The name *wah wah* comes from the sound it produces. It has been a popular effect—usually a pedal 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.

Getting to Know the Fuzz-Wah Interface

The Fuzz-Wah interface is broken down into the following sections.



- *Effect Order buttons:* Select whether the wah wah effect precedes the fuzz effect in the signal chain—Wah-Fuzz—or vice versa—Fuzz-Wah. See Effect Order Buttons.

- *Wah parameters:* Provide control over the type and tone of the wah wah effect. See Wah Parameters.
- *Auto Wah parameters:* Set the depth and envelope times for the automatic wah wah effect. See Auto Wah Parameters.
- *Fuzz parameters:* Set the compression ratio, and control the tone and level of the integrated distortion circuit. See Fuzz Parameters.

Effect Order Buttons

These buttons determine the signal flow of the Fuzz-Wah effect. Click Wah-Fuzz or Fuzz-Wah to choose the desired flow.

Note that the Fuzz-Wah plug-in features an integrated compression circuit. The compressor always precedes the fuzz effect. When Wah-Fuzz is selected, the compressor is positioned between the wah wah and the fuzz effect. When Fuzz-Wah is selected, however, the compressor is placed first in the signal chain.

Wah Parameters

This group of parameters controls the tone and behavior of the wah wah effect.

- *Wah Mode pop-up menu:* Includes the following Wah Wah effect settings:
  - *Off:* Wah Wah effect is disabled.
  - *ResoLP (Resonating Lowpass Filter):* In this mode, the Wah Wah works 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 works as a resonance-capable highpass filter. At the maximum pedal position, only high frequencies can pass.
  - *Peak:* In this mode, the Wah Wah works as a peak, or bell, filter. Frequencies close to the cutoff frequency are emphasized.
  - *CryB:* This setting mimics the sound of the popular Cry Baby wah wah pedal.

- *Morl1:* This setting mimics the sound of a popular wah wah pedal. It features a slight peak characteristic.
- *Morl2:* This setting mimics the sound of a popular distortion wah wah pedal. It has a constant Q(uality) Factor setting.

- *Auto Gain button:* The wah wah effect can cause wide variations in the output level. Turning Auto Gain on compensates for this behavior, and limits the output signal dynamics (see Setting the Wah Wah Level with Auto Gain).
- *Wah Level knob:* Sets the amount of the wah-filtered signal.
- *Relative Q slider:* Adjusts the main filter peak, relative to the model setting, thereby obtaining a sharper or softer wah wah sweep. When set to a value of 0, the original peak level setting of the model is active.
- *Pedal Range slider:* Sets the sweep range of the Wah Wah filter—when controlled with a MIDI foot pedal. This parameter is designed to compensate for the differences in mechanical range between a MIDI foot pedal and a classic Wah Wah pedal (see Setting the Pedal Range).

Auto Wah Parameters

In addition to using MIDI foot pedals, you can control the Wah Wah effect with the Auto Wah feature, which continually performs a filter sweep across the entire range. See Using the Fuzz-Wah.

- *Depth knob:* Sets the depth of the Auto Wah effect. When set to zero the automatic Wah Wah function is disabled.
- *Attack knob:* Sets the time it takes for the Wah Wah filter to fully open.
- *Release knob:* Sets the time it takes for the Wah Wah filter to close.

Fuzz Parameters

These parameters control the integrated distortion and compression circuits. The compressor always precedes the Fuzz effect.

- *Comp (Compression) Ratio knob:* Sets the compression ratio.
- *Fuzz Gain knob:* Sets the level of the Fuzz, or distortion, effect.
- *Fuzz Tone knob:* Adjusts the tonal color of the fuzz effect. Low settings tend to be warmer, and high settings are brighter and harsher.

Using the Fuzz-Wah

The following section provides practical tips for the Fuzz-Wah parameters.

Setting the Wah Wah Level with Auto Gain

The Wah 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.
3. Make a sweep with a high relative Q setting.
4. Switch Auto Gain to off, and repeat the sweep.

Important: Make sure to set a conservative master output level for your host application before trying this. Failure to do so may result in damage to your hearing or speakers.

Setting the Pedal Range

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

The sweep range of the Wah Wah filter is set with the Pedal Range parameters. The highest and lowest possible values reached by a MIDI foot pedal are graphically represented by a gray bracket around the Pedal Position slider (the slider represents the current position of the Wah Wah pedal).

You can set the upper and lower limits of the range independently by dragging the left and right handles of the slider bracket. You can move the entire range by dragging the center section of the slider bracket.

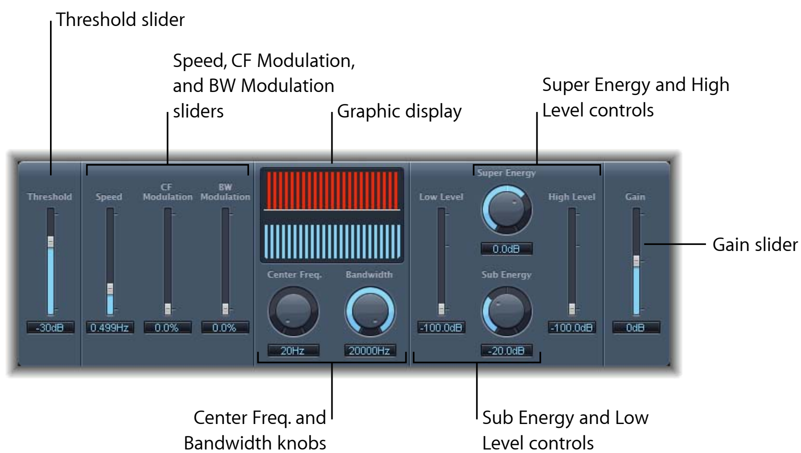
Spectral Gate

The Spectral Gate is an unusual filter effect that can be used as a tool for creative sound design.

It works by dividing the incoming signal into two frequency ranges—above and below a central frequency band that you specify with the Center Freq and Bandwidth parameters. The signal ranges above and below the defined band can be individually processed with the Low Level and High Level parameters and the Super Energy and Sub Energy parameters. See Using the Spectral Gate.

Spectral Gate Parameters

The Spectral Gate panel includes the following parameters:



- *Threshold slider and field:* Sets the threshold level for division of frequency ranges. When the threshold is exceeded, 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 band that you want to process.
- *Bandwidth knob and field:* Sets the width of the frequency band that you want to process.
- *Super Energy knob and field:* Controls the level of the frequency range above the threshold.
- *High Level slider and field:* Blends the frequencies of the original signal—above the selected frequency band—with the processed signal.
- *Sub Energy and field:* Controls the level of the frequency range below the threshold.
- *Low Level slider and field:* Blends the frequencies of the original signal—below the selected frequency band—with the processed signal.
- *Gain slider and field:* Sets the output level of the Spectral Gate.

Using the Spectral Gate

One way to familiarize yourself 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 (20,000 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.

Follow these steps to acquaint yourself with the Spectral Gate

1. Set the frequency band you want to process by using the Center Freq. and Bandwidth parameters.

   The graphic display visually indicates the band defined by these two parameters.

2. After the frequency band is defined, use the Threshold parameter to set the appropriate level.

   All incoming signals above and below the threshold level are divided into upper and lower frequency ranges.

3. 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.

4. You can mix the frequencies that fall outside the frequency band (defined by the Center Freq. and Bandwidth parameters) with the processed signal.

   a. Use the Low Level slider to blend the frequencies below the defined frequency band with the processed signal.

b Use the High Level slider to blend frequencies above the defined frequency band with the processed signal.

5 You can modulate the defined frequency band using the Speed, CF Modulation, and BW Modulation parameters.

a Speed determines the modulation frequency.

b CF (Center Frequency) Modulation defines the intensity of the center frequency modulation.

c BW (Band Width) Modulation controls the amount of bandwidth modulation.

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

Imaging Processors

7

The Logic Express Imaging processors are tools for manipulating the stereo image. This enables 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.

This chapter covers the following:

Direction Mixer

You can use the Direction Mixer to decode middle and side audio recordings or to spread the stereo base of a left/right recording and determine its pan position.

The Direction Mixer works with any type of stereo recording, regardless of the miking technique used. For information about XY, AB, and MS recordings, see Getting to Know Stereo Miking Techniques.

- *Input buttons:* Click the LR button if the input signal is a standard *left/right* signal, and click the MS button if the signal is *middle and side* encoded.

- *Spread slider and field:* Determines the spread of the stereo base in LR input signals. Determines the level of the side signal in MS input signals. See Using the Direction Mixer's Spread Parameter.
- *Direction knob and field:* Determines the pan position for the middle—the center of the stereo base—of the recorded stereo signal. See Using the Direction Mixer's Direction Parameter

Using the Direction Mixer's Spread Parameter

The Direction Mixer's Spread parameter behavior changes when fed LR or MS signals. These differences are outlined below:

When working with LR signals, the following applies to the Direction Mixer's Spread parameter:

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

The following applies when working with MS signals:

- Values of 1 or higher increase the level of the side signal, making it louder than the middle signal.
- At a value of 2, you hear only the side signal.

Using the Direction Mixer's Direction Parameter

When Direction is set to a value of 0, the midpoint of the stereo base in a stereo recording is perfectly centered within the mix.

The following applies when working with LR signals:

- At 90°, the center of the stereo base is panned hard left.
- At –90°, the center of the stereo base is panned hard right.
- Higher values move the center of the stereo base back toward the center of the stereo mix, but this also has the effect of swapping the stereo sides of the recording. For example, at values of 180° or –180°, the center of the stereo base is dead center in the mix, but the left and right sides of the recording are swapped.

The following applies when working with MS signals:

- At 90°, the middle signal is panned hard left.
- At –90°, the middle signal is panned hard right.

- Higher values move the middle signal back toward the center of the stereo mix, but this also has the effect of swapping the side signals of the recording. For example, at values of 180° or –180°, the middle signal is dead center in the mix, but the left and right sides of the side signal are swapped.

Getting to Know Stereo Miking Techniques

There are three commonly used stereo miking variants used in recording: AB, XY, and MS. A stereo recording, put simply, is one that contains two channel signals.

AB and XY recordings both record left and right channel signals, but the middle signal is the result of combining both channels.

MS recordings record a real middle signal, but the left and right channels need to be decoded from the side signal, which is the sum of both left and right channel signals.

Understanding AB Miking

In an AB recording, two microphones—commonly omnidirectional, but any polarity can be used—are equally spaced from the center and pointed directly at the sound source. Spacing between microphones is extremely important for the overall stereo width and perceived positioning of instruments within the stereo field.

The AB technique is commonly used for recording one section of an orchestra, such as the string section, or perhaps a small group of vocalists. It is also useful for recording piano or acoustic guitar.

AB is not well suited to recording a full orchestra or group as it tends to smear the stereo imaging/positioning of off-center instruments. It is also unsuitable for mixing down to mono, as you run the risk of phase cancellations between channels.

Understanding XY Miking

In an XY recording, two directional microphones are symmetrically angled, from the center of the stereo field. The right-hand microphone is aimed at a point between the left side and the center of the sound source. The left-hand microphone is aimed at a point between the right side and the center of the sound source. This results in a 45° to 60° off-axis recording on each channel (or 90° to 120° between channels).

XY recordings tend to be balanced in both channels, with good positional information being encoded. It is commonly used for drum recording. XY recording is also suitable for larger ensembles and many individual instruments.

Typically, XY recordings have a narrower sound field than AB recordings, so they can lack a sense of perceived width when played back. XY recordings can be mixed down to mono.

Understanding MS Miking

To make a Middle Side (MS) 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 that directly faces the sound source 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 one side of a stereo recording. The bidirectional microphone records the side signal to the other side of a stereo recording. MS recordings made in this way can be decoded by the Direction Mixer.

When MS recordings are played back, the side signal is used twice:

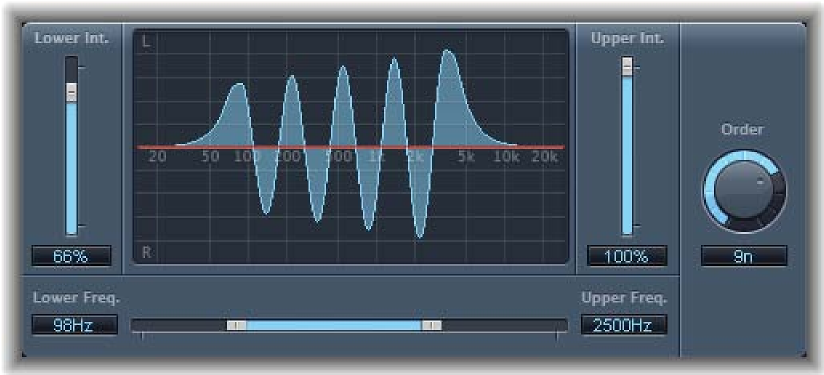
- As recorded
- Panned hard left and phase reversed, panned hard right

MS is ideal for all situations where you need to retain absolute mono compatibility. The advantage of MS recordings over XY recordings is that the stereo middle is positioned on the main recording direction (on-axis) 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, because the recording always retains mono compatibility.

Stereo Spread

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

Stereo Spread 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.



- *Lower Int(ensity) slider and field:* Sets the amount of stereo base extension for the lower frequency bands.
- *Upper Int(ensity) slider and field:* Sets the amount of stereo base extension for the upper frequency bands.

  Note: When setting the Lower and Upper Int. sliders, be aware that the stereo effect is most apparent in the middle and higher frequencies, so distributing low frequencies between the left and right speakers can significantly alter the energy of the overall mix. For this reason, use low values for the Lower Int. parameter, and avoid setting the Lower Freq. parameter 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.
- *Upper and Lower Freq(uency) slider and fields:* Determine the highest and lowest frequencies that will be redistributed in the stereo image.
- *Order knob and field:* Determines the number of frequency bands that the signal is divided into. A value of 8 is usually sufficient for most tasks, but you can use up to 12 bands.

Metering Tools

8

You can use the Metering tools to analyze audio in a variety of ways. These plug-ins offer different facilities to the meters shown in channel strips. They have no effect on the audio signal and are designed for use as diagnostic aids.

Each meter is specifically designed to view different characteristics of an audio signal, making each suitable for particular studio situations. As examples, the BPM Counter displays the tempo, the Correlation Meter displays the phase relationship, and the Level Meter displays the level of an incoming audio signal.

This chapter covers the following:

BPM Counter

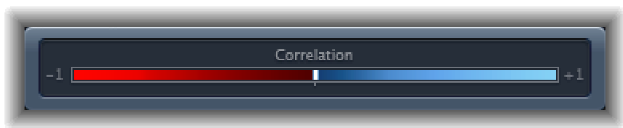
The BPM Counter is used to analyze the tempo of incoming audio in beats per minute (bpm). The detection circuit looks for any transients, also known as impulses, in the input signal. Transients are very fast, non-periodic sound events in the attack portion of the signal. The more obvious this impulse is, the easier it is for the BPM Counter to detect the tempo. As a result, percussive drum and instrumental rhythm parts, such as basslines, are suitable 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 the LED is continuously 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. Click the LED to reset the BPM Counter.

Note: The BPM Counter also detects tempo variations in the signal and tries to analyze 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 is solidly lit and the new tempo displayed.

Correlation Meter

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

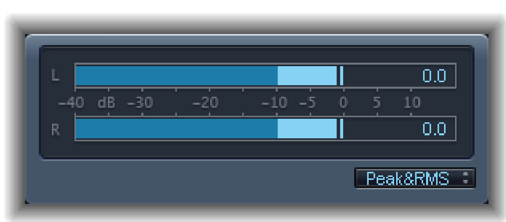


- A correlation of +1 (the far right position) means that the left and right channels correlate 100%—they are completely in phase.
- A correlation of 0 (the center position) indicates the widest permissible left/right divergence, often audible as an extremely wide stereo effect.
- Correlation values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

Level Meter Plug-in

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 to the right of the 0 dB point turns red.

Stereo instances of the Level Meter show independent left and right bars, whereas mono instances display a single bar.



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. Choose the appropriate 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.

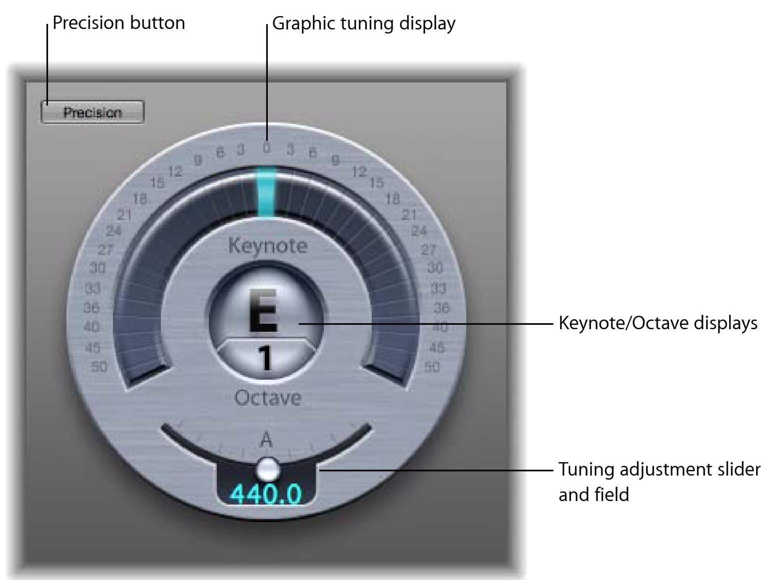
Peak and RMS Explained

The *peak* value is the highest level that the signal will reach. The *RMS* (root mean square) value is the effective value of the total signal. In other words, it is a measurement of the continuous power of the signal.

Human hearing is optimized for capturing continuous signals, making our ears RMS instruments, not peak reading instruments. Therefore, using RMS meters makes sense most of the time. Alternatively, you can use both RMS and Peak meters.

Tuner

You can tune instruments connected to your system with the Tuner utility. This ensures that your external instrument recordings will be in tune with any software instruments, samples, or existing recordings in your projects.



- *Precision button:* In Logic, the Graphic tuning display defaults to a *linear scale*. Enable the Precision button to change the scaling so that it stretches outwards from the center. This parameter is not available in MainStage.

- *Graphic tuning display:* Indicates the pitch of the note in the semicircular area around the Keynote/Octave displays. At the centered (12 o'clock) position, the note is correctly tuned. If the indicator moves to the left of center, the note is flat. If the indicator 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. The range is marked in single semitone steps for the first 6 semitones (sharp or flat). Thereafter, larger increments are shown.

- *Keynote/Octave displays:* The upper, Keynote display shows the target pitch of the note being played (the closest tuned pitch). The lower, Octave display indicates the octave that the incoming note falls into. This matches the MIDI octave scale, with the C above middle C displayed as C4, and middle C displayed as C3.
- *Tuning Adjustment slider and field:* Sets the pitch of the note used as the basis for tuning. By default, the Tuner is set to the project's Tuning parameter value. Drag the knob to the left to lower the pitch corresponding to A. Drag the knob to the right to raise the pitch corresponding to A. The current value is displayed in the field.

To use the Tuner

1. Insert the Tuner into an audio channel strip.
2. Play a single note on the instrument and watch the display. If the note is flat or sharp (of the Keynote), the segments to the left or right of center are illuminated, indicating how far in cents the note is off pitch.
3. Adjust the tuning of your instrument until the indicator is centered in the graphic tuning display.

Modulation Effects

9

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

Effects such as chorus, flanging, and phasing are well-known examples. Modulation effects typically delay the incoming signal by a few milliseconds and use an LFO to modulate the delayed signal. The LFO may also be used to modulate the delay time in some effects.

A low frequency oscillator (LFO) is much like the sound-generating oscillators in synthesizers, but the frequencies generated by an LFO are so low that they can't be heard. Therefore, they are used only for modulation purposes. LFO parameters include speed (or frequency) and depth—also called *intensity*—controls.

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.

Other modulation effects involve pitch. The most basic type of pitch modulation effect is vibrato. It uses an LFO to modulate the frequency of the sound. Unlike other pitch modulation effects, vibrato alters only the delayed signal.

More complex Logic Express modulation effects, such as Ensemble, mix several delayed signals with the original signal.

This chapter covers the following:

- Spreader (p. 154)
- Tremolo Effect (p. 155)

Chorus Effect

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 incoming signal and create the impression that multiple instruments or voices are being played in unison. The slight delay time variations generated by the LFO simulate the subtle pitch and timing differences heard when several musicians or vocalists perform together. Using chorus also adds fullness or richness to the signal, and it can add movement to low or sustained sounds.

- *Intensity slider and field:* Sets 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 Effect

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 display visually represents what is happening with the processed signals.

The Ensemble effect can add a great deal of richness and movement to sounds, particularly when you use a high number of voices. It is very useful for thickening parts, but it can also be used to emulate more extreme pitch variations between voices, resulting in a detuned quality to processed material.

- *Intensity sliders and fields:* Set the amount of modulation for each LFO.
- *Rate knobs and fields:* Control the frequency of each LFO.
- *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.
- *Graphic display:* Indicates the shape and intensity of the modulations.
- *Phase knob and field:* Controls the phase relationship between the individual voice modulations. The value you choose here is dependent on the number of voices, which is why it is shown as a percentage value rather than in degrees. The value 100 (or –100) indicates the greatest possible distance between the modulation phases of all voices.
- *Spread slider and field:* Distributes voices across the stereo field. Set a value of 200% to artificially expand the stereo base. 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:* Determines the level of the effects signal. This is a useful tool that compensates for changes in volume caused by changes to the Voices parameter.

Flanger Effect

The Flanger effect works in much the same way as the Chorus effect, but it 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 that are described as adding a spacey or underwater quality to input signals.

- *Feedback slider and field:* Determines the amount of the effect signal that is routed back into the input. This can change the tonal color and/or make the sweeping effect more pronounced. Negative Feedback values invert the phase of the routed signal.
- *Rate knob and field:* Defines the frequency (the speed) of the LFO.
- *Intensity slider and field:* Determines the modulation amount.
- *Mix slider and field:* Determines the balance between dry and wet signals.

Microphaser

The Microphaser is a simple plug-in that allows you to quickly create swooshing, phasing effects.

- *LFO Rate slider and field:* Defines the frequency (the speed) of the LFO.

- *Feedback slider and field:* Determines the amount of the effect signal that is routed back into the input. This can change the tonal color and/or make the sweeping effect more pronounced.
- *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.

Although rich, combined flanging and chorus effects are possible, the Modulation Delay is capable of producing some extreme modulation effects. These include emulations of tape speed fluctuations and metallic, robot-like modulations of incoming signals.

- *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, don't use any feedback. Negative values invert the phase of the feedback signal, resulting in more chaotic effects.
- *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.
- *De-Warble button:* Ensures that the pitch of the modulated signal remains constant.
- *Const Mod. (Constant Modulation) button:* Ensures that the modulation width remains constant, regardless of the modulation rate.

  Note: When Const Mod is enabled, higher modulation frequencies reduce the modulation width.
- *Mod. Intensity slider and field:* Sets the modulation amount.
- *LFO Mix slider and fields:* Determines the balance between the two LFOs.

- *LFO 1 and LFO 2 Rate knobs and fields:* The left knob sets the modulation rate for the left stereo channel, and the right knob sets the modulation rate for the right stereo channel.

  Note: The right LFO Rate knob is available only in stereo instances, and it can be set separately only if the Left Right Link button is *not* enabled.

- *LFO Left Right Link button:* Available only in stereo instances, it links the modulation rates of the left and right stereo channels. Adjustment of either Rate knob will affect the other channels.
- *LFO Phase knob and field:* Available only in stereo instances, it controls the phase relationship between 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.

  Note: The LFO Phase parameter is available only if the LFO Left Right Link button is active.

- *Volume Mod(ulation) slider and field:* Determines the impact that LFO modulation has on the amplitude of the effect signal.
- *Output Mix slider and field:* Determines the balance between dry and wet signals.
- *All Pass button (Extended Parameters area):* Introduces an additional allpass filter into the signal path. An allpass filter shifts the phase angle of a signal, influencing its stereo image.
- *All Pass Left and All Pass Right sliders and fields (Extended Parameters area):* Determines the frequency at which the phase shift crosses 90° (the half-way point of the total 180°) for each of the stereo channels.

Phaser Effect

The Phaser effect combines the original signal with a copy that is slightly out of phase with the original. This means that the amplitudes of the two signals reach their highest and lowest points at slightly different times. The timing differences between the two signals are modulated by two independent LFOs. In addition, the Phaser includes a filter circuit and a built-in envelope follower that tracks volume changes in the input signal, generating a dynamic control signal. This control signal alters the sweep range. Sonically, phasing is used to create whooshing, sweeping sounds that wander through the frequency spectrum. It is a commonly used guitar effect, but it is suitable for a range of signals.

Phaser Feedback Section

- *Filter button:* Activates the filter section, which processes the feedback signal.
- *LP and HP knobs and fields:* Set the cutoff frequency of the filter section's lowpass (LP) and highpass (HP) filters.
- *Feedback slider and field:* Determines the amount of the effect signal that is routed back into the input of the effect.

Phaser Sweep Section

- *Ceiling and Floor sliders and fields:* Use the individual slider handles to determine the frequency range 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.

  The 4, 6, 8, 10, and 12 settings put five different phaser algorithms at your fingertips. All are modeled on analog circuits, with 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.
- *Env Follow slider and field:* Determines the impact of incoming signal levels on the frequency range (as set with the Ceiling and Floor controls).

Phaser LFO Section

- *LFO 1 and LFO 2 Rate knobs and fields:* Set the speed for each LFO.
- *LFO Mix slider and fields:* Determines the ratio between the two LFOs.
- *Env Follow slider and field:* Determines the impact of incoming signal levels on the speed of LFO 1.
- *Phase knob and field:* Available only in stereo 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.

Phaser Output Section

- *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:* Enables a distortion circuit, suitable for warm overdrive effects.

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.

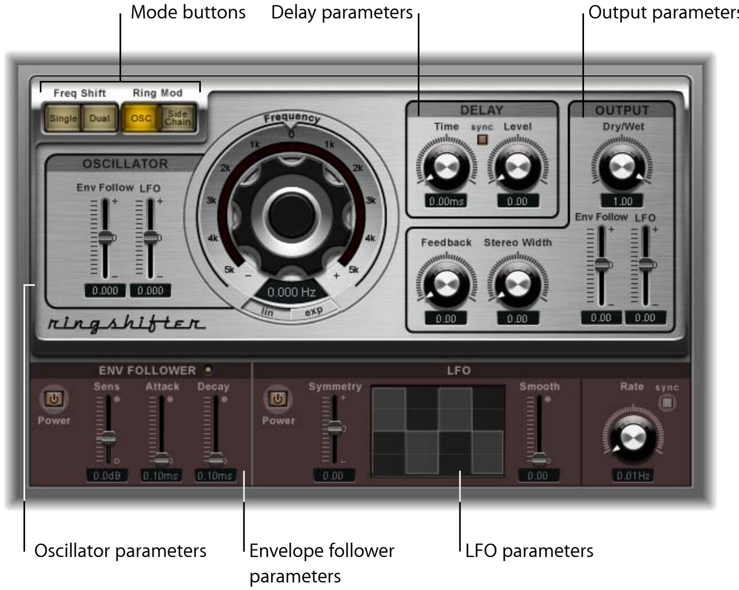
The ring modulator modulates the amplitude of the input signal using either the internal oscillator or a side-chain signal. The frequency spectrum of the resulting effect signal equals the sum and difference of the frequency content in the two original signals. Its sound is often described as *metallic* or *clangorous*. The ring modulator was used extensively on jazz rock and fusion records in the early 1970s.

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 from sweet and spacious phasing effects to strange robotic timbres.

Note: Frequency shifting should not be confused with pitch shifting. Pitch shifting transposes the original signal, leaving its harmonic frequency relationship intact.

Getting to Know the Ringshifter Interface

The Ringshifter interface consists of six main sections.



- *Mode buttons:* Determine whether the Ringshifter operates as frequency shifter or ring modulator See Setting the Ringshifter Mode.
- *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. See Using the Ringshifter's Oscillator.
- *Delay parameters:* Use these to delay the effect signal. See Using the Ringshifter's Delay.
- *Envelope follower parameters:* The oscillator frequency and output signal can be modulated with an envelope follower. See Modulating the Ringshifter with the Envelope Follower.
- *LFO parameters:* The oscillator frequency and output signal can be modulated with an LFO. See Modulating the Ringshifter with the LFO.
- *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. See Controlling the Ringshifter Output Parameters.

Setting the Ringshifter Mode

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.

Using the Ringshifter's 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.

- In the frequency shifter modes, the Frequency parameter controls the amount of frequency shifting (up and/or down) applied to the input signal.

- In the ring modulator OSC mode, the Frequency 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:* Switch the scaling of the Frequency control:
  - *Exp(onential):* Exponential scaling offers extremely small increments around the 0 point, which is useful for programming slow-moving phasing and tremolo effects.
  - *Lin(ear):* Linear scaling resolution is even across the entire control range.
- *Env Follow slider and field:* Determines the impact of incoming signal levels on the oscillator modulation depth.
- *LFO slider and field:* Determines the amount of oscillator modulation by the LFO.

Using the Ringshifter's Delay

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

- *Time knob and field:* Sets the delay time. This is in Hz when running freely, or in note values (including triplet and dotted notes) when the Sync button is active.

- *Sync button:* Synchronizes the delay to the project tempo. You can choose musical note values with the Time knob.
- *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).

Modulating the Ringshifter with the Envelope Follower

The oscillator Frequency and Dry/Wet parameters can be modulated with the internal envelope follower—and the LFO (see Modulating the Ringshifter with the LFO). The oscillator frequency even allows modulation through the 0 Hz point, thus changing the oscillation direction.

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

- *Power button:* Turns the envelope follower on or off and enables the following parameters.
- *Sens(itivity) slider and field:* Determines how responsive the envelope follower is to the input signal. At lower settings, the envelope follower reacts only to the most dominant signal peaks. At higher settings, the envelope follower tracks 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.

Modulating the Ringshifter with the LFO

The oscillator Frequency and Dry/Wet parameters can be modulated with the LFO—and the envelope follower (see Modulating the Ringshifter with the Envelope Follower). The oscillator frequency even allows modulation through the 0 Hz point, thus changing the oscillation direction. The LFO produces continuous, cycled control signals.

- *Power button:* Turns the LFO on or off and enables the following parameters.
- *Symmetry and Smooth sliders and fields:* These controls, on either side of the Waveform display, change the shape of the LFO waveform.
- *Waveform display:* The LFO waveform display provides visual feedback about the waveform shape.
- *Rate knob and field:* Sets the (waveform cycle) speed of the LFO.
- *Sync button:* Synchronizes the LFO cycles (LFO rate) with the project tempo, using musical note values.

Controlling the Ringshifter Output Parameters

The output parameters are used to set the balance between the effect and input signals and also to set the width and feedback of the Ringshifter.

- *Dry/Wet knob and field:* Sets the mix ratio of the dry input signal and the wet effect signal.

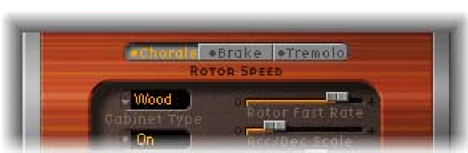
- *Feedback knob and field:* Sets the amount of the signal that is routed back to the effect input. Feedback adds an edge to the Ringshifter sound and is useful for a variety of special effects. It produces a rich phasing sound when used in combination with a slow oscillator sweep. Comb filtering effects are created by using high Feedback settings with a short delay time (less than 10 ms). Use of longer delay times, in conjunction with high Feedback settings, creates continuously rising and falling frequency shift effects.
- *Stereo Width knob and field:* Determines the breadth of the effect signal in the stereo field. Stereo Width affects only the effect signal of the Ringshifter, not the dry input signal.
- *Env Follower slider and field:* Determines the amount of Dry/Wet parameter modulation by the input signal level.
- *LFO slider and field:* Sets the LFO modulation depth of the Dry/Wet parameter.

Rotor Cabinet Effect

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 that pick up the sound.

Basic Rotor Speaker Parameters

The Rotor Cabinet offers the following basic rotor speaker parameters:



- *Rotor Speed buttons:* These switch the rotor speed in the following ways:
  - *Chorale:* Slow movement.
  - *Tremolo:* Fast movement.
  - *Brake:* Stops the rotor.

- *Cabinet Type pop-up menu:* You can choose from the following cabinet models:
  - *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 slightly to the left, and the treble rotor's signal is routed more towards the right.
  - *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 slightly to the left, and the treble rotor signal routed more to the right.

Advanced Rotor Speaker Parameters

The Rotor Cabinet offers the following advanced rotor speaker parameters:

- *Horn Deflector button:* A Leslie cabinet contains a double horn, with a deflector at the horn mouth. This deflector makes the Leslie sound. Some people remove the deflector to increase amplitude modulation, and decrease frequency modulation. You can emulate this by using the Horn Deflector button to switch the deflectors on and off.
- *Motor Ctrl pop-up menu:* You can set different speeds for the bass and treble rotors in the Motor Ctrl pop-up menu:

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

  - *Normal:* Both rotors use the speed determined 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 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 drums are roughly the same. This sounds as if the two are locked, but the effect is clearly audible only during acceleration or deceleration.

- *Rotor Fast Rate slider:* Adjust to set the maximum possible rotor speed (Tremolo). The Tremolo rotation speed is displayed in Hertz.
- *Acc/Dec Scale slider:* The Leslie motors need to physically accelerate and decelerate the speaker horns in the cabinets, and their power to do so is limited. Use the Acc/Dec Scale parameter to determine the time it takes to get the rotors up to a determined speed, and the length of time it takes for them to slow down.
  - Set the slider to the far left to switch to the preset speed immediately.
  - As you drag the slider to the right, it takes more time to hear the speed changes.
  - At the default position (1) the behavior is Leslie-like.

Rotor Cabinet Microphone Parameters

The Rotor Cabinet offers the following microphone parameters:

- *Mic Distance slider:* Determines the distance of the virtual microphones (the listening position) from the emulated speaker cabinet. Use higher values to make the sound darker and less defined. This is typical of microphones when positioned further from the sound source.
- *Mic Angle slider:* Use to define the stereo image, by changing the angle of the simulated microphones.
  - An angle of 0° results in a mono sound.
  - An angle of 180° causes phase cancellations.

Scanner Vibrato Effect

Scanner Vibrato simulates the scanner vibrato section of a Hammond organ. 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.

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:* Use to choose from three Vibrato positions (V1, V2, and V3) or three Chorus positions (C1, C2, and C3).
  - In the Vibrato positions, only the delay line signal is heard, each with different intensities.
  - The three Chorus positions (C1, C2, and C3) 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. This organ-style chorus sounds different from the Logic Express Chorus plug-in.
  - If the C0 setting is chosen, neither the chorus nor vibrato is enabled.
- *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:* When set to a value between 0° and 360°, Stereo Phase determines the phase relationship between left and right channel modulations, thus enabling synchronized stereo effects.

  If you set the knob to "free," 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°, 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."

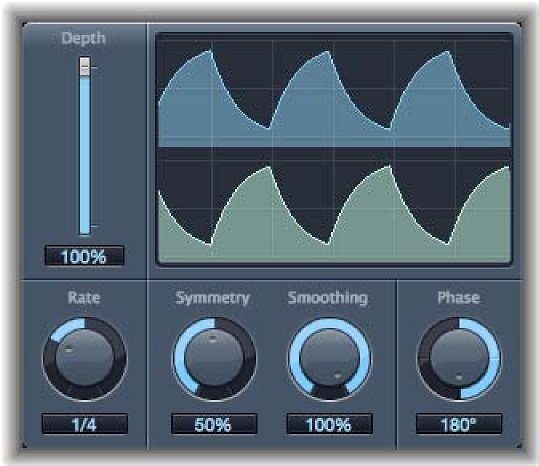
Spreader

Spreader widens the stereo spectrum of a signal. The Spreader effect periodically shifts the frequency range of the original signal, thus changing the perceived width of the signal. The delay between channels can also be specified (in samples), adding to the perceived width and channel separation of a stereo input signal.

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

Tremolo Effect

The Tremolo effect modulates the amplitude of the incoming 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.
- *Waveform display:* Shows the resulting waveform.
- *Rate knob and field:* Sets the frequency of the LFO.
- *Symmetry and Smoothing knobs and fields:* Use these to alter the shape of the LFO waveform.

  If Symmetry is set to 50% and Smoothing to 0%, the LFO waveform has a rectangular shape. This means that the timing of the highest and lowest volume signals is equal, with the switch between both states occurring abruptly.
- *Phase knob and field:* Available only in stereo instances. Controls the phase relationship between the individual channel modulations. At 0, modulation values are reached simultaneously for all channels. Values of 180 or –180 indicate the greatest possible distance between the modulation phases of the channels.
- *Offset slider and field (Extended Parameters area):* Sets the amount that the modulation (cycle) is shifted to the left or right, resulting in subtle or significant tremolo variations.

Pitch Effects

10

You can use the Pitch effects of Logic Express to transpose or correct the pitch of audio signals. These effects can also be used for creating unison or slightly thickened parts, or even for creating harmony voices.

This chapter covers the following:

Pitch Correction Effect

You can use the Pitch Correction effect to correct the pitch of incoming audio signals. 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.

Pitch correction works by accelerating and slowing down the audio playback speed, ensuring that the input signal (sung vocal) always matches the correct note pitch. If you try to correct larger intervals, you can create special effects. Natural articulations of the performance, such as breath noises, are preserved. Any scale can be defined as a pitch reference (technically speaking, this is known as a *pitch quantization grid*), with improperly intonated notes corrected in accordance with this scale.

Note: Polyphonic recordings, such as choirs, 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 signals!

Pitch Correction Parameters

The Pitch Correction effect offers the following parameters.

- *Use Global Tuning button:* Enable to use the project's Tuning settings for the pitch correction process. If disabled, you can use the Ref. Pitch field to freely set the desired reference tuning. See Setting the Pitch Correction Reference Tuning.
- *Normal and Low buttons:* These determine the pitch range that is scanned (for notes that need correction). See Defining the Pitch Correction Effect's Quantization Grid.
- *Ref. Pitch field:* Sets the desired reference tuning, in cents (relative to the root). See Setting the Pitch Correction Reference Tuning.
- *Root pop-up menu and field:* Click to choose the root note of the scale from the Root pop-up menu. See Defining the Pitch Correction Effect's Quantization Grid.
- *Scale pop-up menu and field:* Click to choose different pitch quantization grids from the Scale pop-up menu. See Defining the Pitch Correction Effect's Quantization Grid.
- *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). See Excluding Notes from Pitch Correction.
- *Byp(ass) 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 built-in scale quantization grids. See Excluding Notes from Pitch Correction.
- *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. You can use the display when discussing (and optimizing) the vocal intonation with a singer during a recording session.
- *Response slider and field:* Determines how quickly the voice reaches the corrected destination 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 low, the pitch of the output signal won't change quickly enough. The optimum setting for this parameter depends on the singing style, tempo, vibrato, and accuracy of the original performance.
- *Detune slider and field:* Detunes the output signal by the set value.
- *Input Detune slider and field (Extended Parameters area):* Detunes the input signal by the set value, thus affecting it before any pitch correction takes place. This parameter is of particular benefit when automated.

Defining the Pitch Correction Effect's Quantization Grid

Use the Pitch Correction effect's 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 be used only for audio material that contains extremely low frequencies (below 100 Hz), which may result in inaccurate pitch detection. These parameters have no effect on the sound; they are simply optimized tracking options for the chosen target pitch range.

The Scale pop-up menu allows you to choose different pitch quantization grids. The scale that is set manually (with the keyboard graphic in the plug-in window) is called the User Scale. The default setting is the *chromatic* scale. If you're unsure of the intervals used in any given scale, choose it in the Scale menu and look at the keyboard graphic. You can alter any note in the chosen scale by clicking the keyboard keys. Any such adjustments 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, and the *single* scale defines a single note. Neither of these scales is meant to result in realistic singing voices, so if you're after interesting effects, you should give them both a try.

Open the Root pop-up menu to choose the root note of the scale. (If you chose user scale or chromatic in the Scale pop-up menu, the Root pop-up menu is non-functional.) You may freely transpose the major and minor scales, and scales named after chords.

Excluding Notes from Pitch Correction

You can use the Pitch Correction effect's onscreen keyboard to exclude notes from the pitch quantization grid. When you first open the effect, 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 to an unwanted pitch. For example, the singer may 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. Because the original pitch was sung closer to an E than a 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.

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 buttons 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 enabling or disabling in all situations.

Tip: You'll often find that it's best to correct only the notes with the most harmonic gravity. For example, choose "sus4" from the Scale pop-up menu, 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. Activate the bypass buttons for all other notes and only the most important and sensitive notes will be corrected, while all other singing remains untouched.

Setting the Pitch Correction Reference Tuning

In Logic Express, the settings that appear when you choose File > Project Settings > Tuning determine the tuning reference for all software instruments. In MainStage, choose MainStage > Preferences > Tuning (in the General tab).

If you turn on the Use Global Tuning button in the Pitch Correction window, the host application Tuning settings will be used for the pitch correction process. If this parameter is turned off, you can use the Ref. Pitch field to freely set the desired reference tuning (to the root key/note).

For 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 prevent this by (de)tuning the pitch corrections individually.

Automating the Pitch Correction Effect

The Pitch Correction effect can be fully automated. This means that you can automate the Scale and Root parameters to follow harmonies in the project. Depending on the accuracy of the original intonation, setting the appropriate key (Scale parameter) may suffice. Less precise intonations may need more significant changes to the Scale and Root parameters.

Pitch Shifter II

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

- *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 (1/100th of a semitone).
- *Drums, Speech, and Vocals buttons:* Select one of three optimized algorithms for common types of audio material:
  - *Drums:* Maintains the groove (rhythmic feel) of the source signal.
  - *Speech:* Provides a balance between 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.
  - *Vocals:* Retains the intonation of the source, making it well-suited for signals that are inherently harmonic or melodious, such as string pads.
- *Mix slider and field:* Sets the balance between the effect and original signals.

- *Timing pop-up menu (Extended Parameters area):* Determines how timing is derived: by following the selected algorithm (Preset), by analyzing the incoming signal (Auto), or by using the settings of the Delay, Crossfade, and Stereo Link parameters, described below (Manual).

Note: The following three parameters are active only when "Manual" is chosen in the Timing pop-up menu.

- *Delay slider and field (Extended Parameters area):* Sets the amount of delay applied to the input signal. The lower 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 (Extended Parameters area):* Sets the range (expressed as a percentage of the original signal) used to analyze the input signal.
- *Stereo Link radio buttons (Extended Parameters area):* Select Inv. to invert the stereo channel's signals, with processing for the right channel occurring on the left, and vice versa. Select Normal to leave the signal as it is.

Follow these steps when pitch shifting

1. Set the Semi Tones slider for the amount of transposition, or pitch shift.
2. Set the Cents slider for the amount of detuning.
3. Click the Drums, Speech, or Vocals button to select the algorithm that best matches the material you are working with.

   If you are working with material that doesn't fit any of these categories, experiment with each of the algorithms (starting with Speech), compare the results, and use the one that best suits your material.

   Tip: While auditioning and comparing different settings, it's often a good idea to temporarily set the Mix parameter to 100%, as Pitch Shifter II artifacts are easier to hear.

Vocal Transformer

The Vocal Transformer can be used 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 that mirrors the pitches of a melody. No matter how you change the pitches of the melody, the constituent parts of the signal (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. 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 Vocal Transformer is well suited to extreme vocal effects. The best results are achieved with monophonic signals, including monophonic instrument tracks. It is not designed for polyphonic voices—such as a choir on a single track—or other chordal tracks.

Vocal Transformer Parameters

The Vocal Transformer offers the following parameters.

- *Pitch knob and field:* Determines the amount of transposition applied to the input signal. See Setting Vocal Transformer Pitch and Formant Parameters.
- *Robotize button:* Enables Robotize mode, which is used to augment, diminish, or mirror the melody. See Using Vocal Transformer's Robotize Mode.
- *Pitch Base slider and field (available only in Robotize mode):* Use to transpose the note that the Tracking parameter (see below) is following. See Using Vocal Transformer's Robotize Mode.
- *Tracking slider, field, and buttons (available only in Robotize mode):* Control how the melody is changed in Robotize mode. See Using Vocal Transformer's Robotize Mode.
- *Mix slider and field:* Defines the level ratio between the original (dry) and effect signals.
- *Formant knob and field:* Shifts the formants of the input signal. See Setting Vocal Transformer Pitch and Formant Parameters.
- *Glide slider and field (Extended Parameters area):* Determines the amount of time the vocal transformation takes, allowing sliding transitions to the set Pitch value.
- *Grain Size slider and field (Extended Parameters area):* The Vocal Transformer effect 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.
- *Formants pop-up menu (Extended Parameters area):* Determines whether the Vocal Transformer processes all formants ("Process always" setting), or only the voiced ones ("Keep Unvoiced Formants" setting). The "Keep Unvoiced Formants" option leaves sibilant sounds in a vocal performance untouched. This setting will produce a more natural-sounding transformation effect with some signals.
- *Detune slider and field (Extended Parameters area):* Detunes the input signal by the set value. This parameter is of particular benefit when automated.

Setting Vocal Transformer Pitch and Formant Parameters

Use the Vocal Transformer's Pitch parameter to transpose the pitch of the signal upward or downward. 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 downward, 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 turned 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 doubling of voices is quite effective, with levels easily controlled by the Mix parameter.

Using Vocal Transformer's Robotize Mode

When Robotize is enabled, Vocal Transformer can augment or diminish the melody. You can control the intensity of this distortion with the Tracking parameter.

The Tracking slider and field feature is enhanced by four buttons which immediately set the slider to the most useful values, as follows:

- *–1 (sets the slider to –100%):* All intervals are mirrored.
- *0 (sets the slider to 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.
- *1 (sets the slider to 100%):* The range of the melody is maintained. Higher values augment, and lower values diminish, the melody.
- *2 (sets the slider to 200%):* The intervals are doubled.

The Pitch Base parameter is used to transpose the note that the Tracking parameter is following. As an example: With Tracking set to 0%, the pitch of the (spoken) note will be transposed to the chosen base pitch value.

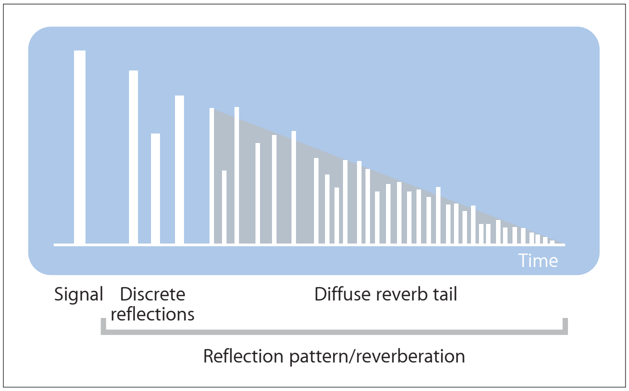
Reverb Effects

11

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

Sound waves repeatedly bounce off the surfaces—walls, ceilings, windows, and so on—of any space, or off objects within a space, gradually dying out until they are inaudible. These bouncing sound waves result in a reflection pattern, more commonly known as a reverberation (or *reverb*).

The starting portion of a reverberation signal consists of a number of discrete reflections that you can clearly discern before the diffuse reverb tail builds up. These early reflections are essential in human perception of spatial characteristics, such as the size and shape of a room.



This chapter covers the following:

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 metal plates and springs, were also 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 differences 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.

Today's computers make 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 characteristics.

AVerb

AVerb is a simple reverb effect that employs a single parameter (Density/Time) to control both the early reflections and diffuse reverb tail. It is a quick-and-easy tool for creating a range of interesting space and echo effects. The AVerb may not be the best choice for simulating real acoustic environments, however.

- *Predelay slider and field:* Determines the time between the original signal and the early reflections of the reverb signal.
- *Reflectivity knob and field:* Defines how reflective the imaginary walls, ceiling, and floor are—in other words, how hard the walls are, and what they're made of. Glass, stone, timber, carpet, and other materials have a dramatic impact on the tone of the reverb.
- *Room Size knob and field:* Defines the dimensions of simulated rooms.

- *Density/Time slider and field:* Determines both the density and duration of the reverb.

  Low values tend to generate clearly discernible early reflection clusters, generating something similar to an echo. High values result in a more reverb-like effect.
- *Mix slider and field:* Sets the balance between the effect (wet) and direct (dry) signals.

EnVerb

EnVerb is a versatile reverb effect with a unique feature: It allows you to freely adjust the envelope—the shape—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 they change the reverb tail over time. The graphic display visually represents the levels over time (the envelope) of the reverb. See EnVerb Time Parameters.
- *Sound parameters:* This area allows you to shape the sound of the reverb signal. You can also split the incoming signal into two bands—with the Crossover parameter—and set the level of the low frequency band separately. See EnVerb Sound Parameters.
- *Mix parameter:* Determines the balance between the effect (wet) and direct (dry) signals.

EnVerb Time Parameters

EnVerb offers the following Time parameters:

- *Dry Signal Delay slider and field:* Determines the delay of the original signal. You can hear the dry signal only when the Mix parameter is set to a value other than 100%.
- *Predelay knob and field:* Sets the time between the original signal and the starting point of the reverb attack phase—the very beginning of the first reflection.
- *Attack knob and field:* Defines the time it takes for the reverb to climb to its peak level.
- *Decay knob and field:* Defines the time it takes for the level of the reverb to drop from its peak to the sustain level.
- *Sustain knob and field:* 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.
- *Hold knob and field:* Sets the duration—the time—of the sustain phase.
- *Release knob and field:* Sets the time that the reverb takes to fade out completely, after it has completed the sustain phase.

EnVerb Sound Parameters

EnVerb offers the following tone control parameters:

- *Density slider and field:* Sets the reverb density.
- *Spread slider and field:* 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 slider and field:* Frequencies above the set value are filtered out of the reverb tail.
- *Crossover slider and field:* Defines the frequency that is used to split the input signal into two frequency bands, for independent processing.
- *Low Freq Level slider and field:* Determines the relative level of (reverb signal) frequencies below the crossover frequency. In most cases you get better-sounding results when you set negative values for this parameter.

GoldVerb

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

The interface is broken down into four parameter areas:

- *Early Reflections parameters:* Used to emulate the original signal's first reflections as they bounce off the walls, ceiling, and floor of a natural room. See GoldVerb Early Reflections Parameters.
- *Reverb parameters:* Control the diffuse reverberations. See GoldVerb Reverb Parameters.
- *Balance ER/Reverb slider:* Controls the balance between the early reflections and reverb signal. When the slider is set to either extreme position, the other signal is not heard.
- *Mix slider and field:* Determines the balance between the effect (wet) and direct (dry) signals.

GoldVerb Early Reflections Parameters

The GoldVerb offers the following Early Reflections parameters:

- *Predelay slider and field:* Determines the amount of time between the start of the original signal and the arrival of the early reflections. Extremely short Predelay settings can color the sound and make it difficult to pinpoint the position of the signal source. Overly long Predelay settings can be perceived as an unnatural echo and can divorce the original signal from its early reflections, leaving an audible gap between them.

  The optimum Predelay setting depends on the type of input signal—or more precisely, the envelope of the input signal. Percussive signals generally require shorter predelays than signals where the attack fades in gradually. A good working method is to use the longest possible Predelay value before you start to hear undesirable side effects, such as an audible echo. When you reach this point, reduce the Predelay setting slightly.
- *Room Shape slider and field:* 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 slider and field:* Determines the dimensions of the room. The numeric value indicates the length of the room's walls—the distance between two corners.
- *Stereo Base slider and field:* Defines the distance between the two virtual microphones that are used to capture the signal in the simulated room.

  Note: Spacing the microphones slightly farther apart than the distance between two human ears generally delivers the best, and most realistic, results. This parameter is available only in stereo instances of the effect.

GoldVerb Reverb Parameters

The GoldVerb offers the following Reverb parameters:

- *Initial Delay slider and field:* Sets the time between the original signal and the diffuse reverb tail. If you're going for a natural-sounding, harmonic reverb, the transition between the early reflections and the reverb tail should be as smooth and seamless as possible. Set the Initial Delay parameter so that it is as long as possible, without a noticeable gap between the early reflections and the reverb tail.
- *Spread slider and field:* 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 knob and field:* Frequencies above the set value are filtered from the reverb signal. Uneven or absorbent surfaces—wallpaper, wood paneling, carpets, and so on–tend to reflect lower frequencies better than higher frequencies. The High Cut filter mimics this effect. If you set the High Cut filter so that it is wide open (maximum value), the reverb will sound as if it is reflecting off stone or glass.
- *Density knob and field:* Controls the density of the diffuse reverb tail. Ordinarily you want the signal to be as dense as possible. In rare instances, however, a high Density value can color the sound, which you can fix by reducing the Density knob value. Conversely, if you select a Density value that is too low, the reverb tail will sound grainy.
- *Reverb Time knob and field:* Time it takes for the reverb level to drop by 60 dB—often indicated as RT60. Most natural rooms have a reverb time somewhere in the range of 1 to 3 seconds. This time is reduced by absorbent surfaces, such as carpet and curtains, and soft or dense furnishings, such as sofas, armchairs, cupboards, and tables. Large empty halls or churches have reverb times of up to 8 seconds, with some cavernous or cathedral-like venues extending beyond that.
- *Diffusion slider and field (Extended Parameters area):* Sets the diffusion of the reverb tail. High Diffusion values represent a regular density, with few alterations in level, times, and panorama position over the course of the diffuse reverb signal. Low Diffusion values result in the reflection density becoming irregular and grainy. This also affects the stereo spectrum. As with Density, find the best balance for the signal.

PlatinumVerb

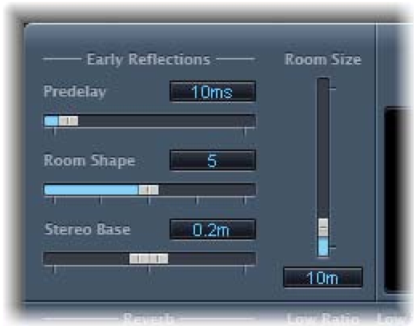
The PlatinumVerb allows you to edit both the early reflections and diffuse reverb tail separately, making it easy to precisely emulate real rooms. Its dual-band Reverb section splits the incoming signal into two bands, each of which is processed and can be edited separately.

The interface is broken down into four parameter areas:

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

PlatinumVerb Early Reflections Parameters

The PlatinumVerb offers the following Early Reflections parameters:



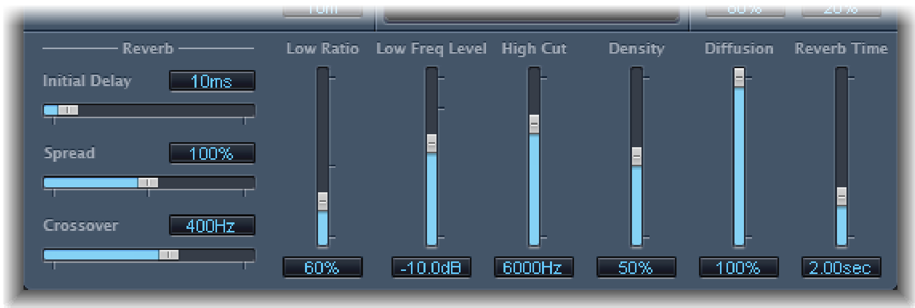
- *Predelay slider and field:* Determines the amount of time between the start of the original signal and the arrival of the early reflections. Extremely short Predelay settings can color the sound and make it difficult to pinpoint the position of the signal source. Overly long Predelay settings can be perceived as an unnatural echo and can divorce the original signal from its early reflections, leaving an audible gap between them.

  The optimum Predelay setting depends on the type of input signal—or more precisely, the envelope of the input signal. Percussive signals generally require shorter predelays than signals where the attack fades in gradually. A good working method is to use the longest possible Predelay value before you start to hear undesirable side effects, such as an audible echo. When you reach this point, reduce the Predelay setting slightly.
- *Room Shape slider and field:* 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 slider and field:* Determines the dimensions of the room. The numeric value indicates the length of the room's walls—the distance between two corners.
- *Stereo Base slider and field:* Defines the distance between the two virtual microphones that are used to capture the signal in the simulated room.

  Note: Spacing the microphones slightly farther apart than the distance between two human ears generally delivers the best, and most realistic, results. This parameter is available only in stereo instances of the effect.
- *ER Scale slider and field (Extended Parameters area):* Scales the early reflections along the time axis, influencing the Room Shape, Room Size, and Stereo Base parameters simultaneously.

PlatinumVerb Reverb Parameters

The PlatinumVerb offers the following Reverb parameters:



- *Initial Delay slider and field:* Sets the time between the original signal and the diffuse reverb tail.
- *Spread slider and field:* Controls the stereo image of the reverb. At 0%, the effect generates a monaural reverb. At 200%, the stereo base is artificially expanded.
- *Crossover slider and field:* Defines the frequency at which the input signal is split into two frequency bands, for separate processing.
- *Low Ratio slider and field:* Determines the relative reverb times of the bass and high bands. It is expressed as a percentage. At 100%, the reverb time of the two bands is identical. At values below 100%, the reverb time of frequencies below the crossover frequency is shorter. At values greater than 100%, the reverb time for low frequencies is longer.
- *Low Freq Level slider and field:* Sets the level of the low frequency reverb signal. At 0 dB, the volume of the two bands is equal. In most mixes, you should set a lower level for the low frequency reverb signal. This enables you to boost the bass level of the incoming signal, making it sound punchier. This also helps to counteract bottom-end masking effects.
- *High Cut slider and field:* Frequencies above the set value are filtered from the reverb signal. 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 (maximum value), the reverb will sound as if it is reflecting off stone or glass.
- *Density slider and field:* Controls the density of the diffuse reverb tail. Ordinarily you want the signal to be as dense as possible. In rare instances, however, a high Density value can color the sound, which you can fix by reducing the Density slider value. Conversely, if you select a Density value that is too low, the reverb tail will sound grainy.

- *Diffusion slider and field:* Sets the diffusion of the reverb tail. High Diffusion values represent a regular density, with few alterations in level, times, and panorama position over the course of the diffuse reverb signal. Low Diffusion values result in the reflection density becoming irregular and grainy. This also affects the stereo spectrum. As with Density, find the best balance for the signal.
- *Reverb Time slider and field:* Determines the reverb time of the high band. Most natural rooms have a reverb time somewhere in the range of 1 to 3 seconds. This time is reduced by absorbent surfaces, such as carpet and curtains, and soft or dense furnishings, such as sofas, armchairs, cupboards, and tables. Large empty halls or churches have reverb times of up to 8 seconds, with some cavernous or cathedral-like venues extending beyond that.

PlatinumVerb Output Parameters

The PlatinumVerb offers the following Output parameters:

- *Dry slider and field:* Controls the amount of the original signal.
- *Wet slider and field:* Controls the amount of the effect signal.

SilverVerb

The SilverVerb is similar to the AVerb, but it provides an additional LFO that can modulate the reverberated signal. It also includes a high cut 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 result in an indistinct mix.

- *Predelay slider and field:* Determines the time between the original signal and the reverb signal.
- *Reflectivity slider and field:* Defines how reflective the imaginary walls, ceiling, and floor are.
- *Room Size slider and field:* Defines the dimensions of a simulated room.
- *Density/Time slider and field:* Determines both the density and the duration of the reverb.
- *Low Cut slider and field:* Frequencies below the set value are filtered out of the reverb signal. This affects only the tone of the reverb signal, not the original signal.
- *High Cut slider and field:* Frequencies above the set value are filtered out of the reverb signal. This affects only the tone of the reverb signal, not the original signal.
- *Mod(ulation) Rate knob and field:* Sets the frequency (the speed) of the LFO.
- *Mod(ulation) Phase knob and field:* 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(ulation) Intensity slider and field:* Sets the modulation amount. A value of 0 turns the delay modulation off.
- *Mix slider and field:* Sets the balance between the effect (wet) and original (dry) signals.

Specialized Effects and Utilities

12

Logic Express includes a bundle of specialized effects and utilities designed to address tasks often encountered during audio production. As examples of where these processors can help: Denoiser eliminates or reduces noise below a threshold level. Enhance Timing enhances the timing of audio recordings. Exciter can add life to your recordings by generating artificial high frequency components. Grooveshifter enabes you to create rhythmic variations in your recordings. SubBass generates an artificial bass signal that is derived from the incoming signal.

This chapter covers the following:

Denoiser

The Denoiser eliminates or reduces any noise below a threshold volume level. The Denoiser uses fast Fourier transform (FFT) analysis to recognize frequency bands of lower volume and less complex harmonic structure. It then reduces these low-level, less complex bands to the appropriate dB level. See Denoiser Main Parameters.

If you use the Denoiser too aggressively, however, the algorithm produces artifacts, which are usually less desirable than the existing noise. If using the Denoiser produces these artifacts, you can use the three Smoothing knobs to reduce or eliminate them. See Denoiser Smoothing Parameters.

To use the Denoiser

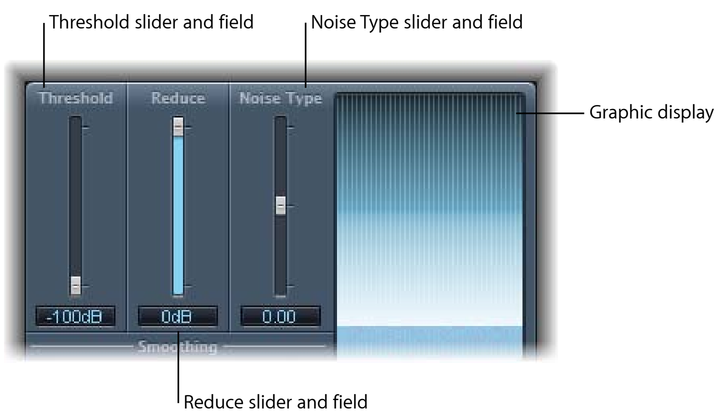
1. Locate a section of the audio where only noise is audible, and set the Threshold value so that only signals at, or below, this level are filtered out.

2 Play the audio signal and set the Reduce value to the point where noise reduction is optimal but little of the appropriate signal is reduced.

3 If you encounter artifacts, use the smoothing parameters.

Denoiser Main Parameters

The Denoiser offers the following main parameters:



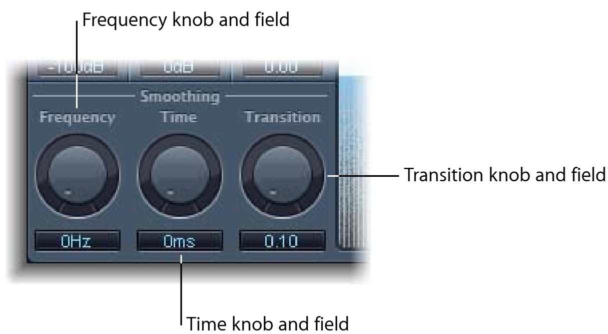
- *Threshold slider and field:* Sets the threshold level. Signals that fall below this level are reduced by the Denoiser.
- *Reduce slider and field:* Sets the amount of noise reduction applied to signals that fall below the threshold. When reducing noise, remember that each 6 dB reduction is equivalent to halving the volume level (and each 6 dB increase equals a doubling of the volume level).

Note: 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 doesn't 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:* Determines the type of noise that 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 (hissy tape 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.

Denoiser Smoothing Parameters

The Denoiser offers the following smoothing parameters:



- *Frequency knob and field:* 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 parameter, the more it changes the neighboring frequency bands to avoid glass noise.
- *Time knob and field:* Sets the time required by the Denoiser to reach (or release) maximum reduction. This is the simplest form of smoothing.
- *Transition knob and field:* 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 parameter, the more similar-level values are changed, in order to avoid glass noise.

Enhance Timing

Enhance Timing is designed to tighten up loose playing of recorded audio in a production. It can be used on a variety of materials and works in real time.

While effective on suitable material, this type of real-time quantization has some limitations. It does 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, however, 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. Therefore, you should try to strike a balance between sound quality and timing enhancement.

Important: For technical reasons, the Enhance Timing plug-in works only on audio channel strips and must be inserted in the *top* Insert slot.

- *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 Note Grid pop-up menu, are corrected.
- *Note Grid pop-up menu:* Provides a choice of four grid divisions. The grid divisions serve as reference points for the timing correction process. As a tip for eighth-note triplets, try the 1/12 note setting.

Exciter

The Exciter generates high frequency components that are not part of the original signal. It does this by employing a nonlinear distortion process that resembles overdrive and distortion effects.

Unlike these effects, however, the Exciter passes the input signal through a highpass filter before feeding it into the harmonics (distortion) generator. This results in artificial harmonics being added to the original signal. These added harmonics contain 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 recordings. It is especially well suited to audio tracks with a weak treble frequency range. The Exciter is also useful as a general tool for enhancing guitar tracks.

- *Frequency display:* Shows the frequency range used as the source signal for the excite process.
- *Frequency slider and field:* Sets the cutoff frequency (in Hertz) of the highpass filter. The input signal passes through the filter before (harmonic) distortion is introduced.
- *Input button:* When the Input button is active, the original (pre-effect) signal is mixed with the effect signal. If you disable Input, only the effect signal is heard.
- *Harmonics knob and field:* Sets the ratio between the effect and original signals, expressed as a percentage. If the Input button is turned off, this parameter has no effect.

  Note: 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:* Color 1 generates a less dense harmonic distortion spectrum. Color 2 generates a more intense harmonic distortion. Color 2 also introduces more (unwanted) intermodulation distortions.

Grooveshifter

The Grooveshifter allows you to rhythmically vary audio recordings, imparting a swing feel to the input signal. 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 automatically follows all changes to the project tempo.

Note: The Grooveshifter relies on perfect matching of the project tempo with the tempo of the treated recording. Any tempo variations deliver less precise results.

Grooveshifter Source Material Parameters

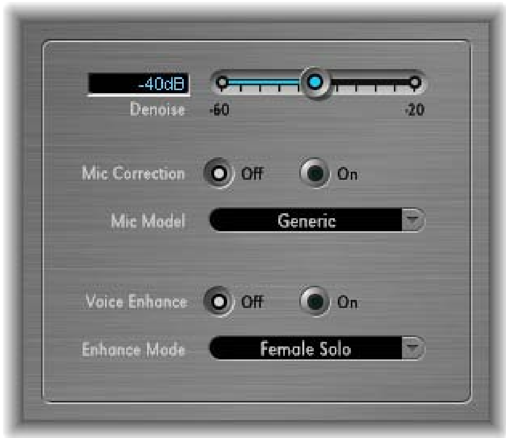
- *Beat and Tonal buttons:* Switch between two algorithms, each optimized for different types of audio material.
  - *Beat algorithm:* Optimized for percussive input material. The Grain Size slider has no effect when Beat is chosen.
  - *Tonal algorithm:* Optimized for tonal input material. Because this algorithm is based on granular synthesis, it offers an additional Grain Size slider.
- *Grain Size slider and field:* Sets the size of the grains—technically-speaking, this determines the analysis precision. The (default) Auto setting automatically derives a suitable grain size value from the incoming signal.

Grooveshifter Swing Parameters

- *Grid buttons:* Determine the beat division used as a timing reference by the algorithm when analyzing the audio material.
  - Choose 1/8 if the audio material contains primarily eighth notes, and choose 1/16 if it consists mostly of sixteenth notes.
- *Swing slider and field:* Determines the amount that even beats are 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.
- *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 Speech Enhancer 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 increasingly suppress background noise but also proportionately increase artifacts.
- *Mic Correction buttons:* Activate the On button to improve the frequency response of recordings made with your built-in microphone. This creates the impression that an up-market microphone was used.
- *Mic Model pop-up menu:* Provides a choice of several microphone models that compensate for tonal characteristics of particular built-in Macintosh microphones.

  Note: You can use the Speech Enhancer effect with other microphones, but microphone correction models are offered only for built-in Macintosh microphones and iSight. Should a non-Apple microphone be used, you will achieve the best results if Mic Correction is set to Generic.
- *Voice Enhance button and Enhance Mode pop-up menu:* This button turns on the Speech Enhancer multiband compression circuit. When it is active, you can choose from four settings that make the recorded voice louder and more intelligible. Choose the setting that best matches your recording situation.
  - *(Female or Male) Solo:* Use when the recorded signal consists of a vocal only.
  - *(Female or Male) Voice Over:* Use when the recorded signal contains both a vocal performance and a musical or atmospheric bed.

SubBass

The SubBass plug-in generates frequencies below those of the original signal, resulting in artificial bass content.

The simplest use for the SubBass is as an octave divider, similar to octaver effect pedals for electric bass guitars. Whereas such pedals can only process a monophonic input sound source of clearly defined pitch, SubBass can be used with complex summed signals as well. See Using SubBass.

SubBass creates two bass signals, derived from two separate portions of the incoming signal. These are defined with the High and Low parameters. See SubBass Parameters.

> *Warning:* Using 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

The SubBass offers the following parameters.

- *High Ratio knob and field:* Adjusts the ratio between the generated signal and the original upper band signal.
- *High Center knob and field:* Sets the center frequency of the upper band.

- *High Bandwidth knob and field:* Sets the width of the upper band.
- *Graphic display:* Shows the selected upper and lower frequency bands.
- *Freq. Mix slider and field:* Adjusts the mix ratio between the upper and lower frequency bands.
- *Low Ratio knob and field:* Adjusts the ratio between the generated signal and the original lower band signal.
- *Low Center knob and field:* Sets the center frequency of the lower band.
- *Low Bandwidth knob and field:* Sets the width of the lower band.
- *Dry slider and field:* Sets the amount of dry (non-effect, original) signal.
- *Wet slider and field:* Sets the amount of wet (effect) signal.

Using SubBass

Unlike a pitch shifter, the waveform of the signal generated by SubBass is not based on the waveform of the input signal, but is sinusoidal—that is, 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 with the Wet and Dry sliders.

Use the High and Low parameters to define the two frequency bands, which SubBass uses to generate tones. High Center and Low Center define the center frequency of each band, and High Bandwidth and Low Bandwidth define the width of each frequency band.

The High Ratio and Low Ratio knobs define the transposition amount for the generated signal in each band. This is expressed as a ratio of the original signal. For 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 SubBass, and compare the extreme low frequency content of your mixes with other productions. It is very easy to go overboard with it.

Utilities and Tools

13

The tools found in the Utility category can help with routine tasks and situations that you may encounter during production, such as the following: Gain plug-ins are used to adjust the level or phase of input signals. I/O Utility enables you to integrate external audio effects into your host application mixer. Test Oscillator generates a static frequency or sine sweep.

This chapter covers the following:

Gain Plug-in

Gain amplifies (or reduces) the signal by a specific decibel amount. It is very useful for quick level adjustments when you are working with automated tracks during post-processing—for example, when you have inserted an 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 slider and field:* Sets the amount of gain.
- *Phase Invert Left and Right buttons:* Invert the phase of the left and right channels, respectively.

- *Balance knob and field:* Adjusts the balance of the incoming signal between the left and right channels.
- *Swap L/R (Left/Right) button:* Swaps the left and right output channels. The swapping occurs after the Balance parameter in the signal path.
- *Mono button:* Outputs the summed mono signal on both the left and right channels.

Note: The Gain plug-in is available in mono, mono to stereo, and stereo instances. In mono and mono to stereo modes, only one Phase Invert button is available. In the mono version, the Stereo Balance, Swap Left/Right, and Mono parameters are disabled.

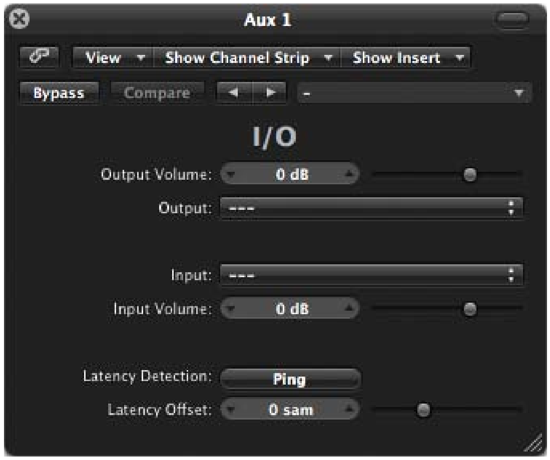
Using Phase Inversion

Inverting phase is useful for dealing with time alignment problems, particularly those caused by simultaneous recording with multiple microphones. 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 may have an audible effect. For example, if you place microphones above and below a snare drum, you may find that inverting the phase of either microphone can improve (or ruin) the sound. As always, rely on your ears.

I/O Utility

The I/O utility enables the use of external audio effects units, similar to the use of internal Logic Express effects.

Note: In practical terms, this makes sense only if you are using an audio interface that provides discrete inputs and outputs (analog or digital) that are used to send signals to and from the external audio effects unit.



- *Output Volume field and slider:* Adjusts the level of the output signal.
- *Output pop-up menu:* Assigns the respective output (or output pair) of your audio hardware.
- *Input pop-up menu:* Assigns the respective input (or input pair) of your audio hardware.

  Note: The Input pop-up menu is only visible when an audio interface with multiple inputs is active.
- *Input Volume field and slider:* Adjusts the level of the input signal.
- *Latency Detection (Ping) button:* Detects the delay between the selected output and input, and compensates the delay accordingly.

  Note: Bypassing any latency-inducing plug-ins on the track will provide you with the most accurate reading.
- *Latency Offset field and slider:* Displays the value for the detected latency between the selected output and input. Also allows you to offset the latency manually.

To integrate and use an external effects unit with the I/O utility

1. Connect an output (or output pair) of your audio interface with the input (pair) on your effects unit. 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.
2. Click an Insert slot of an aux channel strip (being used as a bus send/return), and choose Utility > I/O.

3 In the I/O window, choose both the Outputs and Inputs of your audio hardware (that your effects unit is connected to).

4 Route the signals of any channel strips that you want to process to the bus (aux channel strip) chosen in step 3, and set appropriate Send levels.

5 Adjust the Input or Output volume as required in the I/O window.

6 Click the Latency Detection (Ping) button if you want to detect, and compensate for, any delay between the selected output and input.

When you start playback, the signals of any channel strips routed to the aux channel (chosen in step 3) will be processed by the external effects unit.

Test Oscillator

The Test Oscillator is useful for tuning studio equipment and instruments, and can be inserted as both an instrument or effect plug-in. It operates in two modes, generating either a static frequency or a sine sweep.

In the first mode (default mode), it starts generating the test signal as soon as it is inserted. You can switch it off by bypassing it. In the second mode (activated by clicking the Sine Sweep button), Test Oscillator generates a user-defined frequency spectrum tone sweep—when triggered with the Trigger button.

- *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 are disabled.

- *Frequency knob and field:* Determines the frequency of the oscillator (default is 1 kHz).
- *Sine Sweep button:* Generates a sine wave sweep (of the frequency spectrum you set with the Start Freq and End Freq fields).
- *Time field:* Sets the duration of the sine wave sweep.
- *Start Freq and End Freq fields:* Drag vertically to define the oscillator frequency at the beginning and end of the sine sweep.
- *Sweep Mode pop-up menu (Extended Parameters area):* Choose Linear or Logarithmic (sweep curve).
- *Trigger button and pop-up menu:* Click the Trigger button to trigger the sine sweep. Choose the behavior of the Trigger button in the pop-up menu:
  - *Single:* Triggers the sweep once.
  - *Continuous:* Triggers the sweep indefinitely.
- *Level slider and field:* Determines the overall output level of the Test Oscillator.