MainStage 3 Effects

For OS X

Contents

187 **Chapter 10: Modulation effects**

Amps and pedals

<div style="text-align: right; font-size: 2em; font-weight: bold;">1</div>

Amps and pedals overview

MainStage 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 recreate vintage and modern tube and solid-state amps. Built-in effect units, such as reverb, tremolo, or vibrato, are also reproduced. The modeled amplifiers can be paired with a number of emulated speaker cabinets. These amplifiers and speaker cabinets can be used as a matching set or combined in other 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 chain pedals in any order to create your sound.

Amp Designer

Amp Designer overview

Amp Designer emulates the sound of more than 20 famous guitar amplifiers and the speaker cabinets used with them. Each preconfigured model combines an amp, a cabinet, and EQ that recreates a well-known guitar amplifier sound. You can process guitar signals directly, reproducing the sound of your guitar played through these amplification systems. You can also use Amp Designer for experimental sound design and processing. You can use it with other instruments as well, 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 numerous ways to alter the tone. Virtual microphones are used to pick up the signal of the emulated amplifier and cabinet. You can choose from, and position, seven different microphone types. Amp Designer also emulates classic guitar amplifier effects, including spring reverb, vibrato, and tremolo.

The Amp Designer interface is divided into four main parameter sections.
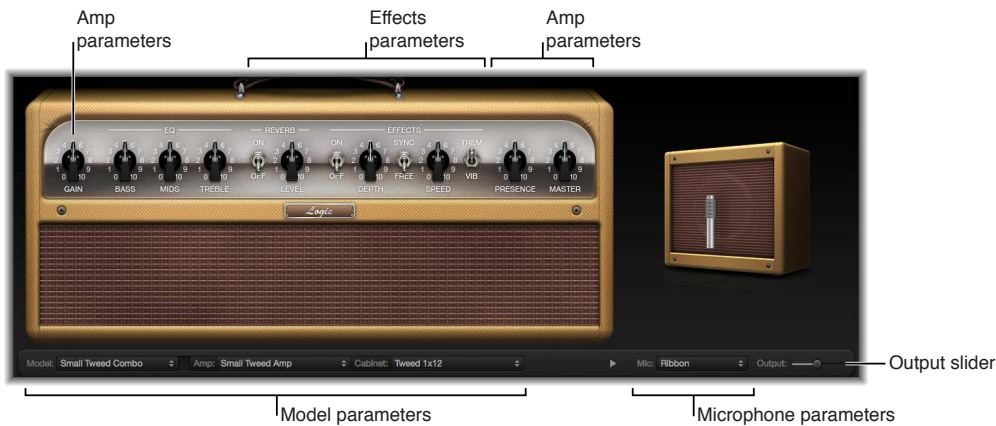
- *Model parameters:* The Model pop-up menu in the black bar at the bottom is used to choose a preconfigured model, consisting of an amplifier, a cabinet, an EQ type, and a microphone type. The other pop-up menus in the black bar enable you to independently choose the type of amplifier, cabinet, and microphone. See Build a custom Amp Designer combo on page 19.

- *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 Amp Designer amplifier controls on page 23.

- *Effects parameters:* Located in the center of the knobs section, these parameters control the integrated effects. See Amp Designer effects overview on page 24.

- *Microphone parameters:* Located at the right of the interface, these parameters set the type and position of the microphone that captures the amplifier and cabinet sound. See Amp Designer microphone parameters on page 26.

- *Output slider:* The Output slider (or the Output field, in the small interface) is found at the lower-right corner of the interface. It serves as the final level control for Amp Designer's 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.

Switch between the full and small 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.

  In the small interface you can access all parameters except microphone selection and positioning.

- To switch back to the full interface, click the disclosure triangle beside the Output field in the small interface.

Click here in
full interface.

Click here in
small interface.

Choose an Amp Designer model

You can use the Model pop-up menu to choose a preconfigured model, or you can build a customized model using the Amp, Cabinet, and Mic pop-up menus. See Build a custom Amp Designer combo. Your choices remain visible in the pop-up menus and they are also illustrated in the visual display above them. For example, if you choose Tweed 4X10 from the Cabinet pop-up menu, you see the Tweed cabinet with four 10" speakers on the right side of the display.

- Choose a preconfigured model, consisting of an amplifier, a cabinet, an EQ type, and a microphone type, from the Model pop-up menu.

Amp Designer models

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 it was also used by blues and rock guitarists. It is more open and transparent-sounding than the Small Tweed Combo, but it 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 Tweed combos are known for.

Tip: Tweed combos are responsive to 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 soloing.

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 a tight low-end and restrained distortion. They are useful 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-saturated chords or strident solos.
Silverface Combo	A 2 x 12" combo with a loud, clean tone. It has a percussive, articulate attack that is suitable 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 reasonable low end impact. It excels at clean tones with a minimal overdrive.
Small Brownface Combo	A 1 x 12" combo that is smooth and rich-sounding, but retains a 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: Although these amps tend toward a clean and tight sound, you can use a Pedalboard distortion stompbox to attain hard-edged crunch sounds with sharp treble and extended low-end definition. See Pedalboard distortion pedals on page 45.

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 4 x 12" cabinets. At medium gain settings, these amps are suitable for thick 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. Tonally, it has a deeper and brighter sound at the low and high end, with a more "scooped" midrange 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.

Tip: The classic British head and 4 x 12" cabinet combo is ideal for riffs at high gain levels. These heads can also sound good through small cabinets, or at clean, low-gain settings.

British Combos

The British Combos capture the brash, treble-rich sound 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 mellow distortion and smooth compression.

Model	Description
British Blues Combo	This 2 x 12" combo has a loud, aggressive tone that is cleaner than the British heads, yet delivers rich distorted tones at high gain settings.
British Combo	A 2 x 12" combo based on early 1960s amps. Perfect for chiming chords and crisp solos.
Small British Combo	A 1 x 12" combo with half the power of the British Combo, this amp offers a 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: You can often use higher Treble and Presence knob settings with the British Combos than with other amp types. 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 a heavy rock tone. See Pedalboard distortion pedals on page 45.

British Alternatives

The late 1960s amplifier heads and combos that inspired the Sunshine models are loud and aggressive, with full mid frequencies. These amps are useful for single note solos, power chords, and big, open chords—making them popular with the "Brit-pop" bands of the 1990s. The Stadium amps are famed for their ability to play at extremely high levels without dissolving into an indistinct 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 is a good choice for powerful pop-rock chords. If the tone is too dark, use a high Treble knob setting to open up the sound.
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 and has tonal qualities similar to the 1960s British Combo. This amp also sounds good with a 4 x 12" cabinet.
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, but it retains 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 smoother than the Stadium Stack.

Tip: The Stadium amps can be slow to distort, so most famous users have paired them with aggressive fuzz pedals. Try combining them with Pedalboard's Candy Fuzz or Fuzz Machine stompboxes. See Pedalboard distortion pedals on page 45.

Metal Stacks

The Metal Stack models are inspired by the powerful, high gain amplifier heads favored by modern hard rock and metal musicians. All are paired with 4 x 12" cabinets. Their signature tones range from heavy distortion to extremely heavy distortion. These models are ideal if you want powerful lows, harsh highs, and long sustain in your guitar tones.

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

Tip: Combining the Turbo Stack with distortion and fuzz pedals can diminish the amp's edgy tone. 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 you can use 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. These models use multiple gain stages to generate smooth, sustain-heavy distortion *and* bold, bright, clean sounds. Can deliver a heavier sound when paired with a 4 x 12" cabinet.
Boutique Retro Combo	A 2 x 12" combo inspired by expensive modern amps that combine the sounds of several 1960s combos. It excels at clean 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. This model has very sensitive tone controls that can deliver countless guitar tones.
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, despite the small speaker.
Transparent Preamp	A preamp stage with no coloration. Note that Transparent Preamp is activated in the Amp pop-up menu, not in the Model pop-up menu.

Tip: Combine the Pawnshop Combo amp with Pedalboard's Hi Drive or Candy Fuzz stompboxes to emulate hard rock tones of the late 1960s. See Pedalboard distortion pedals on page 45.

Amp Designer cabinets

This table outlines 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 from the late 1950s that was originally conceived for bassists but that guitarists use for its sparkling presence.
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 low-end punch.
Blackface 1 x 10	An open-back 1960s cabinet with glassy highs and low/mid body.
Brownface 1 x 12	A balanced 1960s open-back cabinet that is smooth, transparent, and rich-sounding.
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, due to the complex phase cancelations 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 midrange emphasis.
Brown 4 x 12	A closed-back 4 x 12" cabinet with a good low end and complex midrange.
British Blues 2 x 12	A bright-sounding open-back cabinet with solid lows and crisp highs, even at high gain settings.
Modern American 4 x 12	A closed-back 4 x 12" cabinet with a full sound. The lows and mids are denser than the British 4 x 12" cabinets.
Studio 1 x 12	A compact-sounding open-back cabinet with full mids and glassy 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 low/mid transparency.
Boutique British 2 x 12	A 2 x 12" cabinet based on the British 2 x 12. It has a richer midrange and is more powerful in the treble range.
Sunshine 4 x 12	A 4 x 12" closed-back cabinet with a thick, rich midrange.
Sunshine 1 x 12	A single 12" open-back combo amp cabinet with a lively sound that has bright, sweet highs, and transparent mids.

Cabinet	Description
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 warmth 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 midrange and is more powerful 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	A single 8" speaker cabinet that has a strong low-end punch.
Direct	This option bypasses the speaker emulation section.

Tip: A creative sound design option is to choose Direct from the Cabinet pop-up menu, insert Space Designer in the next free Insert slot, then load one of Space Designer's "warped" speaker impulse responses.

Build a custom 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. You create your own by using the Amp, Cabinet, and Mic pop-up menus, located in the black bar at the bottom of the interface, as well as the EQ pop-up menu, which you open by clicking the word *EQ* or *Custom EQ* above the knobs in the left part of the knobs section.

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

EQ pop-up menu

Model pop-up menu
Amp pop-up menu
Cabinet pop-up menu
Mic pop-up menu

Choose an Amp Designer amplifier

■ Choose an amplifier from the Amp pop-up menu in 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 on page 13
- Classic American Combos on page 14
- British Stacks on page 14
- British Combos on page 15
- British Alternatives on page 15
- Metal Stacks on page 16
- Additional Combos on page 16

Choose an Amp Designer cabinet

Cabinets have a huge impact on the character of a guitar sound (see Amp Designer cabinets on page 17).

Whereas certain amplifier and cabinet pairings have been popular for decades, departing from them can be an effective way to create fresh-sounding tones. For example, most players automatically associate British heads with 4 x 12" cabinets. Amp Designer lets you drive a small speaker with a powerful head, or pair a tiny amp with a 4 x 12" cabinet. You can experiment with random amplifier and cabinet combinations, but you can also make an educated guess about nontraditional combinations by considering the variables that determine a cabinet's "sound."

■ Choose a cabinet from the Cabinet pop-up menu in the black bar at the bottom of the Amp Designer interface. Use the following considerations to guide your decision:

- *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 spacious sound. 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 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 in history uses 8" speakers. You can often 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 typically choose cabinets with multiple speakers for their big sound. The number of speakers is less important than it may appear. Phase cancelations occur between the speakers, adding texture and interest to the tone. Much of the "classic rock" sound, for example, is due to tonal peaks and dips caused by interactions between the speakers in a 4 x 12" cabinet.

Choose a microphone type and placement

1 Choose a microphone model from the Mic pop-up menu.

- *Condenser models:* Emulate the sound of high-end studio condenser microphones. The sound of condenser microphones is fine, transparent, and well-balanced. Choose Condenser 87 or Condenser 414.
- *Dynamic models:* Emulate the sound of popular dynamic cardioid microphones. Dynamic microphones sound brighter and more cutting than Condenser models. The mid-range is boosted, with softer lower-mid frequencies, making dynamic microphones a good choice for rock guitar tones, especially if you want guitars to cut through other tracks in a mix. Choose Dynamic 20, Dynamic 57, Dynamic 421, or Dynamic 609.
- *Ribbon 121:* 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.

2 Drag the white dot in the graphic above the Mic pop-up menu to set the microphone position and distance relative to the cabinet.

Choose and adjust an EQ type

1 Click the word *EQ* or *CUSTOM EQ* above the Bass, Mids, and Treble knobs to open the EQ pop-up menu, then choose an EQ model. See Amp Designer EQ types on page 22.

2 Rotate the Bass, Mids, and Treble knobs to adjust the EQ model you choose.

Amp Designer equalizer

Amp Designer equalizer overview

Hardware amplifier tone controls vary among models and manufacturers. For example, the treble knobs on two different models may target different frequencies or provide different levels of cut or boost. Some equalizer (EQ) sections amplify the guitar signal more than others, thus affecting the way the amp distorts.

Amp Designer provides multiple EQ types to mirror these variations in hardware amplifiers. All EQ types have identical controls—Bass, Mids, and Treble—but these controls can behave very differently depending on which EQ type you choose.

Selecting an EQ type other than the one traditionally associated with an amplifier usually results in significant tonal changes. As with hardware amplifiers, Amp Designer's EQs are calibrated to perform well with particular amplifier models. Choosing other EQ types can sometimes produce a thin or unpleasantly distorted tone.

Despite these less pleasant-sounding possibilities, it is worth experimenting with various amplifier and EQ combinations, because many will sound good together.
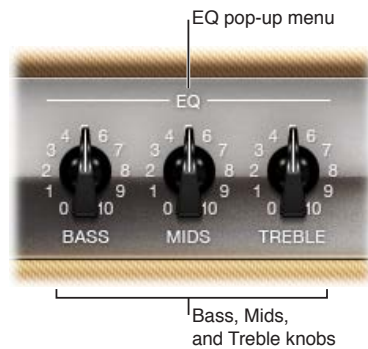
EQ parameters

- *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 EQ types on page 22.

- *Bass, Mids, and Treble knobs:* Rotate to adjust the frequency ranges of the EQ models, similar to the way you would adjust the tone knobs on a hardware guitar amplifier. The behavior and response of these knobs changes when different EQ models are chosen.

Amp Designer EQ types

This table describes the properties of each Amp Designer EQ type.

EQ type	Description
British Bright	Inspired by the EQ of British combo amps of the 1960s, it is loud and aggressive, with stronger 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 similar circuit. It is loud and subject to distortion. This EQ is useful if you want a rougher sound.
U.S. Classic	Derived from the EQ circuit of the American Blackface amps, it has a tone 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.
Modern	Based on a digital EQ unit popular in the 1980s and 1990s, this EQ is useful for sculpting the aggressive highs, deep lows, and scooped mids associated with that era's rock and metal music styles.
Boutique	Replicating the tone section of a "retro modern" boutique amp, it excels at precise EQ adjustments, though its tone may be too clean when used with vintage amplifiers. This EQ is a good choice if you want a cleaner, brighter sound.

Amp Designer amplifier controls

The amp parameters include controls for the input gain, presence, and master output. The Gain knob is located to the left in the knobs section, the Presence and Master knobs are to the right, and the Output parameter is at the lower-right edge of the interface.

Amplifier parameters

- *Gain knob:* Rotate to set the amount of pre-amplification applied to the input signal. This control affects specific amp models in different ways. For example, when you use the British Amp, the maximum gain setting produces a powerful crunch sound. When you use the Vintage British Head or Modern British Head, the same gain setting produces heavy distortion, suitable for lead solos.

- *Presence knob:* Rotate to adjust the ultra-high frequency range—above the range of the Treble control. The Presence parameter affects only the output (Master) stage.

- *Master knob:* Rotate to set the output volume of the amplifier signal sent to the cabinet. For tube amplifiers, increasing the Master level typically produces a compressed and saturated sound, resulting in a more distorted and louder signal.

WARNING: Because high Master knob settings can produce an extremely loud output that can damage your speakers or hearing, start with a low Master knob setting and then slowly increase it.

- *Output slider or field:* Drag to set the final output level of Amp Designer.

  Note: The slider is replaced with a field in the small interface.

Amp Designer effects

Amp Designer effects overview

The effects parameters include reverb, tremolo, and vibrato, which emulate the processors found on many amplifiers. These controls are found in the center of the knobs section.

Reverb, which is controlled by an On/Off switch in the middle, can be added to either tremolo or vibrato, or it can be used independently. See Amp Designer reverb effect on page 24.

You can select either Trem(olo), which modulates the amplitude or volume of the sound, or Vib(rato), which modulates the pitch. See Amp Designer tremolo and vibrato on page 25.

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

Amp Designer reverb effect

Reverb is always available in Amp Designer, even when you are 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. The Reverb pop-up menu is located above these controls. You can add Reverb to either the tremolo or vibrato effect, or you can use it independently.

Reverb parameters

- *On/Off switch:* Turns the reverb effect on or off.
- *Reverb pop-up menu:* Click the word *Reverb* to open the pop-up menu, which includes the following reverb types: 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 types on page 25 for information on these reverb types.
- *Level knob:* Rotate to set the amount of reverb applied to the pre-amplified signal.

Amp Designer reverb types

This table indicates the properties of each Amp Designer reverb type.

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, 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 midrange 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 rich and balanced modern reverb.
Warm Reverb	A lush modern reverb with rich lows/mids and understated highs.

Amp Designer tremolo and vibrato

Tremolo and vibrato are controlled by several switches and two knobs in the effects section. Tremolo modulates the amplitude or volume of the sound, and Vibrato modulates the pitch of the sound.

Tremolo and vibrato parameters

- *On/Off switch:* Click to turn the tremolo or vibrato effect on or off.
- *Trem(olo)/Vib(rato) switch:* Click to choose either tremolo or vibrato.
- *Depth knob:* Rotate to set the intensity of the modulation for either tremolo or vibrato.
- *Speed knob:* Rotate to set the speed of the modulation in hertz. Lower settings produce a smooth, floating sound. Higher settings produce a rotor-like effect.
- *Sync/Free switch:* Select Sync to synchronize the modulation speed with the host application tempo. If you select Free, you can use the Speed knob to set the modulation speed to different bar, beat, and musical note values (1/8, 1/16, and so on, including triplet and dotted-note values).

Amp Designer microphone parameters

Amp Designer provides seven virtual microphone types. As with other components in the tone chain, different selections can yield very different results. After choosing a cabinet, you can set the type of microphone to emulate and can place the microphone, relative to the cabinet.

The Mic pop-up menu is near the right end of the black bar. The speaker-adjustment graphic appears when you move your pointer in 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 to switch to the full interface.

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

Microphone parameters

- *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 is 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:* Choose a microphone model:

  - *Condenser models:* Emulates the sound of high-end studio condenser microphones. The sound of condenser microphones is fine, transparent, and well-balanced. Choose from: Condenser 87 and Condenser 414.

  - *Dynamic models:* Emulates the sound of popular dynamic cardioid microphones. Dynamic microphones sound brighter and more cutting than Condenser models. The mid-range is boosted, with softer lower-mid frequencies, making dynamic microphones a good choice for rock guitar tones; useful if you want guitars to cut through other tracks in a mix. Choose from: Dynamic 20, Dynamic 57, Dynamic 421, and Dynamic 609.

  - *Ribbon 121:* 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, then set track signal levels.

Bass Amp Designer

Bass Amp Designer overview

Bass Amp Designer emulates the sound of three famous bass guitar amplifiers and the speaker cabinets used with them. Each preconfigured model combines an amp and cabinet that recreates a well-known bass guitar amplifier sound. The amp and cabinet can be combined with integrated compression and EQ units to alter the tone. You can process signals directly, reproducing the sound of your bass played through these amplification systems. Virtual microphones are used to pick up the signal of the emulated amplifier and cabinet. You can choose from, and position, three different microphone types.

When recording, many bass players use a direct connection to a mixing board or other recording equipment, often using a passive (non powered) or active (powered) D.I. box (Direct Injection box). The use of a pre-amp with passive or active EQ and a hardware compressor instead of, or in addition to, a D.I. box is extremely popular too. Bass Amp Designer emulates a professional-level American D.I. box.

Bass Amp Designer has a two channel design—one for the pre-amp and one for the D.I. box. This enables you to flexibly change the signal flow for the following playing and recording configurations: pre-amp with passive or active EQ, compressor, a straight power amp, just the sound of the cabinets and microphones, D.I. box alone, bass amp alone, or both in parallel. See Amplifier signal flow and Pre-amp signal flow.
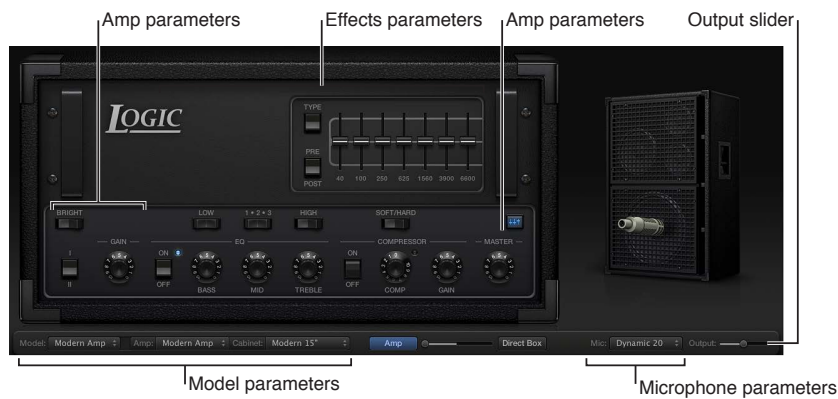
The Bass Amp Designer interface is divided into four main parameter sections.

- *Model parameters:* The Model pop-up menu at the left of the black bar at the bottom is used to choose a preconfigured model, consisting of an amplifier, a cabinet, and a microphone type. The other menus in the black bar enable you to independently choose the type of amplifier, cabinet, and microphone. See Build a custom Bass Amp Designer combo on page 29.

- *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 Bass Amp Designer amplifier controls on page 33.

- *Effects parameters:* Located in the center of the knobs section, these parameters control the integrated EQ and compressor effects. A further graphic or parametric EQ is shown above the compressor controls when the EQ button is turned on. See Bass Amp Designer effects overview on page 34.

- *Microphone parameters:* Located at the right of the interface, these parameters set the type and position of the microphone that captures the amplifier and cabinet sound. See Bass Amp Designer microphone parameters on page 37.

- *Output slider:* The Output slider is found at the lower-right corner of the interface. It serves as the final level control for Bass Amp Designer's 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.

Choose a Bass Amp Designer model

- Choose a preconfigured model, consisting of an amplifier, a cabinet, and a microphone type, from the Model pop-up menu.

You can use the Model pop-up menu to choose a preconfigured model, or you can build a customized model using the Amp, Cabinet, and Mic pop-up menus. See Build a custom Bass Amp Designer combo. Your choices remain visible in the pop-up menus, and they are also illustrated in the visual display above them.

Bass Amp Designer models

Bass amplifier models

Bass Amp Designer emulates the three most iconic tube bass amps and cabinets from the 1960s, 1970s, and 1980s. The table includes the cabinets that each amplifier is normally matched with.

Amp model	Cabinet	Description
Classic Amp	8 x 10 inch speakers	Emulates a classic six-tube bass amp with a tuned, closed-back cabinet introduced in 1960. This model is good for a range of musical styles.
Flip Top Amp	1 x 15 inch speaker	Emulates a 300-watt tube head introduced in 1969. It is ideal for full, fundamental tones.
Modern Amp	3-way speaker array	Emulates a 12-tube 360-watt head introduced in 1989. It is suitable for many musical styles and is the ideal choice for highly articulated performances.

Bass cabinet models

The table outlines the properties of each cabinet model available in Bass Amp Designer.

Cabinet	Description
Modern Cabinet 15"	1 x 15 inch speaker, closed-back design. Very deep and full tone.
Modern Cabinet 10"	1 x 10 inch speaker, closed-back design. A punchy tone.
Modern Cabinet 6"	1 x 6 inch speaker, closed-back design.
Classic Cabinet 8 X 10"	8 x 10 inch speakers, closed-back design.
Flip Top Cabinet 1 X 15"	1 x 15 inch speaker, closed-back design.
Modern 3 Way	1 x 15 inch speaker, 1 x 10 inch speaker, and 1 x 6 inch speaker. You can move the microphone vertically and can position it 20, 30, or 40 cm away from the cabinet.
Direct (PowerAmp Out)	A direct signal from the power stage of the emulated amplifier. The cabinet and microphone are removed from the signal path.
Direct (PreAmp Out)	A direct signal from the pre-amplifier stage of the emulated amplifier. The cabinet, microphone, and power amp are removed from the signal path.

Build a custom Bass 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.

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

Choose a Bass Amp Designer amplifier
- Choose an amplifier from the Amp pop-up menu in the black bar at the bottom of the Amp Designer interface. See Bass amplifier models for details on the characteristics of each amplifier.

Choose a Bass Amp Designer cabinet
Cabinets have a huge impact on the character of a bass guitar sound (see Bass cabinet models on page 29).

Whereas certain amplifier and cabinet pairings have been popular for decades, departing from them is an effective way to create fresh-sounding tones. You can try 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.

- Choose a cabinet from the Cabinet pop-up menu in the black bar at the bottom of the Bass Amp Designer interface.
  - *Old or new speakers:* Some Bass Amp Designer models 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:* Try several sizes and choose the one that works best for your music.

- *Single speakers or multiple speakers:* The number of speakers is less important than it may appear. Phase cancelations occur between the speakers, adding texture and interest to the tone.

Choose a microphone type and placement

1 Click the Mic pop-up menu to choose a microphone model.

- *Condenser 87:* Emulates the sound of a high-end German studio condenser microphone. The sound of condenser microphones is fine, transparent, and well-balanced.

- *Dynamic 20:* Emulates the sound of popular American dynamic cardioid microphones. This microphone type sounds brighter and more cutting than the Condenser 87 model. The lower-mid frequencies are rolled off, making it a good choice for miking rock tones. It is especially useful if you want your bass guitar part to cut through other tracks in a mix.

- *Dynamic 421:* Emulates the sound of a German dynamic cardioid microphone. It can capture a wide frequency range and has a slight emphasis of the treble range. It is useful for clean tones.

2 Drag the white dot in the graphic above the Mic pop-up menu to set the microphone position and distance relative to the cabinet.

Bass Amp Designer signal flow

Amplifier signal flow

Bass Amp Designer has a two-channel design—one for the pre-amp and one for the D.I. box. You can use these independently or can blend them by using the controls on the black bar at the bottom.

Important: The two channels are always used in parallel if the Blend slider is not set to the far right or to the far left position.

The channel signal flow changes when you choose different models from the Cabinet pop-up menu.

Cabinet	Blend slider position	Channel 1 routing	Channel 2 routing
Any speaker cabinet model	Middle	Pre-amp, power amp, cabinet, mic	D.I. box
Direct (PowerAmp Out)	Middle	Pre-amp, power amp	D.I. box
Direct (PreAmp Out)	Middle	Pre-amp	D.I. box
Any speaker cabinet model	Far left	Pre-amp, power amp, cabinet, mic	
Direct (PowerAmp Out)	Far left	Pre-amp, power amp	
Direct (PreAmp Out)	Far left	Pre-amp	
Direct (PreAmp Out)	Far right		D.I. box

Pre-amp signal flow

The pre-amp section is very flexible, and can be used in several ways when you use different combinations of On/Off and Pre/Post switches. The signal flow indicated in the Mode column is in series when multiple processors are used—that is, the output of one processor signal is fed into the next processor.

Mode	EQ On/Off	Compressor On/Off	Additional EQ On/Off	Pre/Post switch
All off	Off	Off	Off	
EQ only	On	Off	Off	
Compressor only	Off	On	Off	
Additional EQ only	Off	Off	On	
EQ into Compressor only	On	On	Off	
EQ into Additional EQ only	On	Off	On	
Additional EQ into Compressor only	Off	On	On	Pre
Compressor into Additional EQ only	Off	On	On	Post
All on (EQ into Additional EQ into Compressor)	On	On	On	Pre
All on (EQ into Compressor into Additional EQ)	On	On	On	Post

Use the D.I. box

The D.I. box is modeled on a highly regarded American D.I. unit.

D.I. box parameters

- *Boost knob:* Rotate to set the input gain of the D.I. box.
- *HF Cut button:* Click to turn on a highpass filter. This is used to reduce noise.
- *Tone knob:* Rotate to set the tonal color of the D.I. box. Choose from the following preset EQ curves:
  - *1:* An EQ curve with a -6 dB scoop from 100 Hz to 10 kHz, most pronounced around 800 Hz. Suitable for acoustic and string instruments, electric bass, and keyboards.
  - *2:* An EQ curve with a very pronounced -24 dB v-shaped notch from 100 Hz to 10 kHz, centered around 800 Hz. Suitable for electric bass guitar.
  - *3:* An EQ curve with a -3 dB scoop from 100 Hz to 10 kHz, most pronounced around 800 to 1200 Hz. Suitable for acoustics, strings, electric and bass guitar, and keyboards.
  - *4:* An EQ curve with a -3 dB scoop from 1 kHz to 10 kHz, most pronounced around 8 kHz. Frequencies between 60 Hz and 1 kHz have a slight boost of 1 or 2 dB above unity. Frequencies above 10 kHz have a +3 dB boost. Suitable for acoustics, strings, electric and bass guitars, and keyboards.
  - *5:* A sloped EQ curve that ramps up from -24 dB at 10 Hz to + 3 dB at around 900 Hz. Suitable for acoustic and electric guitar.
  - *6:* A sloped EQ curve that ramps up from -24 dB at 10 Hz to +3 dB at around 900 Hz. The signal rolls off by approximately 6 dB between 10 and 20 kHz. Suitable for electric and bass guitar.
- *Tone on/off button:* Click to turn on the tone control.

Use the D.I. box only
- Drag the Blend slider located in the black bar to the far right.

Use the D.I. box and the amplifier in parallel
- Drag the Blend slider located in the black bar to any central position—not to the far right or the far left.

Bass Amp Designer amplifier controls

The amp parameters include controls for channel selection, input filter and gain, and master output. The Gain knob is located to the left in the knobs section and the Master knob and Output slider are located at the far right.

Bright switch   Master knob

Gain knob   Output slider

Channel I/II switch

Amplifier parameters

- *Channel I/II switch:* Click to switch between channel I and channel II.
  - Channel I is active, with a gain of 0 dB.
  - Channel II is passive, with a gain of -15 dB.
- *Bright switch:* Click to switch between normal and bright modes. In the bright position, highs and upper mids are added to the tone.

  Note: The increased mid and high range may lead to a perceived low end roll-off. Use the Bass EQ knob if you feel the bottom end needs a boost.
- *Gain knob:* Rotate to set the amount of pre-amplification applied to the input signal. The Gain knob affects amp models differently.
- *Master knob:* Rotate to set the output volume of the amplifier signal sent to the cabinet. Increasing the Master level typically produces a compressed and saturated sound, resulting in a more distorted and louder signal.

  Note: If you choose Direct PowerAmp from the Cabinet pop-up menu, the output signal is routed directly to the Amp/Direct Box Blend fader. However, if you choose Direct PreAmp from the Cabinet pop-up menu, the Master knob acts as pre-amp master gain control before the output signal is routed to the Amp/Direct Box Blend fader.
- *Output slider:* Drag to set the final output level of Bass Amp Designer.

Bass Amp Designer effects

Bass Amp Designer effects overview

Bass Amp Designer provides multiple EQ types to sculpt your instrument tones.

It provides a basic EQ that mirrors the tonal qualities of the integrated EQ of the amplifier model you choose, if applicable. All amplifier model EQs have identical controls: Bass, Mids, and Treble. See Bass Amp Designer EQ.

Bass Amp Designer also offers an additional Graphic or Parametric EQ that you turn on with the EQ switch above the Master knob at the far right. See Bass Amp Designer Graphic EQ and Bass Amp Designer Parametric EQ.

Bass Amp Designer also integrates a dedicated, custom-built compression circuit that is optimized for electric bass. See Bass Amp Designer compressor.

Bass Amp Designer EQ

The EQ section contains a larger and more inclusive set of the EQ units found in the three original bass amps emulated by Bass Amp Designer.

EQ parameters

- *EQ on/off switch:* Click to turn the EQ on or off.

- *Bass, Mids, and Treble knobs:* Rotate to adjust the frequency ranges of the EQ, similar to the tone knobs on a hardware amplifier.

- *Low switch:* Click to switch between two positions that affect the tone and behavior of the Bass EQ knob.

- *1-2-3 switch:* Click to switch between three positions that affect the tone and behavior of the Mids EQ knob.

- *High switch:* Click to switch between two positions that affect the tone and behavior of the Treble EQ knob.

Bass Amp Designer compressor

The internal compression circuit is custom-built for use with Bass Amp Designer. It features an AutoGain function that compensates for volume reductions caused by compression.

Compressor parameters

- *Compressor on/off switch:* Click to turn the Compressor on or off.
- *Fast/Easy switch:* Click to switch between two compression algorithms:
    - *Fast:* Stronger compression, with good control over levels, which makes it easier to fit the bass into an arrangement.
    - *Easy:* Compression with a slow attack and longer sustain phase.
- *Comp(ression) knob:* Rotate to set the amount of compression intensity applied to the input signal.
- *Gain knob:* Rotate to add gain to, or subtract gain from, the gain staging of the internal AutoGain feature.

    Note: AutoGain is always active.

Bass Amp Designer Graphic EQ

Bass Amp Designer offers an additional Graphic or Parametric EQ that you turn on with the EQ switch above the Master knob at the far right.

Note: The Graphic EQ in a pre-compressor signal flow is enabled by default.
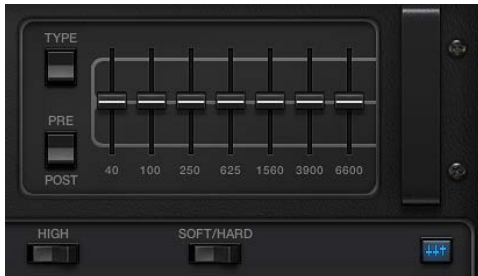
Graphic EQ parameters

- *Type switch:* Click the up position to choose the Graphic EQ. Click the down position to choose the Parametric EQ.

    Graphic and Parametric EQ parameter settings are retained when switching between EQ types and when the additional EQ is turned off. This enables you to quickly make AB comparisons.

- *Pre/Post switch:* Click to determine if the additional EQ is inserted before or after—pre or post—the compressor section within the signal flow.

    Note: This parameter is relevant only if the Compressor is turned on.

- *Frequency sliders:* Drag to set the amount of boost or cut for each frequency band.

Bass Amp Designer Parametric EQ

Bass Amp Designer offers an additional Graphic or Parametric EQ that you turn on with the EQ switch above the Master knob at the far right. The Parametric EQ provides two EQ bands:

- *HiMid:* Controls frequencies in the high and high-mid range.
- *LoMid:* Controls frequencies in the low and low-mid range.

Parametric EQ parameters

- *Type switch:* Click the up position to choose the Graphic EQ. Click the down position to choose the Parametric EQ.

  Graphic and Parametric EQ parameter settings are retained when switching between EQ types and when the additional EQ is turned off. This enables you to quickly make AB comparisons.

- *Pre/Post switch:* Click to determine if the additional EQ is inserted before or after—pre or post—the compressor section within the signal flow.

  Note: This parameter is relevant only if the Compressor is turned on.

- *Gain knobs:* Rotate to adjust the amount of cut or boost applied to the frequency range set with the kHz knob.
- *kHz knobs:* Rotate to set the frequency range that you want to cut or boost with the Gain knob.
- *Q knobs:* Rotate to set the width of the band surrounding the frequency set with the kHz knobs.

  The lower the Q knob value, the wider the band, which means that more frequencies will be affected. The higher the Q knob value, the narrower the band, which means that only the frequencies nearest to the frequency set with the kHz knob will be affected.

Bass Amp Designer microphone parameters

Bass Amp Designer offers three virtual microphone types. As with other components in the tone chain, different selections can yield different results. After choosing a cabinet, you can choose the type of microphone to emulate and you can adjust the position of the microphone, relative to the cabinet.

The Mic pop-up menu is near the right end of the black bar. The speaker-adjustment graphic appears when you move your mouse in the area above the Mic pop-up menu.
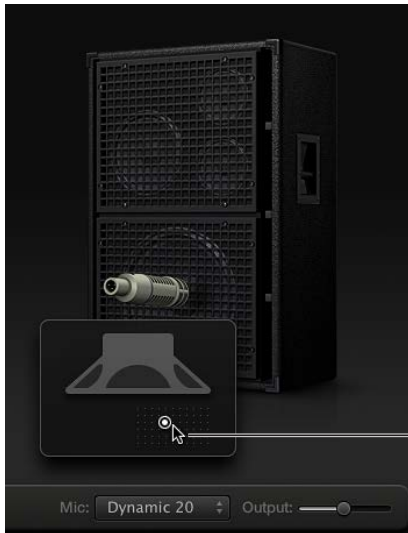
Move your mouse above the Mic pop-up menu to display the speaker-adjustment graphic.

Microphone parameters

- *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. If you place the microphone on the rim of the speaker (off-axis), you obtain a brighter, thinner tone. Moving the microphone closer to the speaker emphasizes bass response.

  The microphone position is shown on the cabinet and is 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:* Choose a microphone model:
  - *Condenser 87:* Emulates the sound of a high-end German studio condenser microphone. The sound of condenser microphones is fine, transparent, and well-balanced.
  - *Dynamic 20:* Emulates the sound of popular American dynamic cardioid microphones. This microphone type sounds brighter and more cutting than the Condenser 87 model. The lower-mid frequencies are rolled off, making it a good choice for miking rock tones. It is especially useful if you want your bass guitar part to cut through other tracks in a mix.
  - *Dynamic 421:* Emulates the sound of a German dynamic cardioid microphone. It can capture a wide frequency range and has a slight emphasis of the treble range. It is useful for clean tones.

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

Pedalboard

Pedalboard overview

Pedalboard simulates the sound of a number of 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.

- *Pedal Browser:* Shows all pedal effects and utilities. These can be dragged into the Pedal area as part of the signal chain. See Use the Pedal Browser on page 39. This interface area is also used for the alternative import mode. See Use Pedalboard's import mode on page 40.
- *Pedal area:* This is where you determine the order of effects and set effect parameters. You can add, replace, and remove stompboxes here. See Use the Pedal area on page 41.
- *Router:* Used to control signal flow in the two effects busses (Bus A and Bus B) available in Pedalboard. See Use Pedalboard's Router on page 42.
- *Macro Controls:* Used to assign eight MIDI controllers, which can be used to control any stompbox parameter in real time. See Use Pedalboard's Macro Controls on page 44.

Use the 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. The effect and utility pedals are described in the following sections:

- Pedalboard distortion pedals on page 45
- Pedalboard modulation pedals on page 47
- Pedalboard delay pedals on page 50
- Pedalboard filter pedals on page 51
- Pedalboard dynamics pedals on page 51
- Pedalboard utility pedals on page 52

View pop-up menu   Import Mode button

Hide or show the Pedal Browser

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

Show specific pedal groups in the Pedal Browser

■ Choose Distortion, Modulation, Delay, Filter, Dynamics, or Utility from the View pop-up menu.

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.

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 replaces the selected pedal.

Use 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 pop-up 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.

Turn import mode on or off

- 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.

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.

Depending on the setting you chose, one or more stompboxes appear in the Pedal Browser. The name of the imported setting is shown at the bottom of the Pedal Browser.

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.

- Make sure 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.

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 is 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.

Use the Pedal area

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

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.

- Make sure 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 Use Pedalboard's Router on page 42.

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 Use Pedalboard's Router on page 42.

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 replace "effect" pedals, but not the Mixer or Splitter utilities. Bus routings, if active, are not changed when an effect pedal is replaced. See Use Pedalboard's Router on page 42.

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.

Use Pedalboard's Router

Pedalboard has two discrete signal busses—Bus A and Bus B—that appear as two horizontal gray lines in the Router 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—the lower of the two lines—by default.

Note: The Router 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 Router remains open even when your pointer is not over it. You can close the Router by clicking the small latch button at the top; the Router then opens or closes automatically when you move your pointer over it.

Notes on Splitter utility 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 point—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 combination, or directly to the right of an inserted Mixer utility.

Create a second bus routing
Do one of the following:

- Move your pointer immediately above the Pedal area to open the Router, and click the name of a stompbox in the Router.

  Two gray lines appear in the Router—the lower one representing Bus A and the upper one Bus B—and the pedal name moves to the upper line. The chosen stompbox is now routed to Bus B, and 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.

Remove the second bus routing
Do one of the following:

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

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

Remove an effect from the second bus

- Click the name of the pedal in the Router. (You can also click the lower gray line immediately above the pedal to remove the pedal from the second bus.)

  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 enables parallel routing of wet and dry signals. Only when all pedal effects are removed from the Pedal area are the Mixer utility and second bus removed.

Determine the split point between busses
When more than one bus is active, a number of dots appear along the "cables"—the gray lines—in the Router. These represent the output (the *socket*) of the pedal to the lower left of the dot.

- Click the appropriate dot to determine the split point—the point where the signal is routed between busses.

  A cable appears between the busses when you click a dot.

  Note: You cannot create a split point directly before or after the Mixer utility.

Switch between a Splitter utility and a bus split point

- *To replace a bus split point with a Splitter utility:* Double-click the bus split point dot in the Router.

  The Splitter utility appears in the Pedal area.

- *To replace a Splitter utility with a bus split point:* Double-click the Splitter label in the Router.

  The Splitter utility is removed from the Pedal area.

Change a Mixer utility position in the Pedal area

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

  If you move the Mixer utility to the left, the "downmix" of Bus A and Bus B occurs at the earlier insertion point. Relevant effect pedals are moved to the right and are inserted into Bus A.

  If you move the Mixer utility to the right, the "downmix" of Bus A and Bus B occurs 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 following or preceding a corresponding split point or Splitter utility.

Change a Splitter utility position in the Pedal area

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

If you move the Splitter utility to the left, the split between Bus A and Bus B occurs at the earlier insertion point. Relevant effect pedals are moved to the right and are inserted into Bus A.

If you move the Splitter utility to the right, the split between Bus A and Bus B occurs 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.

Use Pedalboard's Macro Controls

Pedalboard provides eight Macro Targets—A through H—that 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 MainStage, 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 MainStage Help for details.

Click the disclosure 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 from the corresponding Macro Target pop-up menu.

Assign a Macro A–H Target
Do one of the following:

▪ Choose the parameter that you want to control from any of the Macro A–H Target pop-up menus.

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

▪ Choose the "Auto assign" item from 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.

Pedalboard distortion pedals

This table 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 the Bright and Fat switch positions, respectively.
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 reducing 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.
Grit	A hard and nasty filtered distortion effect that sounds great on keyboards and guitars.
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, slightly 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.

Stompbox	Description
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.
Tube Burner	A vacuum tube-based distortion that provides a wide palette of sounds, ranging from warm grain to crispy overdrive.
Vintage Drive	Overdrive effect that emulates the distortion produced by a field-effect transistor (FET), 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.

Pedalboard modulation pedals

This table describes the modulation effects pedals.

Stompbox	Description
Dr Octave	A classic octaver effect with two independent octave controls plus an integrated overdrive.
Flange Factory	A deluxe flanging effect that allows precise control of every aspect of your sound.
Heavenly Chorus	A rich, sweet-sounding chorus effect that thickens the sound. Rate sets the modulation speed and can either run freely or be synchronized with the host application tempo when you enable 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 either run freely or be synchronized with the host application tempo when you enable 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 effect signal that is routed back into the input. This can change the tonal color, make the sweeping effect more pronounced, or do both.
Phaze 2	A flexible dual-phaser effect. LFO 1 and LFO 2 Rate set the modulation speed and can run freely, or be synchronized with the host application tempo when you enable 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 effect signal that is routed back into the input. This can change the tonal color, make the phasing effect more pronounced, or 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 either run freely or be synchronized with the host application tempo when you enable 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.

Stompbox	Description
Robo Flanger	Flexible flanging effect. Rate sets the modulation speed and can either run freely or be synchronized with the host application tempo when you enable 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 effect signal that is routed back into the input. This can change the tonal color, make the flanging effect more pronounced, or 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.
Roswell Ringer	A ring modulation effect that can make incoming audio sound metallic, or unrecognizable, and 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—with 12 notes per octave—or exponential. FB (feedback) determines the amount of effect signal that is routed back into the input. This can change the tonal color, make the effect more pronounced, or do both. Balance between the original and effect signals is set with the Mix knob. See Ringshifter overview on page 194 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 either run freely or be synchronized with the host application tempo when you enable 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—this applies only when the 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; and Brake stops the speaker rotation. See Rotor Cabinet effect overview on page 200 for background information on the Leslie effect.

Stompbox	Description
Total Tremolo	A flexible tremolo effect—modulation of the signal level. Rate sets the modulation speed and can either run freely or be synchronized with the host application tempo when you enable 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 change the LFO waveform shape. This enables you to create floating changes in level, or abrupt steps. Volume determines the output level of the effect. The 1/2 Speed and 2x 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 either run freely or be synchronized with the host application tempo when you enable 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.
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 either run freely or be synchronized with the host application tempo when you enable 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 on page 204 for background information on this effect.
Wham	A pedal-controlled pitch shifter. Mix sets the level balance between the source and pitch-shifted signals.

Pedalboard delay pedals

This table describes the Delay effects pedals.

Stompbox	Description
Blue Echo	A delay effect. Time sets the modulation speed and can either run freely or be synchronized with the host application tempo when you enable 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. Through passes the source signal through to the next pedal, while delay repeats continue.
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 Boutique, Simple, Vintage, Bright, or Resonant. Mix sets the ratio between the source and effect signals.
Tie Dye Delay	A warm-sounding reverse delay effect that is perfect for fans of 1960s and 1970s psychedelic rock.
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 either run freely or be synchronized with the host application tempo when you enable the Sync button. When synchronized, you can specify bar, beat, and note values, including triplets and dotted notes. Feedback determines the amount of 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.

Pedalboard filter pedals

This table describes the filter effects pedals.

Stompbox	Description
Auto-Funk	An auto-wah (filter) effect. Sensitivity sets a threshold that 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 determines 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 1970s TV police show soundtracks. You control it by dragging the pedal.
Graphic EQ	A classic 7-band EQ 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.

Pedalboard dynamics pedals

This table 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.

Pedalboard utility pedals

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

Stompbox	Description
Mixer	Controls 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 Use Pedalboard's Router on page 42 for more information. 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.
Splitter	A utility that can be inserted anywhere in the signal chain. Splitter has two modes: *Freq:* 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. *Split:* The incoming signal is routed equally to both buses. The Frequency knob has no impact in this mode. See Use Pedalboard's Router on page 42 for more information.

Delay effects

2

Delay effects overview

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.

Delay Designer

Delay Designer overview

Delay Designer is a multitap delay. Unlike traditional delay units that offer only one or two delays (or taps) that may or may not be fed back into the circuit, Delay Designer provides up to 26 individual taps. These taps are all fed from the source signal and can be edited to create unique delay effects.

Delay Designer provides control over the level, pan position, and pitch of each tap. Each tap can also be lowpass or highpass filtered.

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

As the name implies, Delay Designer offers significant sound design potential. You can use it for everything from a basic echo effect to an audio pattern sequencer. You can create complex, evolving, moving rhythms by synchronizing the placement of taps. This leads to further musical possibilities when coupled with judicious use of transposition and filtering. Alternatively, you can set up numerous taps as repeats of other taps, much as you would use the feedback control of a simple delay effect, but with individual control over each repeat.
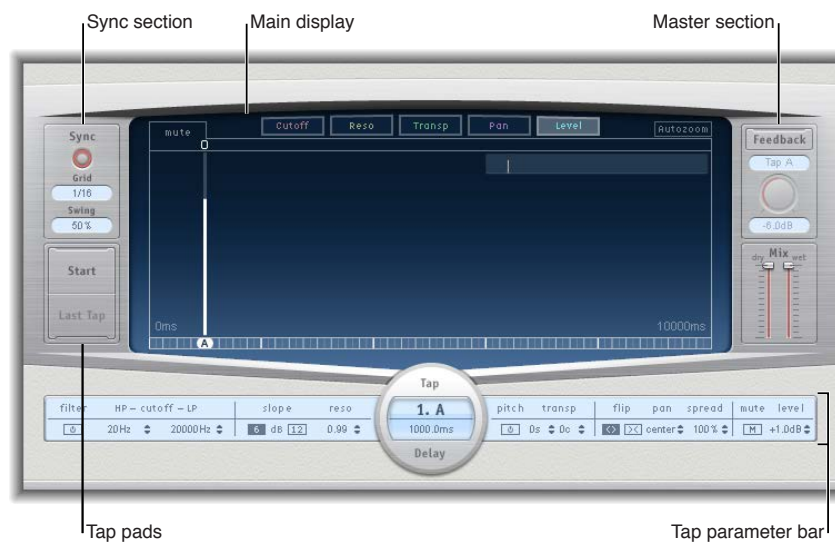
The Delay Designer interface consists of five main sections:

- *Main display:* Provides a visual representation of all taps. You can see and edit the parameters of each tap in this area. See Delay Designer main display on page 55.
- *Tap parameter bar:* Offers a numeric overview of the current parameter settings for the selected tap. You can view and edit the parameters of each tap in this area. See Delay Designer Tap parameter bar on page 67.
- *Tap pads:* You can use these two pads to create taps in Delay Designer. See Create taps in Delay Designer on page 58.
- *Sync section:* You can set all Delay Designer synchronization and quantization parameters in this area. See Delay Designer sync mode on page 68.
- *Master section:* This area contains the global Mix and Feedback parameters. See Delay Designer master parameters on page 69.

Delay Designer main display

Delay Designer's main display is used to view and edit tap parameters. You can choose the parameter to show and quickly zoom or navigate through all taps.
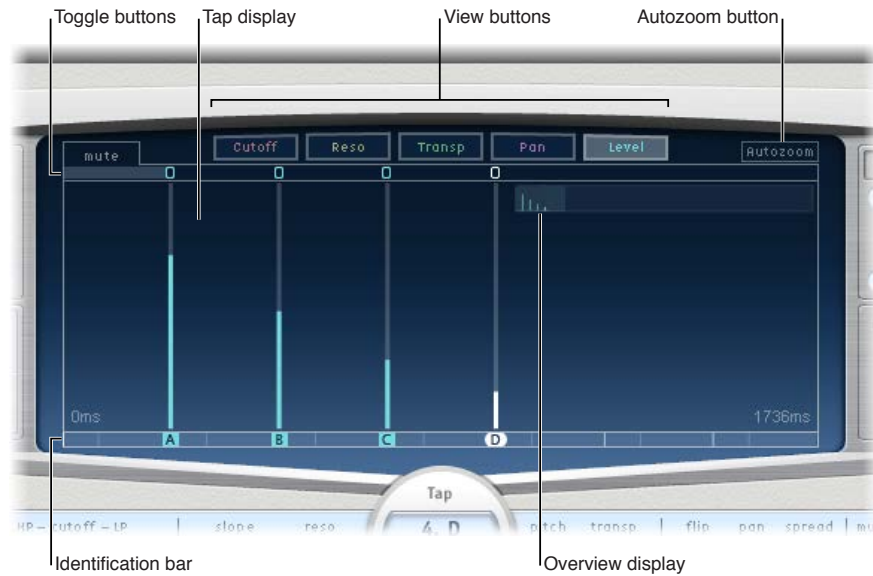
Main display parameters

- *View buttons:* Click to choose the parameter or parameters shown in the Tap display. See Use the Delay Designer Tap display.

- *Autozoom button:* Zooms the Tap display out, making all taps visible. Turn Autozoom off if you want to zoom the display in (by dragging vertically in the Overview display) to view specific taps.

- *Overview display:* Shows all taps in the time range.

- *Toggle buttons:* Click to turn the parameters of a particular tap on or off. The parameter being toggled is selected with the view buttons. The label at the left of the Toggle bar indicates the parameter. See Use Delay Designer's tap toggle buttons on page 57.

- *Tap display:* Represents each tap as a shaded line. Each tap contains a bright bar (or dot for stereo panning) that indicates the value of the parameter. You can directly edit tap parameters in the Tap display area. See Edit parameters in the Tap display on page 61.

- *Identification bar:* Shows an identification letter for each tap. This also serves as a time position indicator for each tap. You can move taps backward or forward in time along this bar/timeline. See Select, move, and delete taps on page 60.

Use the Delay Designer Tap display

The view buttons determine the parameter shown in Delay Designer's Tap display.

The Toggle bar is shown below the view buttons. You can use it to turn parameters on or off for each tap.

You can use Delay Designer's Overview display to zoom and to navigate the Tap display area.

Overview display

Tip: If the Overview display is hidden behind a tap, you can move it to the foreground by holding down Shift.

Select the parameter shown in the Tap display

- Click one (or more) of the following buttons to select the parameter or parameters shown in the Tap display.

- *Cutoff button:* Shows the highpass and lowpass filter cutoff frequencies of taps.
- *Reso(nance) button:* Shows the filter resonance value of each tap.
- *Transp(ose) button:* Shows the pitch transposition of each tap.
- *Pan button:* Shows the pan parameter of each tap.
  - For mono to stereo channels, each tap contains a line showing its pan position.
  - For stereo to stereo channels, each tap contains a dot showing its stereo balance. A line extending outward from the dot indicates the tap's stereo spread.
- *Level button:* Shows the relative volume level of each tap.

  Tip: Press Command-Option to temporarily switch the Tap display to Level view from another view.

Use Delay Designer's tap toggle buttons

Each tap has its own toggle button in the Toggle bar. These buttons provide a quick way to graphically turn parameters on and off. The parameter being toggled is determined by the current view button selection.

1 Click the view button for the parameter you want to toggle.

2 Click the toggle button of each tap that you want to change:

- *Cutoff view:* Turn the filter on or off.
- *Reso view:* Switch the filter slope between 6 dB and 12 dB.
- *Pitch view:* Switch pitch transposition on or off.
- *Pan view:* Switch between the Flip modes.
- *Level view:* Mute or unmute the tap.

Note: The first time you edit a filter or pitch transpose parameter, the filter or pitch transposition module automatically turns on. After you manually turn off either of these modules, however, you need to manually switch it back on.

Temporarily switch the mute state of taps

- In any view, Command-Option-click a toggle button.

When you release the Command and Option keys, the toggle buttons return to their standard functionality in the active view.

Zoom the Tap display
Do one of the following:

- Vertically drag the highlighted section (the bright rectangle) in the Overview display.

- Horizontally drag the highlighted bars—to the left or right of the bright rectangle—in the Overview display.

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

Move to different sections of the Tap display
- Horizontally drag the middle of the bright rectangle in the Overview display.

The zoomed view in the Tap display updates as you drag.

Create taps in Delay Designer

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

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

After a tap has been created, you can adjust its position, or you can remove it. See Select, move, and delete taps on page 60.

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

The Identification bar shows the letter of each visible tap. The Tap Delay field of the Tap parameter bar displays the letter of the currently selected tap or the letter of the tap being edited when multiple taps are selected (for details, see Select, move, and delete taps on page 60).

Create taps with the Tap pad

1 Click the upper pad (Start).

Note: Whenever you click the Start pad, it automatically erases all existing taps. Because of this behavior, after you create your initial taps, you will want to create subsequent taps by clicking in the Identification bar.

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

2 To begin recording new taps, click the Tap button.

3 To create new taps, click the Tap button.

These are created at the exact moments in time of each click, adopting the rhythm of your click pattern.

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

The final tap is added, ending tap recording, and assigning the last tap as the feedback tap (for more information about the feedback tap, see Delay Designer master parameters on page 69).

Note: If you do not click the Last Tap button, tap recording automatically stops after 10 seconds or when the 26th tap is created, whichever comes first.

Create a tap in the Identification bar

▪ Click the position where you want to add a tap.

Copy taps in the Identification bar

▪ Option-drag a selection of one or more taps to the position where you want to add the tap or taps.

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

Select, move, and delete taps

There is always at least one selected tap. You can easily distinguish selected taps by color—the Toggle bar icons and the Identification bar letters of selected taps are white.

You can move a tap backward or forward in time or completely remove it.

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

Select a tap
Do one of the following:

- Click a tap in the Tap display.

- Click the tap letter in the Identification bar.

- Click one of the arrows to the left of the Tap name to select the next or previous tap.

- Choose the tap letter from the pop-up menu to the right of the Tap name.
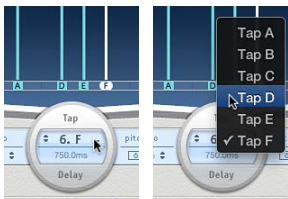
Select multiple taps
Do one of the following:

- *To select multiple taps:* Drag across the background of the Tap display.

- *To select multiple nonadjacent taps:* Shift-click specific taps in the Tap display.

Move a selected tap in time

- In the Identification bar, drag a tap to the left to go forward in time, or to the right to go backward in time.

This method also works when more than one tap is selected.

Note: Editing the Delay Time parameter in the Tap Delay field of the Tap parameter bar also moves a tap in time. For more details about the Tap Delay field and editing taps, see Delay Designer Tap parameter bar on page 67.

Delete a tap

Do one of the following:

- Select a tap, then press the Delete key.
- In the Identification bar, drag a tap letter downward, out of the Tap display.

This method also works when more than one tap is selected.

Delete all selected taps

- Control-click (or right-click) a tap, then choose "Delete tap(s)" from the shortcut menu.

Edit parameters in the Tap display

You can graphically edit any tap parameter that is represented as a vertical line in Delay Designer's Tap display. The Tap display is ideal if you want to edit the parameters of one tap relative to other taps or when you need to edit or align multiple taps simultaneously.

Edit a tap parameter in the Tap display

1 Click the view button of the parameter you want to edit.

2 Vertically drag the bright line of the tap you want to edit (or one of the selected taps, if multiple taps are selected).

If you selected multiple taps, the values of all selected taps are changed relative to each other.

Note: The method outlined above is slightly different for the Filter Cutoff and Pan parameters. See the tasks below.

Set the values of multiple taps

- Command-drag horizontally and vertically across several taps in the Tap display.

Parameter values change to match the pointer position as you drag across the taps. Command-dragging across several taps lets you draw value curves, much like using a pencil to create a curved line on a piece of paper.

Align the values of several taps

1 Command-click in the Tap display, and drag while holding down the Command key.

A line trails behind the pointer as you drag.
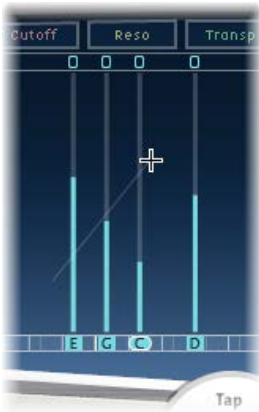
2 Click the appropriate position to mark the end point of the line.

The values of taps that fall between the start and end points are aligned along the line.
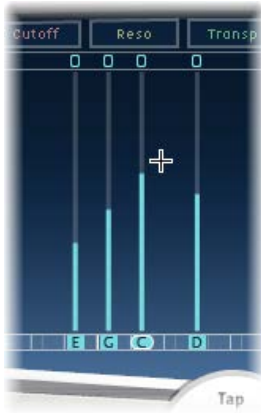
Reset the value of a tap
You can use Delay Designer's Tap display or Tap parameter bar to reset tap parameters to their default values.

- *To reset a parameter to its default setting in the Tap display:* Option-click a tap to reset the selected parameter to its default setting.

  If multiple taps are selected, Option-clicking any tap resets the chosen parameter to its default value for all selected taps.

- *To reset a parameter to its default setting in the Tap parameter bar:* Option-click a parameter value to reset it to the default setting.

  If multiple taps are selected, Option-clicking a parameter of any tap resets all selected taps to the default value for that parameter.

Edit filter cutoff in the Tap display
In Cutoff view, each tap actually shows two parameters: highpass and lowpass filter cutoff frequency.

- Drag the cutoff frequency line—the upper line is lowpass and the lower line is highpass—to independently adjust filter cutoff values. Both cutoff frequencies can be adjusted simultaneously by dragging in the area between them.

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

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

Edit pan in the Tap display
The way the Pan parameter is represented in the Pan view is entirely dependent on the input channel configuration—mono to stereo or stereo to stereo.

- In mono input/stereo output configurations, all taps are initially panned to the center.
- In stereo input/stereo output configurations, the Pan parameter adjusts the stereo balance, not the position of the tap in the stereo field.

Note: Pan is not available in mono configurations.
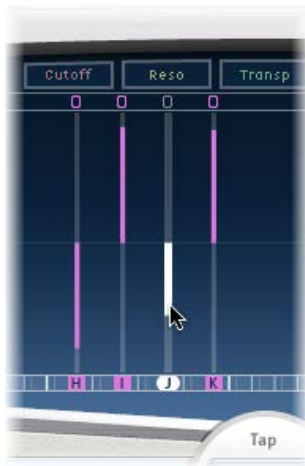
- *To edit the pan position in mono input/stereo output configurations:* Drag vertically from the center of the tap in the direction you want to pan the tap or taps.

  A white line extends outward from the center in the direction you have dragged, reflecting the pan position of the tap or taps.

  Lines above the center position indicate pans to the left, and lines below the center position denote pans to the right. Left (blue) and right (green) channels are easily identified.
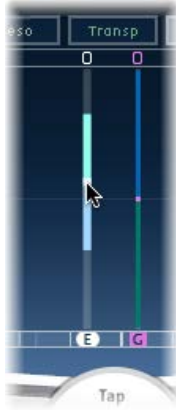
- *To adjust the stereo balance in stereo input/stereo output configurations:* Drag the Pan parameter— which appears as a dot on the tap—up or down the tap to adjust the stereo balance.

  By default, stereo spread is set to 100%. To adjust the spread width, drag either side of the dot. As you do so, the width of the line extending outward from the dot changes. Keep an eye on the Spread parameter in the Tap parameter bar while you are adjusting.

Edit taps with shortcut menu commands

■ Control-click (or right-click) a tap in the Tap display, then choose one of the following commands from the shortcut menu:

- *Copy sound parameters:* Copies all parameters (except the delay time) of the selected tap or taps to the Clipboard.

- *Paste sound parameters:* Pastes the tap parameters from the Clipboard into the selected tap or taps. If there are more taps in the Clipboard than are selected in the Tap display, the extra taps in the Clipboard are ignored.

- *Reset sound parameters to default values:* Resets all parameters of all selected taps (except the delay time) to the default values.

- *2 x delay time:* Doubles the delay time of all selected taps. For example, the delay times of three taps are set as follows: Tap A = 250 ms, Tap B = 500 ms, and Tap C = 750 ms. If you select these three taps and choose "2 x delay time," the taps are changed as follows: Tap A = 500 ms, Tap B = 1000 ms, and Tap C = 1500 ms. In other words, a rhythmic delay pattern unfolds half as fast. (In musical terms, it is played in half time.)

- *1/2 x delay time:* Halves the delay time of all selected taps. Using the example above, choosing "1/2 x delay time" changes the taps as follows: Tap A = 125 ms, Tap B = 250 ms, and Tap C = 375 ms. In other words, a rhythmic delay pattern unfolds twice as fast. (In musical terms, it is played in double time.)

- *Delete tap(s):* Deletes all selected taps.

Delay Designer Tap parameter bar

The Tap parameter bar provides access to all parameters of the selected tap. It also shows several parameters that are not available in the Tap display, such as Transpose and Flip.

Editing the parameters of a single, selected tap is fast and precise because all parameters are visible, with no need to switch display views or estimate values with vertical lines. If you choose multiple taps in the Tap display, the values of all selected taps are changed relative to each other.

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

Tap parameter bar controls

- *Filter On/Off button:* Turns the highpass and lowpass filters on or off (for the selected tap).
- *HP-Cutoff-LP fields:* Drag to set the cutoff frequencies (in Hz) for the highpass and lowpass filters.
- *Slope buttons:* Determine the steepness of the highpass and lowpass filter slope. Click the 6 dB button for a gentler filter slope, or click the 12 dB button for a steeper, more pronounced filtering effect.

  Note: You cannot set the slope of the highpass and lowpass filters independently.

- *Reso(nance) field:* Drag to set the amount of filter resonance for both filters.
- *Tap Delay fields:* Show the number and name of the selected tap in the upper section and the delay time in the lower section.
- *Pitch On/Off button:* Click to turn pitch transposition on or off (for the selected tap).
- *Transp(ose) fields:* The left field transposes pitch in semitones. The right field fine-tunes each semitone step in cents (1/100th of a semitone).
- *Flip buttons:* Swap the left and right side of the stereo image. Clicking these buttons reverses the tap position from left to right, or vice versa. For example, if a tap is set to 55% left, clicking the flip button swaps it to 55% right.
- *Pan field:* Drag to set pan position for mono signals or stereo balance for stereo signals.
  - Pan displays a percentage between 100% (full left) and −100% (full right), which represents the pan position or balance of the tap. A value of 0% represents the center panorama position.
- *Spread field:* Drag to set the width of the stereo spread for the selected tap in stereo-to-stereo instances.
- *Mute button:* Click to mute (silence) or unmute the selected tap.
- *Level field:* Drag to set the output level for the selected tap.

Delay Designer sync mode

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

When sync mode is on, a grid that matches the chosen Grid parameter value is shown in the Identification bar. All taps are moved toward the closest delay time value on the grid. Subsequently created or moved taps are snapped to positions on the grid.

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

Note: Delay Designer has a maximum delay time of 10 seconds. This means that if you load a setting into a project with a slower tempo (than the setting's tempo), some taps may fall outside the 10 second limit. In such cases, these taps do not play but are retained as part of the setting.

Sync parameters

• *Sync button:* Turns synchronized mode on or off.

• *Grid pop-up menu:* Choose a grid resolution from several musical note durations. The grid resolution, along with the project tempo, determines the length of each grid increment. As you change grid resolutions, the increments shown in the Identification bar change accordingly. This also determines a step limitation for all taps.

For example, imagine a project with a tempo of 120 bpm. The Grid pop-up menu value is set to 1/16 notes. At this tempo and grid resolution, each grid increment is 125 milliseconds (ms) apart. If Tap A is currently set to 380 ms, turning on sync mode shifts Tap A to 375 ms. If you try to move Tap A forward in time, it snaps to 500 ms, 625 ms, 750 ms, and so on. At a resolution of 1/8 notes, the steps are 250 milliseconds apart, so Tap A automatically snaps to the nearest division (500 ms) and could be moved to 750 ms, 1000 ms, 1250 ms, and so on.

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

    Tip: Use subtle grid position variations of every second increment (values between 45% and 55%) to create a less rigid rhythmic feel. High Swing values are unsubtle because they place every second increment directly beside the subsequent increment. Make use of higher values to create interesting and intricate double rhythms with some taps, while retaining the grid to lock other taps into more rigid synchronization with the project tempo.

Delay Designer master parameters

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

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

Delay Designer's global Feedback parameter does, however, enable you to send the output of one user-defined tap back through the effect input, to create a self-sustaining rhythm or pattern. This tap is known as the *feedback tap*.

Master parameters

- *Feedback button:* Turn the feedback tap on or off.
- *Feedback Tap pop-up menu:* Choose a tap as the feedback tap.
- *Feedback Level knob:* Rotate to set the feedback tap output level (before it is routed back into Delay Designer's input).
    - A value of 0% equals no feedback.
    - A value of 100% sends the feedback tap back into Delay Designer's input at full volume.

    Note: If Feedback is enabled and you begin creating taps with the Tap pads, Feedback is automatically turned off. When you stop creating taps with the Tap pads, Feedback is automatically re-enabled.

- *Mix sliders:* Drag to independently set the levels of the dry input signal and the post-processing wet signal.

Echo

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

Echo parameters

- *Time pop-up menu:* Choose 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:* Drag to determine how often the delay effect is repeated.
- *Color slider and field:* Drag to set the harmonic content (color) of the delay signal.
- *Dry and Wet sliders and fields:* Drag to set 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 correcting 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 emulates an acoustic (microphone) separation of 10 centimeters.
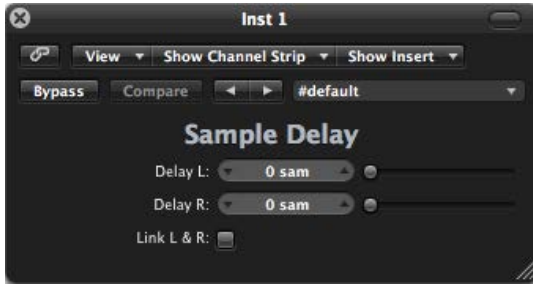
Sample Delay parameters

- *Delay slider and field (L and R in stereo version):* Drag to set the number of samples that the incoming signal is delayed by.
- *Link L & R button (only in stereo version):* Turn on to make sure that the number of samples is identical for both channels. Adjusting one channel value adjusts the other.

Stereo Delay

Stereo Delay lets you set the Delay, Feedback, and Mix parameters separately for the left and right channels. The Crossfeed knob (for each stereo side) sets the feedback intensity level of each signal being routed to the opposite stereo side. You can use Stereo Delay on mono tracks or busses when you want to create independent delays for the two stereo sides.

Note: If you use Stereo Delay 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.

The parameters for the left and right delays are identical. The descriptions below describe the left channel—the right channel parameter title is provided in parentheses, if named differently. Parameters common to both channels are described after the channel parameters.

Channel parameters

- *Left (Right) Input pop-up menu:* Choose the input signal for the two stereo sides. Options include Off, Left, Right, L + R, and L − R.
- *Left (Right) Delay field:* Drag to set the delay time in milliseconds. (The parameter is dimmed when you synchronize the delay time with the project tempo.)
- *Groove slider and field:* Drag to determine the proximity of every second delay repeat to the absolute grid position—in other words, how close every second delay repeat is.
- *Note buttons:* Click to 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:* Rotate to set the amount of feedback for the left and right delay signals.
- *Crossfeed Left to Right (Crossfeed Right to Left) knob and field:* Rotate to transfer the feedback signal of the left channel to the right channel, and vice versa.
- *Feedback Phase button:* Click to invert the phase of the corresponding channel's feedback signal.
- *Crossfeed Phase button:* Click to invert the phase of the crossfed feedback signals.

Common parameters

- *Beat Sync button:* Turn on to synchronize delay repeats to the project tempo.
- *Output Mix (Left and Right) sliders and fields:* Drag to independently control the level of the left and right channel signals.
- *Low Cut and High Cut sliders and fields:* Drag to cut frequencies below the Low Cut value and above the High Cut value from the source signal.

Tape Delay

Tape Delay simulates the sound of vintage tape echo machines. It can run at a free rate or can be synchronized with the project tempo. The effect is equipped with a highpass and lowpass filter in the feedback loop, making it easy to create authentic dub echo effects. Tape Delay also includes an LFO for delay time modulation. This can be used to produce chorus effects, even on long delays.

Tape Delay parameters

- *Feedback slider:* Drag to set the amount of delayed and filtered signal that is routed back to the input. Set to the lowest possible value to generate a single echo. Set to 100% to endlessly repeat the signal. The levels of the original signal and taps (echo repeats) tend to accumulate and may cause distortion. Use the internal tape saturation circuit to make these overdriven signals sound pleasant.
- *Freeze button:* Captures the current delay repeats and sustains them until Freeze is turned off.
- *Delay field:* Drag to set the delay time in milliseconds. (This parameter is dimmed when you synchronize the delay time to the project tempo.)
- *Sync button:* Click to synchronize delay repeats with the project tempo (including tempo changes).
- *Tempo field:* Drag to set the delay time in beats per minute. (This parameter is dimmed when you synchronize the delay time to the project tempo.)
- *Groove slider and field:* Drag to determine 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:* Drag to cut frequencies below the Low Cut value and above the High Cut value from the source signal. You can shape the sound of taps (delay repeats) 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 toward the left. For ever thinner echoes, move the Low Cut slider toward the right. If you can't hear the effect, check the Dry and Wet controls *and* the filter settings.
- *Smooth slider and field:* Drag to even out the LFO and flutter effect.
- *LFO Rate knob and field:* Rotate to set the speed of the LFO.
- *LFO Depth knob and field:* Rotate to set 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:* Drag to set the speed variation.
  - *Flutter Intensity:* Drag to determine the intensity of the effect.
- *Dry and Wet sliders and fields:* Drag to independently control the amount of original and effect signal.
- *Distortion Level slider and field (Extended Parameters area):* Drag to set the level of the distorted (tape saturation) signal.

Distortion effects

3

Distortion effects overview

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. They are still used in musical instrument amplifiers today. When overdriven, tubes produce a musically pleasing distortion that 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 tone, 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.

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.
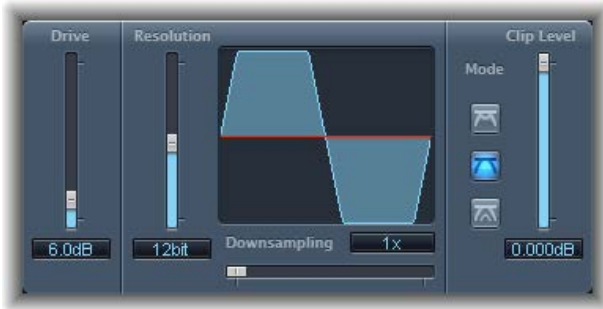
Bitcrusher parameters

- *Drive slider and field:* Drag to set the amount of gain applied to the input signal (in decibels).

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

- *Resolution slider and field:* Drag to set 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:* Drag to reduce the sample rate. A value of 1x has no effect on the signal, a value of 2x halves the sample rate, and a value of 10x reduces the sample rate to one-tenth of the original. (For example, if you set Downsampling to 10x, 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 button:* The center portion of the signal is halved in level above the threshold, resulting in a softer distortion. The start and end levels of the clipped signal are unchanged.

    - *Cut button:* Causes abrupt distortion when the clipping threshold is exceeded. Clipping that occurs in most digital systems is closest to Cut mode.

    - *Displaced button:* The start, mid, and end levels of the signal (above the threshold) are offset, resulting in less severe distortion 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:* Drag to set the point (below the clipping threshold of the channel) at which the signal starts clipping.

- *Mix slider and field (Extended Parameters area):* Drag to set 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 heavy distortions.

Clip Distortion has an unusual combination of serially connected filters. The incoming signal is amplified by the Drive value, passes through a highpass filter, then is 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 good with this sort of nonlinear distortion.

Clip Distortion parameters

- *Drive slider and field:* Drag to set 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:* Drag to set 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:* Drag to set the amount of nonlinear (asymmetrical) distortion applied to the signal.
- *Clip Filter slider and field:* Drag to set the cutoff frequency (in hertz) of the first lowpass filter.
- *Mix slider and field:* Drag to set the ratio between the effect (wet) signal and original (dry) signals, following the Clip Filter.
- *Sum LPF knob and field:* Drag to set the cutoff frequency (in hertz) of the lowpass filter. This processes the mixed signal.
- *(High Shelving) Frequency knob and field:* Rotate to set 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:* Rotate to set the amount of gain applied to the output of signals above the high shelving filter frequency.
- *Input Gain field and slider (Extended Parameters area):* Drag to set the amount of gain applied to the input signal.
- *Output Gain field and slider (Extended Parameters area):* Drag to set the amount of gain applied to the output signal.

Distortion effect

The Distortion effect simulates the low fidelity distortion generated by a bipolar transistor. You can use it to simulate playing a musical instrument through a highly overdriven amplifier or to create unique distorted sounds.

Distortion parameters

- *Drive slider and field:* Drag to set the amount of saturation applied to the signal.
- *Display:* Shows the impact of parameters on the signal.
- *Tone knob and field:* Rotate to set the frequency for the high cut filter. Filtering the harmonically rich distorted signal produces a softer tone.
- *Output slider and field:* Drag to set the output level. This enables you to compensate for increases in loudness caused by adding distortion.
- *Level Compensation checkbox (Extended parameter):* Turn on to reference the overall processing of the signal to 0 dB, making the output louder.

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 can use it creatively when designing new sounds.

Distortion II parameters

- *PreGain knob:* Rotate to set the amount of gain applied to the input signal.
- *Drive knob:* Rotate to set the amount of saturation applied to the signal.
- *Tone knob:* Rotate to set 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.
  - *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), 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.

Overdrive parameters

- *Drive slider and field:* Drag to set the saturation amount for the simulated transistor.
- *Display:* Shows the impact of parameters on the signal.
- *Tone knob and field:* Rotate to set the frequency for the high cut filter. Filtering the harmonically rich distorted signal produces a softer tone.
- *Output slider and field:* Drag to set the output level. This enables 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 overview on page 187). 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.

Phase Distortion parameters

- *Monitor button:* Turn on to hear the input signal in isolation. Turn off to hear the mixed signal.
- *Cutoff knob and field:* Rotate to set the (center) cutoff frequency of the lowpass filter.
- *Resonance knob and field:* Rotate to emphasize frequencies surrounding the cutoff frequency.
- *Display:* Shows the impact of parameters on the signal.
- *Mix slider and field:* Drag to set the percentage of the effect signal mixed with the original signal.
- *Max Modulation slider and field:* Drag to set the maximum delay time.
- *Intensity slider and field:* Drag to set the amount of modulation applied to the signal.
- *Phase Reverse checkbox (Extended Parameters area):* Select 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

Dynamics processors overview

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 or to highlight the most important sounds, while ensuring that softer sounds are not lost in the mix.

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

- *Compressors:* Downward compressors behave like an automatic volume control, lowering the volume whenever it rises above a certain level, called the *threshold*.

  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 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.

Adaptive Limiter

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

Adaptive Limiter is typically used on the final mix, where it can be placed after a compressor, such as Multipressor, and before a final gain control, resulting in a mix of maximum loudness. Adaptive Limiter can produce a louder-sounding mix than can be achieved by normalizing the signal.

Note: Using Adaptive Limiter adds latency when the Lookahead parameter is active. The effect is typically used for mixing and mastering previously recorded tracks, not while recording. You should bypass Adaptive Limiter while recording.

Adaptive Limiter parameters

- *Input meters:* Show input levels in real time. The Margin field shows the highest input level. You can reset the Margin field by clicking it.
- *Input Scale knob and field:* Rotate to scale the input level. Scaling is useful for handling very high-level or low-level input signals. It "squeezes" the higher and lower signal levels into a range that allows the Gain knob to work effectively. Avoid input levels above 0 dBFS, which can result in unwanted distortion.
- *Gain knob and field:* Rotate to set the amount of gain after input scaling.
- *Out Ceiling knob and field:* Rotate to set the maximum output level, or ceiling. The signal will not rise above this.

- *Output meters:* Show output levels, enabling you to see the results of the limiting process. The Margin field shows the highest output level. You can reset the Margin field by clicking it.
- *Mode buttons (Extended Parameters area):* Click to choose the type of peak smoothing:
  - *OptFit:* Limiting follows a linear curve, which allows signal peaks above 0 dB.
  - *NoOver:* Avoids distortion artifacts from the output hardware by ensuring that the signal does not exceed 0 dB.
- *Lookahead field and slider (Extended Parameters area):* Drag to adjust the playback buffer size (how far in the future the file is analyzed for peaks).
- *Remove DC checkbox (Extended Parameters area):* Select to activate a highpass filter that removes direct current (DC) from the signal. DC can be introduced by lower-quality audio hardware.
- *Inter Sample Peak Detection checkbox (Extended Parameters area):* Select to detect inter-sample peaks in the signal.

Compressor

Compressor overview

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 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

- *Circuit Type pop-up menu:* Choose the type of circuit emulated by Compressor. The choices are Platinum, Studio or Vintage VCA or FET, and Vintage Opto.
- *Side Chain Detection pop-up menu:* Choose the signal type to exceed or fall below the threshold. Max uses the maximum level of each side-chained signal. Sum uses the summed level of all side-chained signals.
    - 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:* Rotate to set the time it takes for 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. The input (level) is shown on the x-axis and output (level) on the y-axis.
- *Release knob and field:* Rotate to set the time it takes for Compressor to stop reducing the signal after the signal level falls below the threshold.
- *Auto button:* Turn on to make the release time dynamically adjust to the audio material.
- *Ratio slider and field:* Drag to set the compression ratio—the ratio of signal reduction when the threshold is exceeded.
- *Knee slider and field:* Drag to set 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:* Drag to set the threshold level—signals above this threshold value are reduced in level.
- *Peak/RMS buttons:* Click to determine if signal analysis uses the Peak or RMS method when using the Platinum circuit type.
- *Gain slider and field:* Drag to set 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:* Drag to set 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. The choices are Off, Soft, Hard, and Clip.
- *Activity pop-up menu (Extended Parameters area):* Turns the side chain on or off.
- *Mode pop-up menu (Extended Parameters area):* Choose the type of filter used for the side chain. The choices are LP (lowpass), BP (bandpass), HP (highpass), ParEQ (parametric), and HS (high shelving).
- *Frequency slider and field (Extended Parameters area):* Drag to set the center frequency for the side-chain filter.
- *Q slider and field (Extended Parameters area):* Drag to set the width of the frequency band affected by the side-chain filter.
- *Gain slider and field (Extended Parameters area):* Drag to set the amount of gain applied to the side-chain signal.
- *Mix slider and field (Extended Parameters area):* Drag to set the balance between dry (source) and wet (effect) signals.

Use Compressor

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

Compressor Threshold and Ratio

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

The Ratio parameter 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.

For 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.

Compressor envelope times

The Attack and Release parameters shape the dynamic response of Compressor. The Attack parameter determines the time it takes after the signal exceeds the threshold level before 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 results of your settings for the Attack and Release parameters depend not only on the type of source material but on the compression ratio and threshold settings.

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*.

Other Compressor parameters

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

Note: If you turn on Auto Gain and RMS simultaneously, the signal may become oversaturated. If you hear any distortion, turn off Auto Gain and adjust the Gain slider until the distortion is inaudible.

Use a side chain with Compressor

Use of a side chain with a compressor is common. The dynamics (level changes) of another channel strip is used 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.

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.

1 Insert Compressor into a channel strip.

2 In the Compressor plug-in window header, choose the channel strip that carries the desired signal (side-chain source) from the Side Chain pop-up menu.

3 Choose the desired analysis method (Max or Sum) from the Side Chain Detection pop-up menu.

4 Adjust Compressor parameters.

DeEsser

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 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. DeEsser has extremely fast attack and release times.

When using 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 compared in 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.

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 DeEsser's interface, and the Suppressor parameters are on the right. The center section includes the Detector and Suppressor displays and the Smoothing slider.

DeEsser Detector parameters

- *Detector Frequency knob and field:* Rotate to set the frequency range for analysis.
- *Detector Sensitivity knob and field:* Rotate to set the degree of responsiveness to the input signal.
- *Monitor pop-up menu:* Choose the signal type that you want to monitor. 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 parameters

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

DeEsser common parameters

- *Detector and Suppressor frequency displays:* The upper display shows the Detector frequency range. The lower display shows the Suppressor frequency range (in hertz).

- *Smoothing slider:* Drag to set the reaction speed for the gain reduction start and end phases. Smoothing controls both the attack and release times, as they are used by compressors.

Enveloper

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 dynamics processors. In contrast to a compressor or expander, 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.

The most important Enveloper parameters 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 reverb-drenched tracks sound drier. This is particularly useful when you are working with drum loops, but it has many other applications as well.

Enveloper parameters

- *Threshold slider and field:* Drag to set the threshold level. Signals that exceed the threshold have their attack and release phase levels altered. In general, you should set the Threshold to the minimum value and leave it there. Only when you significantly raise the release phase level, which also boosts any noise in the original recording, should you raise the Threshold slider slightly. This limits Enveloper to affecting only the useful part of the signal.

- *(Attack) Gain slider and field:* Drag to boost or attenuate 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:* Drag to set the pre-read analysis time for the incoming signal. The Lookahead slider defines how far into the future of the incoming signal Enveloper looks, to anticipate upcoming events. You generally do not 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.

- *Display:* Shows the attack and release curves applied to the signal.

- *(Attack) Time knob and field:* Rotate to set the time it takes for the signal to increase from the threshold level to the maximum Gain level. Attack Time values of around 20 ms and Release Time values of 1500 ms are a good starting point.

- *(Release) Time knob and field:* Rotate to set the time it takes for the signal to fall from the maximum gain level to the threshold level.

- *(Release) Gain slider and field:* Drag to boost or attenuate 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:* Drag to set the level of the output 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.

Expander

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

Expander parameters

- *Threshold slider and field:* Drag to set the threshold level. Signals above this level are expanded.

- *Peak/RMS buttons:* Click to determine whether the Peak or RMS method is used to analyze the signal.

- *Attack knob and field:* Rotate to set the time it takes for Expander to respond to signals that exceed the threshold level.

- *Expansion display:* Shows the expansion curve applied to the signal.

- *Release knob and field:* Rotate to set the time it takes for Expander to stop processing the signal after it falls below the threshold level.

- *Ratio slider and field:* Drag to set the expansion ratio—the ratio of signal expansion when the threshold is exceeded.

  Note: Because 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:* Drag to determine 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:* Drag to set the amount of output gain.

- *Auto Gain button:* Turn on to compensate 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 adjusts the signal appropriately.

Limiter

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.

Limiter is used primarily when mastering. Typically, you apply 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.

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, Limiter reduces the level before clipping can occur. Limiter cannot, however, fix audio that is clipped during recording.

Limiter parameters

- *Gain reduction meter:* Shows the amount of limiting in real time.
- *Gain slider and field:* Drag to set the amount of gain applied to the input signal.
- *Lookahead slider and field:* Drag to adjust how far ahead (in milliseconds) 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 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:* Drag to set the time it takes for Limiter to stop processing, after the signal falls below the threshold level.
- *Output Level knob and field:* Rotate to set the output level of the signal.
- *Softknee button:* Turn on to limit the signal 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.
- *Inter Sample Peak Detection checkbox (Extended Parameters area):* Select to detect inter-sample peaks in the signal.

Multipressor

Multipressor overview

Multipressor (an abbreviation for *multiband compressor*) is a versatile audio mastering tool. It splits the incoming signal into different frequency bands—up to four—and enables you to independently compress each band. After compression is applied, the bands are combined into a single output signal.

The advantage of compressing different frequency bands separately is that it allows more compression to be applied to bands that need it, without affecting other bands. This avoids the "pumping" effect often associated with high amounts of compression.

Because the use of higher compression ratios on specific frequency bands is possible, Multipressor can achieve a higher average volume without causing audible artifacts.

Raising the overall volume level can result in a corresponding increase in the existing noise floor. Each frequency band features downward expansion, which enables you to reduce or suppress this noise.

Downward expansion works as a counterpart to compression. Whereas a compressor compresses the dynamic range of higher volume levels, the downward expander expands the dynamic range of the lower volume levels. With downward expansion, the signal is reduced in level when it falls below the threshold level. This works in a similar way to a noise gate, but rather than abruptly cutting off the sound, it smoothly fades the volume with an adjustable ratio.

Multipressor Display parameters

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

Graphic display section

Frequency band section   Output section
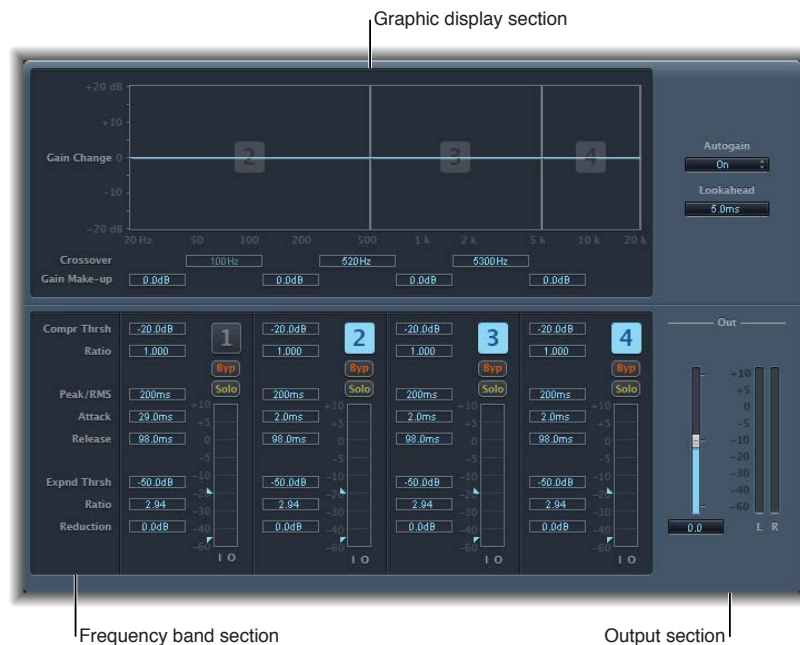
Display parameters

- *Graphic display:* Shows and allows adjustment of frequency and gain for each frequency band. The amount of gain change from 0 dB is indicated by blue bars. The band number appears in the center of active bands. You can adjust each frequency band in the following ways:
  - Drag the horizontal bar up or down to adjust the gain makeup for that band.
  - Drag the vertical edges of a band to the left or right to set the crossover frequencies, which adjusts the band's frequency range.
- *Crossover fields:* Drag to set the crossover frequency between adjacent bands.
- *Gain Make-up fields:* Drag to set the amount of the gain make-up for each band.

Multipressor Frequency Band parameters

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

Graphic display section

Frequency band section

Output section
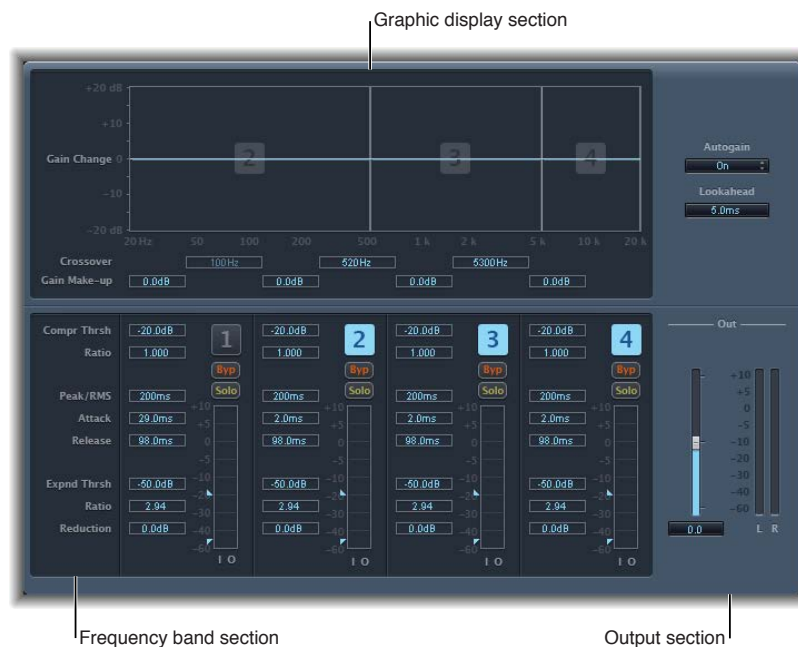
Frequency band parameters

- *Compr Thresh (Compression Threshold) fields:* Drag to set the compression threshold for the selected band. Setting the parameter to 0 dB results in no compression of the band.
- *Compr Ratio (Compression Ratio) fields:* Drag to set the compression ratio for the selected band. Setting the parameter to 1:1 results in no compression of the band.
- *Expnd Thresh (Expand Threshold) fields:* Drag to set the expansion threshold for the selected band. Setting the parameter to its minimum value (−60 dB) means that only signals that fall below this level are expanded.
- *Expnd Ratio (Expand Ratio) fields:* Drag to set the expansion ratio for the selected band.
- *Expnd Reduction (Expand Reduction) fields:* Drag to set the amount of downward expansion for the selected band.
- *Peak/RMS fields:* Drag to set a smaller value for shorter peak detection, or a larger value for RMS detection, in milliseconds.

- *Attack fields:* Drag to set the time before compression starts for the selected band, after the signal exceeds the threshold.
- *Release fields:* Drag to set the time before compression stops on the selected band, after the signal falls below the threshold.
- *Band on/off buttons:* Turn each band on or off. When enabled, the button is highlighted, and the corresponding band appears in the graphic display area above.
- *Byp(ass) buttons:* Turn on to bypass the selected frequency band.
- *Solo buttons:* Turn on to hear compression on only the selected frequency band.
- *Level meters:* The bar on the left shows the input level, and the bar on the right shows the output level.
- *Threshold arrows:* Two arrows appear to the left of each Level meter.
  - Click the upper arrow to adjust the Compression Threshold (Compr Thresh).
  - Click the lower arrow to adjust the Expansion Threshold (Expnd Thresh).

Multipressor Output parameters

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

Graphic display section

Frequency band section

Output section

Output parameters
- *Auto Gain button:* Turn on to reference the overall processing of the signal to 0 dB, making the output louder.
- *Lookahead value field:* Drag to set how far ahead the effect analyzes the incoming signal, allowing faster reactions to peak volumes.
- *Out slider:* Drag to set the overall gain at Multipressor's output.
- *Level meter:* Shows the overall output level.

Use Multipressor

In the graphic display, the blue bars show the gain change—not merely the gain reduction—as with a standard compressor. The gain change display is a composite value consisting of the compression reduction, plus the expander reduction, plus the auto gain compensation, plus the gain make-up.

Compression parameters

The Compression Threshold and Compression Ratio parameters are the key parameters for controlling compression. Usually the most useful combinations of these two settings are a low Compression Threshold with a low Compression Ratio, or a high Compression Threshold with a high Compression Ratio.

Downward Expansion parameters

The Expansion Threshold, Expansion Ratio, and Expansion Reduction parameters are the key parameters for controlling downward expansion. They determine the strength of the expansion applied to the chosen range.

Peak/RMS and Envelope parameters

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

Output parameters

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

Noise Gate

Noise Gate overview

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.

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

- *Threshold slider and field:* Drag to set the threshold level. Signals that fall below the threshold are reduced in level.
- *Reduction slider and field:* Drag to set the amount of signal reduction.
- *Attack knob and field:* Rotate to set the time it takes to fully open the gate after the signal exceeds the threshold.
- *Hold knob and field:* Rotate to set the time the gate is kept open after the signal falls below the threshold.
- *Release knob and field:* Rotate to set the time it takes to reach maximum attenuation after the signal falls below the threshold.
- *Hysteresis slider and field:* Drag to set 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 level is close to the threshold level.
- *Lookahead slider and field:* Drag to set how far ahead Noise Gate analyzes the incoming signal, allowing the effect to respond more quickly to peak levels.
- *Monitor button:* Turn on to hear the side-chain signal, including the effect of the High Cut and Low Cut filters.
- *High Cut slider and field:* Drag to set the upper cutoff frequency for the side-chain signal.
- *Low Cut slider and field:* Drag to set 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.

Use 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 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 gate. This is useful when the signal level hovers around the Threshold level, causing Noise Gate to switch on and off repeatedly, and producing the undesirable chattering effect. The Hysteresis slider essentially sets the 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, 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 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 Noise Gate.

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 above, 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 off monitoring to set a suitable Threshold level more easily.

Equalizers

5

Equalizers overview

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-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, an instrument, a vocal performance, or a 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 Channel EQ or Linear Phase 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 of any audio source, be it an individual audio signal or an entire mix.

Channel EQ

Channel EQ overview

Channel EQ is a 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 shows changes to the frequency curve of the audio signal in real time, allowing you to see which parts of the frequency spectrum may need adjustment.

You can use 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 display's controls make it easy to view and change the audio signal in real time.

Tip: The parameters of Channel EQ and Linear Phase EQ are identical, enabling you to freely copy settings between them. In MainStage, if you replace a Channel EQ with a Linear Phase EQ (or vice versa) in the same Insert slot, the current settings are automatically transferred to the new EQ.

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 parameters

- *Master Gain slider and field:* Drag to set 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:* Choose the sample resolution for the Analyzer. Choose from the following menu items: low (1024 points), medium (2048 points), and high (4096 points).
- *Band On/Off buttons:* Turn the corresponding band on or off. Each button's 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. The scale is shown in dB.
  - 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 pointer when it is moved over a pivot point.
- *Frequency fields:* Drag to adjust the frequency of each band.
- *Gain/Slope fields:* Drag to set the amount of gain for each band. For bands 1 and 8, this changes the slope of the filter.
- *Q fields:* Drag to 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:* Turns on 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):* Click to choose Peak or RMS.
- *Analyzer Decay slider and field (Extended Parameters area):* Drag to set 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.
  - *Strong:* Preserves most of the perceived bandwidth.
  - *Light or medium:* Allows some change as you raise or lower the gain.
  - *Asymmetric:* These 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 Strength setting, the actual Q values will be different than when the automation was recorded.

Channel EQ use tips

The way you use Channel EQ depends on the audio material and your intended outcome. 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. Watch 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 attention to sections where the signal distorts or clips. Use the graphic display or parameter controls to adjust the frequency bands.

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 or parameters so that only a narrow range of frequencies is affected, or widen it to alter a broader frequency 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 area around the zero line, drag the dB scale, on the left side of the graphic display upward. Drag downward to decrease the resolution.

Channel EQ Analyzer

The Analyzer uses 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 and frequency ranges.

The bands derived from FFT analysis are scaled logarithmically—there are more bands in higher octaves than in lower octaves.

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: High Analyzer 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 turn off the Analyzer or close the Channel EQ window after setting EQ parameters.

Linear Phase EQ

Linear Phase EQ overview

The high-quality Linear Phase EQ effect is similar to Channel EQ, sharing the same parameters and eight-band layout. You can copy settings between them. If you replace Channel EQ with Linear Phase EQ (or vice versa) in the same Insert slot, the current settings are automatically transferred to the new EQ.

Linear Phase EQ uses a different underlying technology that preserves the phase of the audio signal. Phase coherency is always maintained, even when you apply extreme EQ curves to the sharpest signal transients.

A further difference between Channel EQ and Linear Phase EQ is that the latter uses a fixed amount of CPU resources, regardless of the number of active bands. Linear Phase EQ also introduces greater amounts of latency.

Note: It is recommended that you use Linear Phase EQ for mastering recorded audio and avoid use when playing software instruments live, for example.

Linear Phase EQ parameters

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

Linear Phase EQ parameters

- *Master Gain slider and field:* Drag to set the overall output level of the signal after boosting or cutting individual frequency bands.
- *Analyzer button:* Click to turn the Analyzer on or off.
- *Pre/Post EQ button:* Click to determine if the Analyzer shows the frequency curve before or after EQ is applied, when Analyzer mode is active.
- *Resolution pop-up menu:* Choose the sample resolution for the Analyzer. Choose from the following menu items: low (1024 points), medium (2048 points), and high (4096 points).
- *Band On/Off buttons:* Turn the corresponding band on or off. Each button's 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. The scale is shown in dB.
  - 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 pointer when it is moved over a pivot point.
- *Frequency fields:* Drag to adjust the frequency of each band.
- *Gain/Slope fields:* Drag to set the amount of gain for each band. For bands 1 and 8, this changes the slope of the filter.
- *Q fields:* Drag to adjust the Q 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:* Click to turn on 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):* Click to choose Peak or RMS.

- *Analyzer Decay slider and field (Extended Parameters area):* Drag to adjust the decay rate (in dB per second) of the Analyzer curve (peak decay in Peak mode or an averaged decay in RMS mode).

- *Gain-Q Couple Strength pop-up menu (Extended Parameters area):* Choose the amount of Gain-Q coupling.

  - *Strong:* Preserves most of the perceived bandwidth.

  - *Light and medium:* Allows some change as you raise or lower the gain.

  - *Asymmetric:* Features 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 Strength setting, the actual Q values will be different than they were when the automation was recorded.

Linear Phase EQ use tips

Linear Phase EQ is typically used as a mastering tool that is inserted into master or output channel strips. The way you use Linear Phase EQ depends on the audio material and your intended outcome. A useful workflow for many situations is as follows: Set Linear Phase EQ to a flat response (no frequencies boosted or cut), turn on the Analyzer, then play the audio signal. Watch 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 attention to sections where the signal distorts or clips. Use the graphic display or parameter controls to adjust the frequency bands.

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. Use the Q parameter or parameters so that only a narrow range of frequencies is affected.

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 adjust the decibel scale of the graphic display by vertically dragging either the left or right edge of the dB scale 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 area around the zero line, drag the left side of the dB scale upward. Drag downward to decrease the resolution.

Linear Phase EQ Analyzer

The Analyzer uses 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 or frequency ranges.

The bands derived from FFT analysis are scaled logarithmically—there are more bands in higher octaves than in lower octaves.

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 −40 dB. The Analyzer display is always dB-linear.

Note: High Analyzer 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 turn off the Analyzer or close the Channel EQ window after setting EQ parameters.

Match EQ

Match EQ overview

Match EQ allows you to analyze and store the average frequency spectrum of an audio file as a template. You can apply this template to another audio signal so that it matches the spectrum of the original file. This is also known as a *fingerprint EQ*, where one sonic fingerprint is applied to another signal.

Match EQ enables you to acoustically match the tonal quality or overall sound of different songs you plan to include on an album, for example, or to impart the color of any source recording to your own projects.

Match EQ is a learning equalizer that analyzes the frequency spectrum of an audio signal such as an audio file, a channel strip input signal, or a template. The average frequency spectrum of the source file (the template) and of the current material (this can be the entire project or individual channel strips within it) is analyzed. These two spectra are then matched, creating a filter curve. This filter curve adapts the frequency response of the current material to match that of the template. Before applying the filter curve, you can modify it by boosting or cutting any number of frequencies or by inverting the curve.

The Analyzer allows you to visually compare the frequency spectrum of the source file and the resulting curve, making it easier to make manual corrections at specific points within the spectrum.

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

Match EQ parameters

Match EQ offers the following parameters.
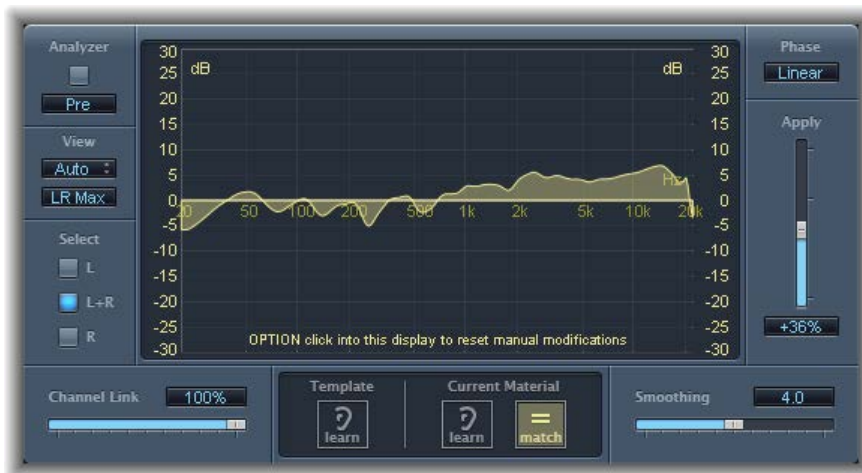
Match EQ parameters

- *Analyzer button:* Turns the Analyzer function on or off.
- *Pre/Post button:* Click to determine if the Analyzer looks at the signal before (Pre) or after (Post) the filter curve is applied.
- *View pop-up menu:* Set the information shown in the graphic display. Choices are:
  - *Auto:* Displays information for the current function, as set with the active button below the graphic display.
  - *Template:* Displays the learned frequency curve template for the source file. This is shown in red.
  - *Current Material:* Displays the frequency curve for the audio learned as current material. This is shown in green.
  - *Filter:* Displays the filter curve created by matching the template and the current material. This is shown in yellow.
- *View button:* Determines if separate curves are displayed by the Analyzer (L&R for stereo) or the summed maximum level is shown (LR Max).

  Note: The View parameters are disabled when you use the effect on a mono channel.
- *Select buttons:* Apply changes to the filter curve (created by matching the template with the current material) to: the left (L), right (R), or both channels (L+R).

  Note: The Select parameters are disabled when you use the effect on a mono channel.
- *Channel Link slider and field:* Drag to refine settings made with the Select buttons.
  - When set to 100%, L and R channels are represented by a common EQ curve.
  - When set to 0%, a separate filter curve is displayed for each channel (chosen with the Select buttons).
  - Settings between 0 and 100% blend these values with your filter curve changes for each channel. This results in a hybrid curve.

  Note: The Channel Link parameters are disabled when you use the effect on a mono channel.
- *Graphic display:* Displays the filter curve created by matching the template to the current material. You can edit the filter curve (see Edit the Match EQ filter curve on page 108).

- *Template Learn button:* Starts/stops the process of learning the frequency spectrum of the source file.
- *Current Material Learn button:* Starts/stops the process of learning the frequency spectrum of the project you want to match with the source file.
- *Current Material Match button:* Click to match the frequency spectrum of the current material to that of the template (source) file.
- *Phase pop-up menu:* Switches the operational principle of the filter curve.
  - *Linear:* Prevents processing from altering the signal phase, but latency is higher.
  - *Minimal:* Alters the signal phase (slightly), but latency is reduced.
  - *Minimal, Zero Latency:* Adds no latency, but has a higher CPU overhead than the other options.
- *Apply slider and field:* Drag to determine the impact of the filter curve on the signal.
  - Values above 100% magnify the effect.
  - Values below 100% reduce it.
  - Negative values (−1% to −100%) invert the peaks and troughs in the filter curve.
  - A value of 100% has no impact on the filter curve.
- *Smoothing slider and field:* Drag to set the amount of smoothing for the filter curve, using a constant bandwidth set in semitone steps. A value of 0.0 has no impact on the filter curve. A value of 1.0 means a smoothing bandwidth of one semitone. A value of 4.0 means a smoothing bandwidth of four semitones (a major third). A value of 12.0 means a smoothing bandwidth of one octave, and so on.

  Note: Smoothing has no effect on any manual changes you make to the filter curve.

- *Fade Extremes checkbox (Extended Parameters area):* Select to smooth the filter curve at the high and low extremes of the frequency spectrum.

Use Match EQ

These tasks are those commonly used with Match EQ to match the frequency spectrum of a mix with the spectrum of a source audio file. You can adapt some, or all, to your own workflow.

Learn or create a Match EQ template

You can drag an audio file to the Template Learn or Current Material Learn buttons for use as either the template or the current material. A progress bar appears while Match EQ is analyzing the file. You can also load a previously saved plug-in setting, or you can import the settings of another unsaved Match EQ instance by copying and pasting.

Do one of the following:

- Drag an audio file from the Finder to the Template Learn button, and select the source channel strip as a sidechain.
- Use Match EQ on the source channel strip and save a setting. Import this setting into the target Match EQ instance.

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

Match the EQ of a project mix to the EQ of a source audio file

1 In the project you want to match to the source audio file, insert Match EQ (typically on Output 1-2).

2 Drag the source audio file to the Template Learn button.

3 Return to the start of your mix, click Current Material Learn, and play your mix (the current material) from start to finish.

4 When you are done, click Current Material Match (this automatically turns off the Current Material Learn button).

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

Note: Only one of the Learn buttons can be turned on at a time. For example, if the Learn button in the Template section is on and you click the Learn button in the Current Material section, analysis of the template file stops, the current status is used as the spectral template, and analysis of the incoming audio signal (Current Material) begins.

Edit spectra with the Match EQ shortcut menu
This menu provides commands that can be applied to the spectrums of either the template or the current material.

- Control-click (or right-click) either Learn button, then choose one of the following from the shortcut menu:
  - *Clear Current Material Spectrum:* Clears the current spectrum.
  - *Copy Current Spectrum:* Copies the current spectrum to the Clipboard (this can be used by any Match EQ instance in the current project).
  - *Paste Current Spectrum:* Pastes the Clipboard contents to the current Match EQ instance.
  - *Load Current Material Spectrum from setting file:* Loads the spectrum from a stored setting file.
  - *Generate Current Material Spectrum from audio file:* Generates a frequency spectrum for an audio file that you have chosen.

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

Do either of the following:

- Drag the Apply slider down from the default 100% value to avoid extreme spectral changes to your mix.

- Drag the Smoothing slider to adjust the spectral detail of the generated EQ curve—if required.

Use the matched EQ on a channel strip

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

1 Choose the channel strip that you want to match from the Sidechain pop-up menu of the Match EQ window.

2 Click the Template Learn button.

3 Play the entire source audio file from start to finish. To stop the learn process, click the Template Learn button again.

4 Return to the start of your mix, click Current Material Learn, and play your mix (the current material) from start to finish.

5 When you are done, click Current Material Match (this automatically turns off the Current Material Learn button).

Edit the Match EQ filter curve

You can edit the filter curve in the graphic display by adjusting the various points shown in each band. As you drag a point, the current value appears in a small box inside the graphic display, allowing precise changes.

Adjust Match EQ curve values

Do any of the following:

- To shift the peak frequency for the band (over the entire spectrum), drag horizontally.

- To adjust the gain of the band, drag vertically.

- To adjust the Q Factor, Shift-drag vertically.

- To reset the gain to 0 dB, Option-drag.

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

  The Q factor of the filter is determined (and set) by the vertical distance between the clicked position and the curve.

Set the Q factor in Match EQ

Do either of the following:

- To set the maximum Q value of 10 (for notch-like filters), click the curve.

- To decrease the Q value, click above or below the curve. The farther you click from the curve, the smaller the value (down to the minimum of 0.3).

Change the Match EQ scale range

The colors and modes of the dB scales on the left and right of the display are automatically adapted to the active function. If the Analyzer is active, the left scale displays the average spectrum in the signal, while the right scale serves as a reference for the peak values of the Analyzer. A dynamic range of 60 dB is shown by default. If this is not precise enough for your edits, you can increase the range.

- Drag either scale to set values of up to +20 dB and −100 dB.

Change Match EQ gain with the scales

- Drag either scale to adjust the overall gain of the filter curve from −30 to +30 dB.

The left scale—and the right, if the Analyzer is inactive—shows the dB values for the filter curve.

Single-Band EQ

The single-band EQ can operate in several modes. When you choose an EQ from the EQ Mode pop-up menu, the parameters shown below change. You can choose:

- *Low Cut or High Cut Filter:* Low Cut Filter attenuates the frequency range that falls below the selected frequency. High Cut Filter attenuates the frequency range above the selected frequency.
- *High Shelf or Low Shelf EQ:* Low Shelving EQ affects only the frequency range that falls below the selected frequency. High Shelving EQ affects only the frequency range above the selected frequency.
- *Parametric EQ:* 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.

Single-Band EQ parameters

- *Frequency slider and field:* Drag to set the cutoff frequency.
- *Gain slider and field (Shelf and Parametric filters only):* Drag to set the amount of cut or boost.
- *Slope pop-up menu (Cut filters only):* Choose the amount of cut, in decibels per octave. The higher the value, the more pronounced the effect.
- *Q-Factor slider and field:* Drag to set the Q (bandwidth).

Filter effects

6

Filter effects overview

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

MainStage 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. They are not usually used as "effects" per se, but as tools to refine the frequency spectrum of a sound or mix. See Equalizers overview on page 98.

AutoFilter

AutoFilter overview

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.

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

- *Threshold slider:* Sets an input level that—if exceeded—triggers the envelope or LFO that dynamically modulates filter cutoff frequency. See AutoFilter threshold on page 111.

- *Envelope parameters:* Define how the filter cutoff frequency is modulated over time. See AutoFilter envelope on page 112.

- *LFO parameters:* Define how the filter cutoff frequency is modulated by the LFO. See AutoFilter LFO on page 113.

- *Filter parameters:* Control the tonal color of the filtered sound. See AutoFilter filter on page 114.

- *Distortion parameters:* Distort the signal both before and after the filter. See AutoFilter distortion on page 115.

- *Output parameters:* Set the level of both the dry and effect signal. See AutoFilter output on page 115.

AutoFilter threshold

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.

Note: Retriggering of the envelope or LFO occurs only if the Retrigger button is active.

You can use the envelope and LFO to modulate the filter cutoff frequency.

AutoFilter envelope

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

Envelope parameters

- *Attack knob and field:* Rotate to set the attack time for the envelope.
- *Decay knob and field:* Rotate to set the decay time for the envelope.
- *Sustain knob and field:* Rotate to set 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:* Rotate to set the release time for the envelope. This is triggered as soon as the input signal falls below the threshold.
- *Dynamic knob and field:* Rotate to determine the input signal modulation amount. You can modulate the peak value of the envelope section by varying this control.
- *Cutoff Mod. slider and field:* Drag to determine the impact of the envelope on the cutoff frequency.

AutoFilter LFO

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

LFO parameters

- *Coarse Rate slider and field:* Drag to set the speed of LFO modulation. Use to set the LFO frequency in hertz.

  Note: The labels shown for the Rate knob, slider, and field change when you activate Beat Sync. Only the Rate knob and field are then available.

- *Fine Rate knob:* Rotate to set the speed of LFO modulation. Use to fine-tune the LFO frequency.

- *Beat Sync button:* Turn on to synchronize the LFO to the host application tempo. You can choose from bar values, triplet values, and more by using the Rate knob and field.

- *Phase knob and field:* Rotate to set the phase relationship between the LFO rate and the host application tempo—when Beat Sync is active. This parameter is dimmed when Beat Sync is disabled.

- *Decay/Delay knob and field:* Rotate to set the time it takes for the LFO to go from 0 to its maximum value.

- *Rate Mod. knob and field:* Rotate to set 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:* Rotate to set the phase relationship of the LFO modulations between the two channels (stereo only).

- *Cutoff Mod. slider and field:* Drag to determine the impact of the LFO on the cutoff frequency.

- *Retrigger button:* Turn on to start the LFO waveform at 0 each time the threshold is exceeded.

- *Waveform buttons:* Click to select the shape of the LFO waveform: descending sawtooth, ascending sawtooth, triangle, pulse wave, or random.

- *Pulse Width slider and field:* Drag to alter the curve shape of the selected waveform.

AutoFilter filter

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

Filter parameters

- *Cutoff knob and field:* Rotate to set 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:* Rotate to boost or cut signals in the frequency band that surrounds the cutoff frequency. Very high Resonance values cause the filter to begin oscillating at the cutoff frequency. This self-oscillation occurs before you reach the maximum Resonance value.

- *Fatness slider and field:* Drag to boost 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:* Click to set the slope of the lowpass filter to 6, 12, 18, or 24 dB per octave.

  Note: Clicking one of these buttons automatically chooses the lowpass (LP) filter and slope, overriding any active State Variable Filter button.

AutoFilter distortion

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

Distortion parameters

- *Input knob and field:* Rotate to set the amount of distortion applied before the filter section processes the signal.

- *Output knob and field:* Rotate to set the amount of distortion applied after the filter section processes the signal.

AutoFilter output

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

Output parameters

- *Dry Signal slider and field:* Drag to set the amount of original, dry signal added to the filtered signal.

- *Main Out slider and field:* Drag to set the overall output level. This enables you to compensate for higher levels caused by the use of distortion or by the filtering process itself.

EVOC 20 Filterbank

EVOC 20 Filterbank overview

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.

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, and the Output parameters section along the right side.
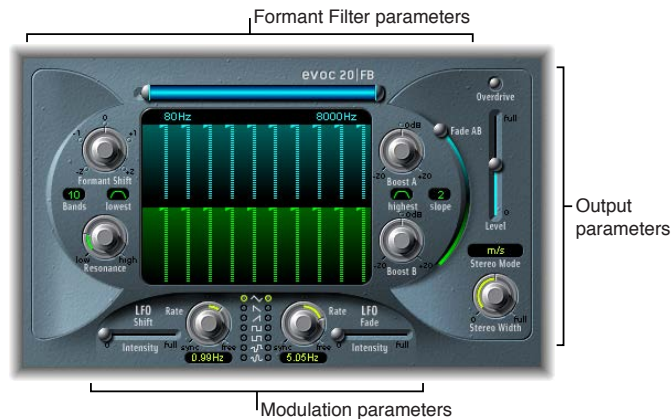
- *Formant Filter parameters:* Control the frequency bands in the two filter banks—the upper, blue, filter bank A and the lower, green, filter bank B. See EVOC 20 Filterbank Formant Filter on page 117.
- *Modulation parameters:* Control how Formant Filter parameters are modulated. See EVOC 20 Filterbank modulation on page 118.
- *Output parameters:* Control the overall output level and panning of the EVOC 20 Filterbank. See EVOC 20 Filterbank output parameters on page 119.

A short introduction to formants

A formant is a peak in the frequency spectrum of a sound. In the context of 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.

EVOC 20 Filterbank Formant Filter

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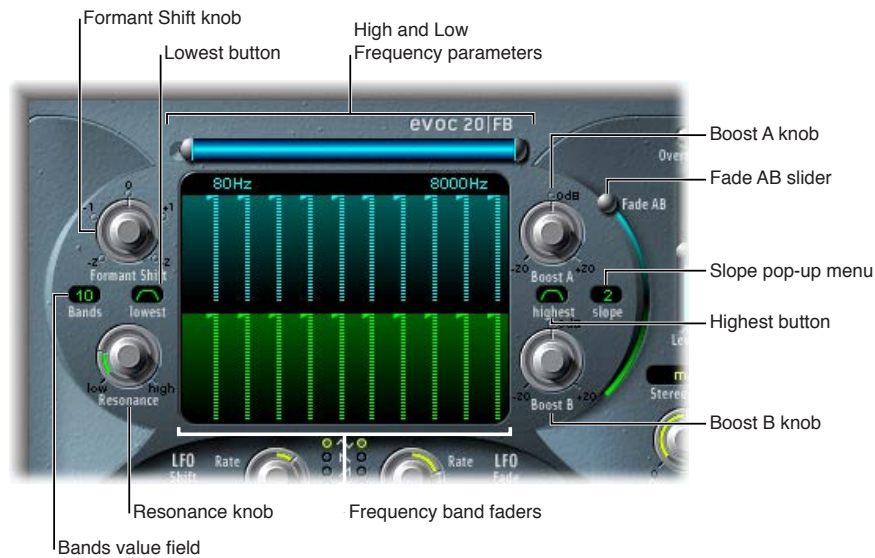
Formant Filter parameters

- *High and Low Frequency parameters:* Drag to 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. The silver handles on the left and right ends of the blue bar set the Low Frequency and High Frequency values, respectively. You can move the entire frequency range by dragging the blue bar.
  - You can also drag in the numeric fields that are below the blue bar to adjust the frequency values separately.
- *Frequency band faders:* Drag to set the level of each frequency band in filter bank A—the blue faders—or filter bank B—the green faders. You can quickly create complex level curves by dragging horizontally, or "drawing," across either row of faders.
- *Formant Shift knob:* Rotate to move 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:* Drag to set the number of frequency bands—up to 20—in each filter bank.
- *Lowest button:* Click to switch the lowest filter band between bandpass or highpass filter operation. In bandpass mode, the frequencies above and below the lowest band are ignored. In highpass mode, all frequencies below the lowest band are filtered.
- *Highest button:* Click to switch the highest filter band between bandpass or lowpass filter operation. In bandpass mode, the frequencies above and below the highest band are ignored. In lowpass mode, all frequencies above the highest band are filtered.
- *Resonance knob:* Rotate to determine the basic sonic character of both filter banks. High Resonance settings emphasize the center frequency of each band and result in a sharper, brighter character. Low settings result in a softer character.

- *Boost A and Boost B knobs:* Rotate to set the amount of boost—or cut—applied to the frequency bands in filter bank A or B. You can use these knobs to compensate for the reduction in volume caused by lowering the level of one or more bands. If you use Boost A and Boost B to set the mix relationship between filter bank levels, you can use Fade AB (see "Fade AB slider" below) to alter the tonal color, but not the levels.

- *Slope pop-up menu:* Choose the amount of filter attenuation applied to all filters in both filter banks. You can choose 1, which sounds softer at 6 dB/octave, or 2, which sounds brighter at 12 dB/octave.

- *Fade AB slider:* Drag to crossfade between filter bank A and filter bank B. At the top position, only bank A is audible, and 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

The modulation section contains two LFOs. The Shift LFO parameters on the left side control the Formant Shift parameter, and the Fade LFO parameters on the right side control the Fade AB parameter.

Modulation parameters

- *Shift LFO Intensity slider:* Drag to set the amount of Formant Shift modulation by the Shift LFO.

- *Rate knobs and fields:* Rotate to set the speed of modulation. Values to the left are synchronized with the host application tempo and include bar values, triplet values, and so on. Values to the right are nonsynchronized, or free, 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.

- *Waveform buttons:* Click to set the waveform type used by the Shift LFO (left column) or the Fade LFO (right column). You can choose from the following waveforms for each LFO:
    - 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)
    - Random stepped waveform (S&H)
    - Smoothed random waveform

- *Fade LFO Intensity slider:* Drag to control the amount of Fade AB modulation by the Fade LFO.

    Tip: LFO modulations are the key to interesting effects. 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. Also try inserting 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.

Output parameters

- *Overdrive button:* Click to turn the overdrive circuit on or off.

  Note: To hear the overdrive effect, you might need to boost the level of one or both filter banks.

- *Level slider:* Drag to set the volume of the output signal.

- *Stereo Mode pop-up menu:* Choose the input/output mode.

  - In s/s mode (stereo input/stereo output), the left and right channels are processed by separate filter banks.

  - In m/s mode (mono input/stereo output), a stereo input signal is first summed to mono before being routed to the filter banks.

- *Stereo Width knob:* Rotate to distribute the output signals of the filter bands in the stereo field.

  - At the 0 position to the left, the outputs of all bands are centered.

  - At the centered position at the top, the outputs of all bands ascend from left to right.

  - At the full position to the right, the bands are output to the left and right channels alternately.

EVOC 20 TrackOscillator

EVOC 20 TrackOscillator overview

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.

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 EVOC 20 TrackOscillator is inserted into 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 EVOC 20 TrackOscillator is inserted into, a side chain signal, or the tracking oscillator.

Because you can select both the analysis and synthesis input signals, EVOC 20 TrackOscillator is not limited to pitch tracking effects; you can also use it 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. Or you could use it to process drum loops with side-chained signals, such as other drum loops or rhythmic guitar, clavinet, and piano parts.

Vocoder overview

The word *vocoder* is an abbreviation for *voice encoder*. A vocoder analyzes and transfers the sonic character of the audio signal arriving at its analysis input 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 may be familiar with tracks such as "O Superman" by Laurie Anderson, "Funkytown" 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.

The speech analyzer and synthesizer features of a vocoder are two bandpass filter banks. Bandpass filters allow a frequency band—a slice in the overall frequency spectrum—to pass through unchanged. Frequencies that fall outside the band are cut.

In the EVOC 20 plug-ins, these filter banks are named the *analysis* and *synthesis* banks. 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, 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 with voltage-controlled amplifiers (VCAs) in analog vocoders. Volume changes to the bands in the analysis filter bank are imposed on the matching bands in the synthesis filter bank. These filter level changes are heard as a synthetic reproduction of the original input signal—or a mix of the two filter bank signals.

EVOC 20 TrackOscillator interface

EVOC 20 TrackOscillator is divided into a number of parameter sections.
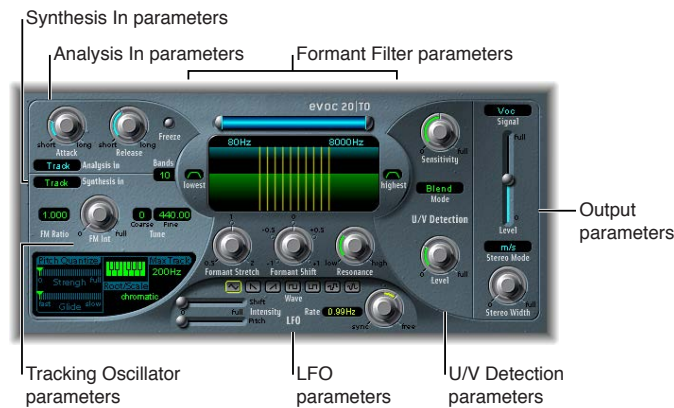
- *Analysis In parameters:* Determine how the input signal is analyzed and used by the analysis filter bank. See EVOC 20 TrackOscillator analysis in parameters on page 122.
- *U/V Detection parameters:* Detect the unvoiced portions of the sound in the analysis signal, improving speech intelligibility. See EVOC 20 TrackOscillator U/V detection parameters on page 124.
- *Synthesis In parameters:* Determine how the input signal is used by the synthesis filter bank. See EVOC 20 TrackOscillator synthesis in parameters on page 125.
- *Tracking Oscillator parameters:* Determine how the analysis input signal is used by the oscillator. See Tracking oscillator parameters on page 126.
- *Formant Filter parameters:* Configure the analysis and synthesis filter banks. See EVOC 20 TrackOscillator formant filter on page 128.
- *LFO parameters:* Modulate either the oscillator pitch or the Formant Shift parameter. See EVOC 20 TrackOscillator modulation on page 129.
- *Output parameters:* Configure the output signal of the EVOC 20 TrackOscillator. See EVOC 20 TrackOscillator output parameters on page 130.

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.

Analysis In parameters

- *Attack knob:* Rotate to determine how quickly each envelope follower—coupled to each analysis filter band—reacts to rising signals.

- *Release knob:* Rotate to determine how quickly each envelope follower—coupled to each analysis filter band—reacts to falling signals.

- *Freeze button:* Turn on to hold—or freeze—the current analysis sound spectrum indefinitely. When Freeze is enabled, the analysis filter bank ignores the input source, and the Attack and Release knobs have no effect.

- *Bands field:* Drag to set the number of frequency bands analyzed and then used by the synthesis engine. Up to 20 bands can be used.

- *Analysis In pop-up menu:* Choose the analysis signal source:

  - *Track:* Uses the input audio signal of the channel strip the EVOC 20 TrackOscillator is inserted into as the analysis signal.

  - *SideCh(ain):* Uses a side chain as the analysis signal. You choose the side-chain source channel strip from the Side Chain pop-up menu in the upper-right corner 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.

Use EVOC 20 TrackOscillator analysis in

You should be precise with the Analysis In parameters in order to attain the best possible speech intelligibility and the most accurate tracking. Follow these tasks and tips to obtain the best results.

Set Attack and Release times

■ Rotate the Attack and Release knobs to set times that provide the most articulated sound.

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, translates into a less articulated vocoder effect. Set the Attack parameter to the lowest possible value to enhance articulation.

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, translates 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.

Freeze the input signal

■ Click the Freeze button to hold, or sustain, the sound spectrum of the analysis input signal.

By freezing the input signal you can capture a particular characteristic of the signal, which is then imposed as a complex sustained filter shape on the Synthesis section. Here are some 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.
- People cannot sustain sung notes indefinitely. To compensate for this human limitation, 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.

Tip: The Freeze parameter can be automated, which may be useful in this situation.

Set the number of filter bank bands

■ Vertically drag the Bands field to set the number of frequency bands the EVOC 20 TrackOscillator's filter bank uses.

The greater the number of frequency 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 such as speech and vocals 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, can produce unusual results. Processing a signal with no audible pitch, such as drum loop, also delivers unusual results, but 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 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 sounds 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.

EVOC 20 TrackOscillator includes an Unvoiced/Voiced detector. This unit detects the unvoiced portions of the sound in the analysis signal and then substitutes the corresponding portions in the synthesis signal with noise, 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.

A short introduction to formants

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

U/V detection parameters

- *Sensitivity knob:* Rotate to determine how responsive U/V detection is. Turn to the right for higher settings, where 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 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. (The U/V sound source is determined by the Mode menu, as described below.)

- *Mode menu:* Choose the sound sources used to replace the unvoiced content of the input signal.

  - *Noise:* Uses noise alone for the unvoiced portions of the sound.

  - *N + Syn (Noise + Synthesizer):* 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:* Rotate to set the volume of the signal used to replace the unvoiced content of the input signal.

  Important: Be careful with the Level knob, particularly when using a high Sensitivity value, 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 parameters
- *Synthesis In pop-up menu:* Choose the tracking signal source:
  - *Track:* Uses the input audio signal of the channel strip that EVOC 20 TrackOscillator is inserted into as the synthesis signal, which drives the internal synthesizer.
  - *SideCh (SideChain):* Uses a side chain as the synthesis signal. You choose the side-chain source channel from the Side Chain pop-up menu in the upper-right corner of the EVOC 20 TrackOscillator window.
  - *Osc. (Oscillator):* 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.

  Note: If you choose Side Chain and no Side Chain channel is assigned, EVOC 20 TrackOscillator reverts to Track mode.

- *Bands field:* Drag to set the number of frequency bands analyzed and then used by the synthesis engine.

EVOC 20 TrackOscillator oscillators

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. See EVOC 20 TrackOscillator synthesis in parameters.

Tracking oscillator parameters

- *FM Ratio field:* Drag to set 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:* Rotate to determine 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 field:* Drag to set the pitch offset of the oscillator in semitones.
- *Fine Tune field:* Drag to set the pitch offset in cents. One cent equals 1/100 of a semitone.

Use tracking oscillator pitch 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 strong pitch corrections and can be used creatively on unpitched material with high harmonic content, such as cymbals and high-hats.

Tracking oscillator pitch parameters

- *Pitch Quantize Strength slider:* Drag to determine how pronounced the automatic pitch correction is.
- *Pitch Quantize Glide slider:* Drag to set the amount of time pitch correction takes, allowing sliding transitions to quantized pitches.
- *Root/Scale keyboard:* Click notes to define the pitch or pitches that the tracking oscillator is quantized to.
- *Root/Scale pop-up menu:* Click below Scale to choose the scale that the tracking oscillator is quantized to.

  Note: There are two discrete fields available for this parameter—Root and Scale. The Root (key) can be changed independently of the scale chosen in the pop-up menu.

- *Max Track field:* Drag to set 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.

Choose a root or scale

1 Click the green value field under Scale to open the pop-up menu.

2 Choose the scale or chord you want to use as the basis for pitch correction.

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

Add notes to, or remove notes from, the chosen scale or chord

- *To add notes to the scale or chord:* Click unused keys on the small keyboard to highlight them in green.

- *To remove notes from the scale or chord:* Click selected notes, which then are no longer highlighted.

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" from the Root/Scale pop-up menu.

EVOC 20 TrackOscillator formant filter

EVOC 20 TrackOscillator features two formant filter banks—one for the Analysis In section and one for the Synthesis In section. The entire frequency spectrum of an incoming signal is analyzed by the Analysis section and is divided equally into a number of frequency bands. Each filter bank can control up to 20 of these frequency bands.

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 reflected in the Formant Filter display, thus providing feedback on what is happening to the signal as it is routed through the two formant filter banks.

Formant filter parameters

- *Low and High Frequency parameters:* Drag to determine the lowest and highest frequencies allowed to pass by the filter section. Frequencies that fall outside these boundaries are cut.
  - The length of the horizontal blue bar at the top represents the frequency range for both analysis and synthesis—unless Formant Stretch or Formant Shift is used. 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 drag in the numeric fields to adjust the frequency values separately.
- *Lowest button:* Click to switch the lowest filter band between bandpass or highpass filter operation. In bandpass mode, the frequencies above and below the lowest band are ignored. In highpass mode, all frequencies below the lowest band are filtered.
- *Highest button:* Click to switch the highest filter band between bandpass or lowpass filter operation. In bandpass mode, the frequencies above and below the highest band are ignored. In lowpass mode, all frequencies above the highest band are filtered.
- *Formant Stretch knob:* Rotate to change 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 High and Low Frequency parameters.

  When Formant Stretch is set to 0, the width and distribution of the bands in the synthesis filter bank at the bottom match 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.
- *Formant Shift knob:* Rotate to move all bands in the synthesis filter bank up or down the frequency spectrum.

  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 relation 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 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: Use of the Formant Stretch and Formant Shift parameters can result in the generation of unusual resonant frequencies when high Resonance settings are used.

- *Resonance knob:* Rotate to change 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.

EVOC 20 TrackOscillator modulation

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

Modulation parameters

- *Shift Intensity slider:* Drag to set the amount of formant shift modulation by the LFO.
- *Pitch Intensity slider:* Drag to set the amount of pitch modulation—vibrato—by the LFO.
- *Waveform buttons:* Click to set the waveform type used by the LFO. You can choose from the following waveforms:
  - 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)
  - Random stepped waveform (S&H)
  - Smoothed random waveform
- *LFO Rate knob and field:* Rotate to set the speed of modulation. Values to the left are synchronized with the host application tempo and include bar values, triplet values, and so on. Values to the right are nonsynchronized, or free, 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.

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.

Output parameters

- *Signal pop-up menu:* Choose the signal that is sent to the plug-in's main outputs:
  - *Voc(oder):* Hear the vocoder effect.
  - *Syn(thesis):* Hear only the synthesizer signal.
  - *Ana(lysis):* Hear only the analysis signal.

  Note: The last two settings are mainly useful for monitoring purposes.
- *Level slider:* Drag to set the overall volume of the output signal.
- *Stereo Mode pop-up menu:* Choose the input/output mode.
  - In s/s mode (stereo input/stereo output), the left and right channels are processed by separate filter banks.
  - In m/s mode (mono input/stereo output), a stereo input signal is first summed to mono before being routed to the filter banks.
- *Stereo Width knob:* Rotate to distribute the output signals of the synthesis section's filter bands in the stereo field.
  - At the 0 position to the left, the outputs of all bands are centered.
  - At the centered position, the outputs of all bands ascend from left to right.
  - At the Full position to the right, the bands are output—alternately—to the left and right channels.

Fuzz-Wah

Fuzz-Wah overview

The Fuzz-Wah plug-in emulates classic wah effects, combined with compression and fuzz distortion effects. 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, a lowpass, or—less commonly—a highpass filter.

Drag a panel to determine the order of the effects chain.

Wah parameters   Compressor parameters   Fuzz parameters
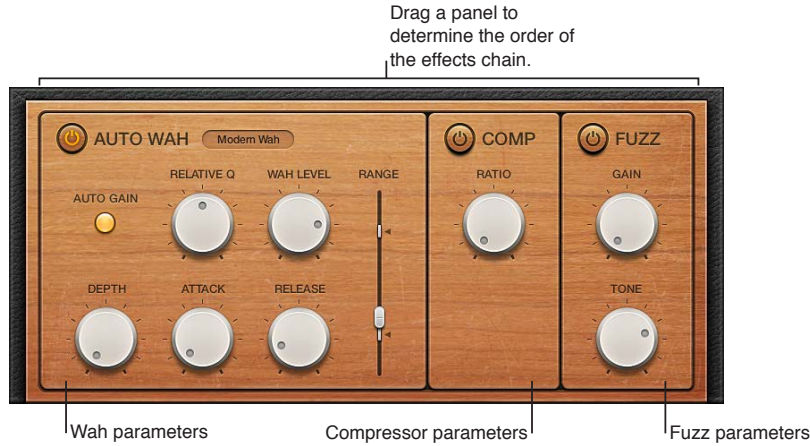
Fuzz-Wah effects work in series—where the output of one effect is fed into the next in an effects chain. The routing order lets you choose whether a distorted signal should be wah-filtered (for funkier sounds) or the wah-filtered sound should be distorted (for screaming sounds)—as an example.

Horizontally drag the name of the effect to determine the order of the effects chain.

Auto Wah parameters

These parameters determine the tone and behavior of the wah effect.

You can control the wah effect with the Auto Wah feature, which continually performs a filter sweep across the entire range. You can also control the wah sweep with MIDI foot pedals or other controllers.

Auto Wah parameters

- *On/off button:* Turns the Auto Wah effect on or off.
- *Wah Type pop-up menu:* Choose a Wah effect type.
  - *Classic Wah:* This setting mimics the sound of a popular wah pedal with a slight peak characteristic.
  - *Retro Wah:* This setting mimics the sound of a popular vintage wah pedal.
  - *Modern Wah:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting. The Q 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.
  - *Opto Wah 1:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting.
  - *Opto Wah 2:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting.
  - *Resonant LP:* In this mode, the Wah works as a resonance-capable lowpass filter. At the minimum pedal position, only low frequencies can pass.
  - *Resonant HP:* In this mode, the Wah works as a resonance-capable highpass filter. At the maximum pedal position, only high frequencies can pass.
  - *Peak:* In this mode, the Wah works as a peak (bell) filter. Frequencies close to the cutoff frequency are emphasized.
- *Auto Gain button:* Turn on to limit the output signal dynamic range. The wah effect can cause the output level to vary widely. Auto Gain compensates for this tendency and keeps the output signal within a more stable range.
- *Relative Q knob:* Rotate to adjust the main filter peak, resulting in a sharper or softer wah sweep.
- *Wah Level knob:* Rotate to set the amount of the wah-filtered signal.
- *Depth knob:* Rotate to set the depth of the Auto Wah effect. When set to 0, the automatic wah feature is disabled.
- *Attack knob:* Rotate to set the time it takes for the wah filter to fully open.
- *Release knob:* Rotate to set the time it takes for the wah filter to close.
- *(Pedal) Range slider:* Drag to sweep the wah filter. The two smaller sliders set the maximum and minimum values of the sweep range.

Fuzz-Wah Compressor parameters

The Compressor effect is normally used just before the Fuzz (distortion) effect. This allows you to increase or decrease the perceived gain, thus providing a suitable input level to the distortion circuit. You can, however, place the Compressor at any position in the effects chain or can disable it completely.

Compressor effect parameters

- *On/off button:* Turns the Compressor effect on or off.
- *Ratio knob:* Rotate to adjust the compression slope. The additional gain offered by the compression circuit—when directly preceding the Fuzz effect—lets you create crunchy distortions.

Fuzz parameters

These parameters control the integrated distortion circuit.

Fuzz parameters

- *On/off button:* Turns the Fuzz distortion effect on or off.
- *Fuzz Gain knob:* Rotate to set the level of distortion.
- *Fuzz Tone knob:* Rotate to adjust the tonal color of the distortion. Low settings tend to be warmer, and high settings are brighter and harsher.

Spectral Gate

Spectral Gate overview

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 Use Spectral Gate on page 135.
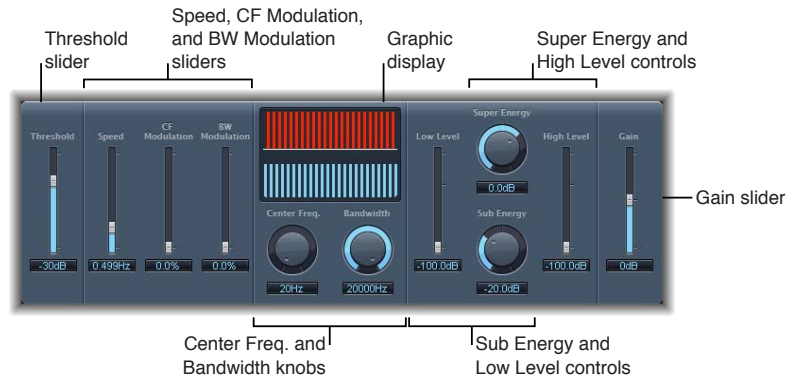
Spectral Gate parameters

- *Threshold slider and field:* Drag to set the threshold level that is used to divide 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:* Drag to set the modulation frequency for the defined frequency band.
- *CF (center frequency) Modulation slider and field:* Drag to set the intensity of center frequency modulation.
- *BW (bandwidth) Modulation slider and field:* Drag to set the amount of bandwidth modulation.
- *Graphic display:* Shows the frequency band defined by the Center Freq and Bandwidth parameters.
- *Center Freq(uency) knob and field:* Rotate to set the center frequency of the band that you want to process.
- *Bandwidth knob and field:* Rotate to set the width of the frequency band that you want to process.
- *Super Energy knob and field:* Rotate to set the level of the frequency range above the threshold.
- *High Level slider and field:* Drag to mix the frequencies of the original signal—above the selected frequency band—with the processed signal.
- *Sub Energy knob and field:* Rotate to set the level of the frequency range below the threshold.
- *Low Level slider and field:* Drag to mix the frequencies of the original signal—below the selected frequency band—with the processed signal.
- *Gain slider and field:* Drag to set the overall output level.

Use Spectral Gate

One way you can familiarize yourself with the operation of Spectral Gate is to start with a drum loop. Set Center Freq to its minimum value (20 Hz) and Bandwidth to its maximum value (20,000 Hz) so that the entire frequency range is processed. Turn up the Super Energy and Sub Energy knobs, one at a time, and then try different Threshold settings to get a sense of how different Threshold levels affect the sound of Super Energy and Sub Energy. When you have obtained a sound that you like or consider useful, narrow the Bandwidth parameter drastically, gradually increase 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 Modulation or BW Modulation sliders.

Get started with Spectral Gate

1 Rotate the Center Freq and Bandwidth knobs to set the frequency band you want to process.

    The graphic display shows the band that you define with these two parameters.

2 Drag the Threshold slider to set the appropriate level.

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

3 Rotate the Super Energy knob to control the level of the frequencies above the threshold, and rotate the Sub Energy knob to control the level of the frequencies below the threshold.

4 Drag the Low Level slider and the High Level slider to mix the frequencies that fall outside the defined frequency band with the processed signal.

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

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

5 Drag the Speed, CF Modulation, and BW Modulation sliders to modulate the defined frequency band.

    • Use the Speed slider to determine the modulation frequency.

    • Use the CF Modulation slider to define the intensity of the center frequency modulation.

    • Use the BW Modulation slider to control the amount of bandwidth modulation.

6 Drag the Gain slider to adjust the final output level of the processed signal.

Imaging processors

<div style="text-align: right; font-size: 3em;">7</div>

Imaging processors overview

The imaging processors are tools for manipulating the stereo image. You can use them 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.

Direction Mixer

Direction Mixer overview

You can use 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.

Direction Mixer works with any type of stereo recording, regardless of the miking technique used. For information about the most common stereo miking techniques—AB, XY, and MS—see Stereo miking techniques on page 137.

Direction Mixer parameters

- *Input buttons:* Click to set the input signal type. 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:* Drag to determine the spread of the stereo base in LR input signals or to set the side signal level in MS input signals. Spread parameter behavior changes when fed LR or MS signals. These differences are outlined below:

  When you are working with LR signals:

  - 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.

When you are 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.

- *Direction knob and field:* Rotate to set the pan position for the middle—the center of the stereo base—of the recorded stereo signal. 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.

When you are 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 either 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.

When you are 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 either 180° or −180°, the middle signal is dead center in the mix, but the left and right sides of the side signal are swapped.

Stereo miking techniques

There are three commonly used stereo miking variations used in recording: AB, XY, and MS. A stereo recording 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 middle signal, but the left and right channels are decoded from the side signal, which is the sum of both left and right channel signals.

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 because phase cancelations can occur between channels.

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. XY recording is commonly used for drum recording and 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.

MS miking

To make a Middle and Side (MS) recording, two microphones are positioned as closely together as possible—usually placed 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 generally used when mastering. There are several ways to extend the stereo base (or the perception of space), including using reverbs or other effects and altering the signal's phase. These options can sound good, but they can 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 it is used on mono recordings.
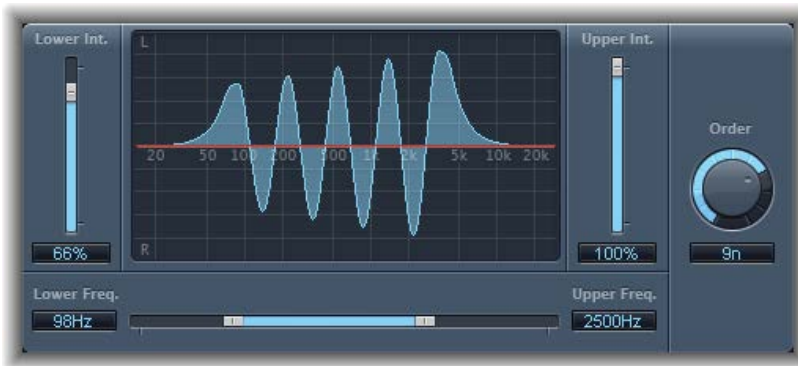
Stereo Spread parameters

- *Lower Int(ensity) slider and field:* Drag to set the amount of stereo base extension for the lower frequency bands.

- *Upper Int(ensity) slider and field:* Drag to set the amount of stereo base extension for the upper frequency bands.

  Note: When you are setting the Lower Int and Upper Int sliders, be aware that the stereo effect is most apparent in the middle and higher frequencies. Distributing low frequencies between the left and right speakers can significantly alter the energy of the overall mix. 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 effect intensity in the upper and lower frequency bands. The upper section represents the left channel. 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:* Drag to determine the highest and lowest frequencies that are redistributed in the stereo image.

- *Order knob and field:* Rotate to determine 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

Metering tools overview

You can use the Metering tools to analyze audio in a variety of ways. These plug-ins offer you different ways to view your audio than the meters shown in channel strips. The Metering plug-ins have no effect on the audio signal and are intended 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. For example, BPM Counter displays the tempo, Correlation Meter displays the phase relationship, and Level Meter displays the level of an incoming audio signal.

BPM Counter

BPM Counter analyzes 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 BPM Counter to detect the tempo.

Percussive drum and instrumental rhythm parts, such as basslines, are suitable for tempo analysis, whereas pad sounds are unsuitable candidates for tempo analysis.

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 BPM Counter.

Note: 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 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

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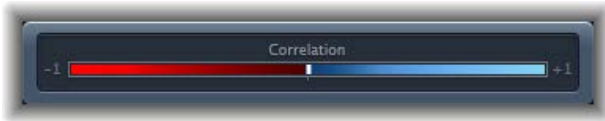
- A correlation of +1 (the far right position) means that the left and right channels correlate 100%—they are completely in phase.
- A correlation of 0 (the center position) indicates the widest permissible left/right divergence, often audible as an extremely wide stereo effect.
- Correlation values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

Level Meter plug-in

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 Level Meter instances show independent left and right bars, whereas mono instances display a single bar.
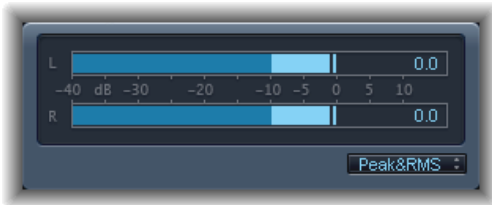
The current peak values are displayed numerically, superimposed over the graphic display. You can reset these values by clicking in the display.

Level Meter parameters
- *Display type pop-up menu:* Choose a display setting using Peak, RMS, Peak & RMS, Inter Sample Peak, Inter Sample Peak & RMS characteristics.

  The Inter Sample Peak options display inter-sample values.

  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 levels
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.

MultiMeter

MultiMeter overview

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

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

You can view either the Analyzer or Goniometer results in the main display area. You switch the view and set other MultiMeter parameters with the controls on the left side of the window.

Although you can insert MultiMeter directly into any channel strip, it is more commonly used in the master channel strip of the host application—when you are working on the overall mix.

MultiMeter Analyzer parameters

In Analyzer mode, MultiMeter's main display shows the frequency spectrum of the input signal as 31 independent frequency bands. Each frequency band represents one-third of an octave. The Analyzer parameters are used to activate Analyzer mode and to customize the way that the incoming signal is shown in the main display.

Analyzer parameters   Scale

MultiMeter Analyzer parameters

- *Analyzer button:* Switches the main display to Analyzer mode.
- *Left, Right, LRmax, and Mono buttons:* Click to determine the channels shown in the Analyzer results in the main display.
  - *Left or Right:* Displays the left or right channels.
  - *LRmax:* Displays the maximum level of the stereo inputs.
  - *Mono:* Displays the spectrum of the mono sum of both (stereo) inputs.
- *View fields:* Drag to alter the way values are shown in the Analyzer by setting the maximum level displayed (Top) and the overall dynamic range (Range).
- *Mode buttons:* Click to determine how levels are displayed. You can choose from Peak, Slow RMS, or Fast RMS characteristics.
  - The two RMS modes show the effective signal average and provide a representative overview of perceived volume levels.
  - The Peak mode shows level peaks accurately.
- *Scale (shown in main display):* Indicates the scale of levels. Drag the scale vertically to adjust. Changing the scale is useful when you are analyzing highly compressed material because it makes it easier to identify small level differences.

MultiMeter Goniometer parameters

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

The idea of the goniometer was born with the advent of early two-channel oscilloscopes. To use such devices as goniometers, users would connect the left and the right stereo channels to the X and Y inputs, while rotating the display by 45° to produce a useful visualization of the signal's stereo phase.

The signal trace slowly fades to black, imitating the retro glow of the tubes found in older goniometers, while also enhancing the readability of the display.
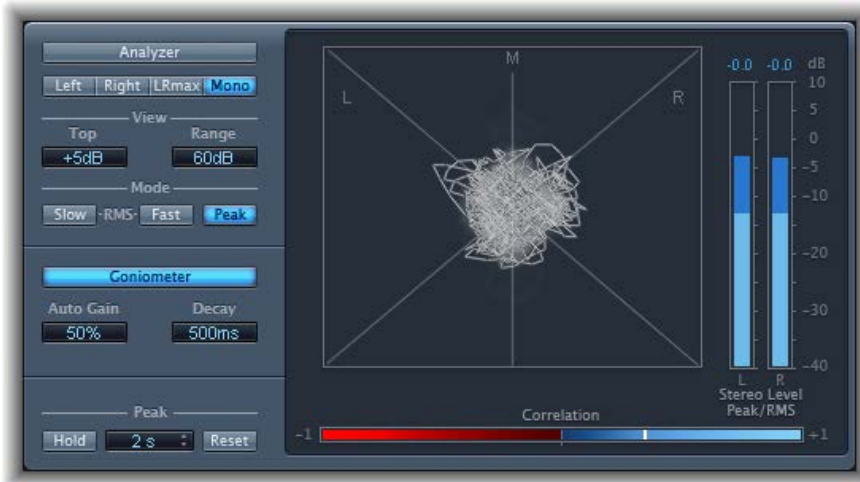
MultiMeter Goniometer parameters

- *Goniometer button:* Switches the main display to Goniometer mode.

- *Auto Gain field:* Drag to set the amount of display compensation for low input levels. You can set Auto Gain levels in 10% increments or set it to off.

   Note: To avoid confusion with the Auto Gain parameter found in other effects and processors (such as the compressors), Auto Gain is only used as a display parameter in the meters. It increases display levels to enhance readability. It does not change the actual audio levels.

- *Decay field:* Drag to determine the time it takes for the Goniometer trace to fade to black.

MultiMeter Level Meter

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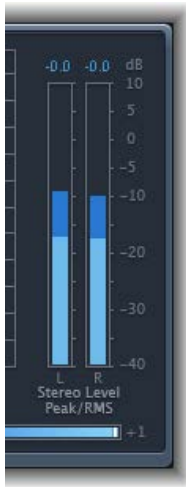
RMS and peak levels are shown simultaneously, with RMS levels appearing as dark blue bars and Peak levels appearing as light blue bars. When the level exceeds 0 dB, the portion of the bar above the 0 dB mark turns red.

Current peak values are displayed numerically (in dB increments) above the Level Meter. Click in the display to reset peak values.

MultiMeter Correlation Meter

The Correlation Meter gauges the phase relationship of a stereo signal. The Correlation Meter's scale values indicate the following:

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

MultiMeter Peak parameters

The MultiMeter Peak parameters are used to enable or disable the peak hold function and to reset the peak segments of all meter types. You can also determine a temporary peak hold duration.

MultiMeter Peak parameters

- *Hold button:* Click to turn on peak hold for all metering tools in the MultiMeter, as follows:
  - *Analyzer:* A small yellow segment above each 1/3-octave level bar indicates the most recent peak level.
  - *Goniometer:* All illuminated pixels are held during a peak hold.
  - *Correlation Meter:* The horizontal area around the white correlation indicator denotes phase correlation deviations in real time, in both directions. A vertical red line to the left of the correlation indicator shows the maximum negative phase deviation value. You can reset this line by clicking it during playback.
  - *Level Meter:* A small yellow segment above each stereo level bar indicates the most recent peak level.
- *Hold Time pop-up menu:* Choose the hold time for all metering tools. Choose 2, 4, or 6 seconds—or infinite.

  Note: The (peak) Hold button must be turned on for the selected time value to have an effect.

- *Reset button:* Click to reset the peak hold segments of all metering tools.
- *Inter Sample Peak Detection checkbox (Extended Parameters area):* Select to detect inter-sample peaks in the signal.

Use the Tuner utility

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

Tune Deviation display
Mute button
Drag to set pitch.
Keynote
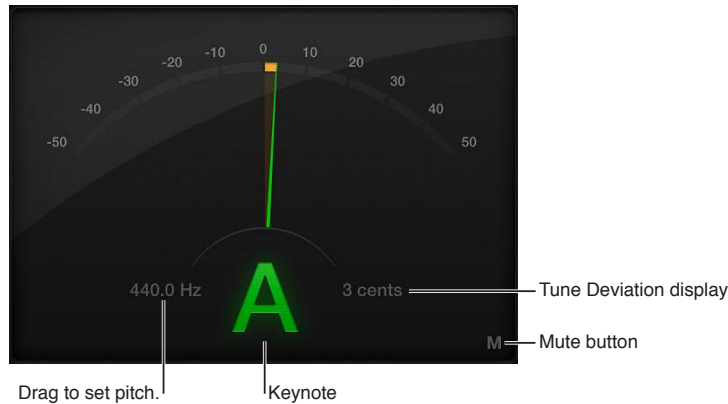
Tuner parameters

- *Graphic Tuning display:* Indicates the pitch of the note in cents. 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. Color is also used to indicate tuning accuracy, with green denoting a tuned signal, and orange indicating a detuned signal.

- *Reference Tuning field:* Drag vertically to set the pitch of the note used as the basis for tuning. The default is for note A at 440 Hz and can be set in a range from 410 to 470 in 0.1 Hz steps.

- *Keynote display:* Shows the target pitch of the note being played (the closest tuned pitch).

- *Tune Deviation display:* Shows the tuning deviation in cents.

- *Mute button:* Click to mute the active channel.

Use the Tuner

1 Insert the Tuner plug-in into an audio channel strip.

2 Play a single note on the instrument and watch the Graphic Tuning and Keynote displays. If the note is flat or sharp of the keynote, orange segments are shown in the Graphic Tuning display, the Keynote is shown in orange, and the Tune Deviation display indicates 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 and the Tune Deviation field shows zero (0 cents).

The Graphic Tuning display and Keynote are shown in green when correctly tuned.

MIDI plug-ins

9

Use MIDI plug-ins

MIDI plug-ins are inserted in software instrument channel strips and process or generate MIDI data—played from a MIDI region or a MIDI keyboard—in real time.

MIDI plug-ins are connected in series *before* the audio path of a software instrument channel strip.

MIDI plug-ins have a MIDI input, the MIDI processor, and a MIDI output. The output signals sent from MIDI plug-ins are standard MIDI events such as MIDI note or controller messages.

Insert one or more MIDI plug-ins

1 Click the space between the EQ graphic and the Instrument slot of a software instrument channel strip.

2 Choose the name of the MIDI plug-in you want to use from the MIDI plug-ins pop-up menu.

   The selected MIDI plug-in window opens, and a green label with the MIDI plug-in name is shown in the channel strip.

3 To insert additional MIDI plug-ins, move the pointer over the top or bottom edge of the inserted MIDI plug-in label in the channel strip, then click when you see a green line.

   Note: The plug-in window does not open automatically for MIDI plug-ins inserted in this way. Click the label to open the plug-in window.

Change the order of MIDI plug-ins

- Drag the label of the MIDI plug-in that you want to move over the label of the target MIDI plug-in.
  - *If the target MIDI plug-in is in the top slot:* It is moved down the list.
  - *If the target MIDI plug-in is in the bottom slot:* It is moved up the list.
  - *If the target MIDI plug-in is in a middle slot:* The positions of the two plug-ins are swapped.

Remove a MIDI plug-in

- Click the arrows near the right edge of the MIDI plug-in label that you want to remove, then choose No Plug-in from the pop-up menu.

Bypass a MIDI plug-in

1 Click the Bypass button near the left edge of the MIDI plug-in label.

   The plug-in label is dimmed. All parameter settings of the plug-in are retained when bypassed.

2 Click the Bypass button to restore the MIDI plug-in.

Arpeggiator MIDI plug-in

Arpeggiator overview

The Arpeggiator MIDI plug-in generates musically interesting arpeggios based on incoming MIDI notes. It provides split and remote features that allow you to control nearly all Arpeggiator functions without taking your hands off the keyboard, making it a powerful live performance tool.

An arpeggio is a succession of notes in a chord. Rather than all notes being played at one time, they are played one after the other in a pattern: up, down, random, and so on. The Arpeggiator plug-in provides a number of preset patterns, inclusive of switchable variations and inversions. Inversions change the root note of the chord from the lowest note, resulting in a different start note in arpeggiated patterns. These features let you quickly switch between patterns and feels when performing live, or when creating new projects in the studio.

The Arpeggiator MIDI plug-in is divided into three areas.
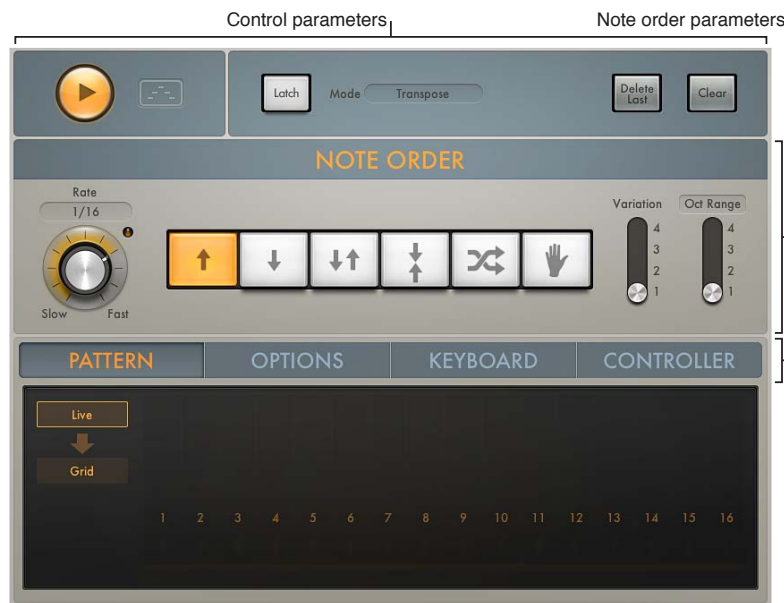
- *Control parameters:* This area contains the Play and Latch controls. See Arpeggiator control parameters.
- *Note order parameters:* The note order parameters determine the arpeggio type and include four variations or inversions, the arpeggio octave range, and the arpeggio speed. See Arpeggiator note order parameters overview.
- *Advanced parameters:* The advanced Arpeggiator controls are divided into four tabs. Click the Pattern, Options, Keyboard, or Controller tab to open each parameter group. See Arpeggiator pattern parameters overview, Arpeggiator options parameters, Arpeggiator keyboard parameters, and Assign Arpeggiator controller parameters.

Arpeggiator control parameters

The control parameters start and stop the Arpeggiator and determine the latching behavior. You can also capture a live arpeggio as a MIDI region.
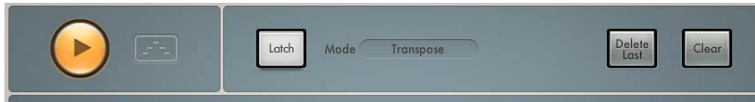
Control parameters

- *Play button:* Click to start or stop arpeggiated playback of note input from a MIDI keyboard or a MIDI region. The Play button is lit when in play mode. When the Arpeggiator plug-in is stopped, incoming MIDI notes are passed through, and the settings of the split and remote keyboard parameters are retained. See Arpeggiator keyboard parameters.
    - *When MainStage is in play mode:* The arpeggio starts playing whenever the Arpeggiator Play button is on, including when the plug-in is first inserted. Arpeggio playback is linked to the song position.
    - *When MainStage is stopped:* Arpeggio playback stops. Incoming MIDI notes are passed through, and keyboard split and remote settings are retained.

    Note: You can click the Arpeggiator Play button while MainStage is stopped to begin arpeggio playback from the first step in the arpeggio.

- *Capture live performance button:* Click the button, then drag the *playing* arpeggio to any software instrument track. The currently playing arpeggio pattern is placed as a MIDI region at the target position.

- *Latch button:* Click to turn Latch mode on or off. This enables an arpeggio to run without the need for you to hold down keys. Latch mode behavior is determined with the Latch mode pop-up menu.

- *Latch mode pop-up menu:* Choose a Latch mode:
    - *Reset:* The first key played clears the currently latched notes.
    - *Transpose:* Play a single key to transpose the arpeggio relative to the note value of the pressed key and the lowest arpeggiated note.

    Note: Pressing more than one key simultaneously clears currently latched notes and starts a new arpeggio.

    - *Gated Transpose:* This option is the same as Transpose Latch mode with the difference that the arpeggio only plays while a key is pressed. As soon as the key is released, the arpeggio is muted.
    - *Add:* Play keys—one by one, or simultaneously as a chord—to add them to the latched arpeggio. You can play the same key multiple times and the note repeats the number of times it is struck.
    - *Add Temporarily:* This option is the same as Add Latch mode except that played notes are added to the latched arpeggio only while held. When a temporarily added key is released, it is removed from the arpeggio.
    - *Through:* All incoming MIDI notes are passed through the Arpeggiator plug-in, enabling you to play along with a latched arpeggio.

- *Delete Last button:* Click to delete the last event—a note, rest, or tie—that was added to the arpeggio.

    Note: Each event is allocated a unique position identification number and the "last" event has the highest position identification number.

- *Clear button:* Click to remove all notes from the Arpeggiator plug-in latch memory. The arpeggio stops playing and all position identification numbers are reset to zero, enabling you to create a new arpeggio without turning off Latch mode, which can be useful in a live situation when preparing for a chord change.
- *Silent Capture checkbox (extended parameter):* Click the disclosure triangle at the lower left to display the extended parameters. Select the Silent Capture checkbox to capture an arpeggio step by step without being disturbed by the immediate response of the running arpeggiator.
  - When enabled, the arpeggiator is stopped and Latch/Add mode is engaged.
  - When disabled, Play is re-engaged (if previously active) and Latch mode switches to Transpose.

Arpeggiator note order parameters

Arpeggiator note order parameters overview

The note order parameters provide control of a pre-programmed order of notes that are played at a preset playback rate. Once all notes are played, the arpeggio cycles from the start. When you play a single key, it is repeated. When you play multiple keys, the held notes are played one after the other. As you play additional notes, these are seamlessly added to the arpeggio. When you release notes, they are removed from the arpeggio. The arpeggio stops when you release all played keys (unless Latch mode is turned on in the Arpeggiator control parameters).

The Arpeggiator plug-in automatically assigns position identification numbers to each note in the order they are played. These position identification numbers associate an event, such as a note, rest, or a tie, with a particular step. This lets you switch between note order presets while retaining a rest on the third step, for example.

Note order parameters

All note order parameters can be changed while an arpeggio is playing. Changes are immediate and are seamlessly applied to the running arpeggio.

- *Rate knob and field:* Rotate to set the arpeggiator rate. Choose from: 1/4, 1/8, 1/16 (including triplet and dotted notes), and 1/32. You can also click the field to choose a value from a pop-up menu. The LED indicates the rate and briefly changes color at the start of each new cycle.
- *Direction buttons:* Click a button to set the arpeggio direction.
  - *Up:* The arpeggio is played from the lowest note to the highest note.
  - *Down:* The arpeggio is played from the highest note to the lowest note.
  - *Up/Down:* The arpeggio plays up and down, from the lowest note; the highest and lowest notes repeat.
  - *Outside-in:* The arpeggio plays the highest then the lowest notes, then the second highest and second lowest, the third highest and third lowest, and so on.
  - *Random:* Arpeggiated notes play in a random order.
  - *As played:* All notes play in the order they were triggered.

- *Lock button:* Works in conjunction with the As played button. When you first click the As played button, an open lock symbol is shown. Click the open lock symbol once you have triggered an arpeggio to lock the current note order, indicated by a closed lock. This note order and feel is retained for any newly triggered arpeggios, but with new notes replacing the original notes. Click the lock symbol again to clear the locked note order and to revert to the standard "as played" behavior. The lock state and note order can be saved with a setting.
- *Variation switch:* Move to one of the four positions to determine the type of variation. See Arpeggiator note order variations for details.
- *Oct Range/Inversions button:* Click to switch between two modes: Octave Range or Inversions. The four-position Oct Range/Inversion switch below the buttons is used to determine the octave range or the chord inversion pattern.
- *Oct Range/Inversion switch:* Move to one of the four positions to determine the octave range or the chord inversion pattern. See Arpeggiator note order inversions for details on the four switch position behaviors in Inversions mode.

  In Octave Range mode:

  - *Position 1:* The arpeggio repeats without transposition.
  - *Position 2:* The lowest note is transposed by one octave. Once repeated, the arpeggio restarts in the original octave.
  - *Position 3:* The first repetition is transposed by one octave, and the second repetition is transposed by two octaves. Once the second repetition is played, the arpeggio restarts in the original octave.
  - *Position 4:* The first repetition is transposed by one octave, the second by two octaves, and the third by three octaves. Once the third repetition is played, the arpeggio restarts in the original octave.

  In Inversions mode:

  - *Position 1:* The arpeggio repeats without inverting the held notes.
  - *Position 2:* The arpeggio is inverted once during the first repetition. Once repeated, the arpeggio restarts.
  - *Position 3:* The arpeggio is inverted twice, once each during the first and the second repetition. Once the second repetition is played, the arpeggio restarts.
  - *Position 4:* The arpeggio is inverted three times, once each during the first, second, and third repetitions. Once the third repetition is played, the arpeggio restarts.

Arpeggiator note order variations

The table outlines the Arpeggiator behavior in each note order preset when the Variation switch is set to the four available positions.

Note order	Variation 1	Variation 2	Variation 3	Variation 4
Up	Plays from the lowest to highest note in consecutive order and restarts when all keys are played.	Plays the second step first. This variation consists of four steps; all pressed keys are divided into groups of four with the note order applied to all groups. If there are fewer than four notes, the steps without an assigned key are skipped. Once all keys are played, the arpeggio restarts with the lowest note.	Plays the third step first. This variation consists of four steps; all pressed keys are divided into groups of four with the note order applied to all groups. If there are fewer than four notes, the steps without an assigned key are skipped. Once all keys are played, the arpeggio restarts with the lowest note.	This variation, which consists of three steps, plays up and overlaps; all pressed keys are divided into groups of three with the note order applied to all groups. If there are fewer than three notes, the steps without an assigned key are skipped. Once all keys are played, the arpeggio restarts with the lowest note.
Down	Plays from the highest to lowest note in consecutive order and restarts when all keys are played.	Plays the second step first. This variation consists of four steps; all pressed keys are divided into groups of four with the note order applied to all groups. If there are fewer than four notes, the steps without an assigned key are skipped. Once all keys are played, the arpeggio restarts with the highest note.	Plays the third step first. This variation consists of four steps; all pressed keys are divided into groups of four with the note order applied to all groups. If there are fewer than four notes, the steps without an assigned key are skipped. Once all keys are played, the arpeggio restarts with the highest note.	This variation, which consists of three steps, plays down and overlaps; all pressed keys are divided into groups of three with the note order applied to all groups. If there are fewer than three notes, the steps without an assigned key are skipped. Once all keys are played, the arpeggio restarts with the highest note.
Up and down	Plays from the lowest to highest note in consecutive order, then plays from the highest to the lowest note, and restarts when all keys are played.	Plays from the lowest to highest note in consecutive order, then plays from the second highest to the second lowest note, and restarts when all keys are played.	This two-step variation works with pairs of notes. The second note of the pair plays first. In a four-note chord, the order is 2, 1, 4, 3. Once the pattern is played, the note order is reversed, then the arpeggio restarts.	This three-step variation works with trios of notes. The note order is 1, 3, 2. Once the pattern is played, the note order is reversed, then the arpeggio restarts.

Note order	Variation 1	Variation 2	Variation 3	Variation 4
Outside-in	Plays the highest note, then the lowest note, then plays the second highest and the second lowest note, and so on. The arpeggio restarts when all keys are played.	Plays the lowest note, then the highest note, then plays the second lowest and the second highest note, and so on. The arpeggio restarts when all keys are played.	This is an inside-out variation. The number of played keys is divided by two (rounded up to the nearest whole number). The highest center note is played, then the low-center note, and so on. In a six-note chord, the order is 4, 3, 5, 2, 6, 1. The arpeggio restarts when all keys are played.	This is an inside-out variation. The number of played keys is divided by two (rounded up to the nearest whole number). The lowest center note is played, then the high-center note, and so on. In a six-note chord, the order is 3, 4, 2, 5, 1, 6. The arpeggio restarts when all keys are played.
Random	Played note order is randomly generated and can include duplicate notes.	Played note order is randomly generated but no note is played twice. The arpeggio restarts when all keys are played.	This variation favors low notes. Played note order is randomly generated and can include duplicate notes.	This variation favors high notes. Played note order is randomly generated and can include duplicate notes.
As played	Plays all notes in the order they were played, then restarts.	Plays all notes in the reverse order they were played, then restarts.	Plays all notes in the order they were played, then plays notes in reverse order, doubling the first and last played notes. The arpeggio restarts once all notes are played.	Plays all notes in the order they were played, then plays notes in reverse order, but does not repeat the first and last played notes. The arpeggio restarts once all notes are played.

Arpeggiator note order inversions

The table outlines the Arpeggiator behavior in each note order preset when the Oct Range/Inversion switch is set to the four positions in Inversions mode (set with the Oct Range/Inversions button). Inversions change the root note of the chord, resulting in a different start note to arpeggiated patterns.

Note order	Inversion 1	Inversion 2	Inversion 3	Inversion 4
Up	Plays the original chord, then three inversions in consecutive order and restarts. Playback order: original, 1, 2, 3.	Plays the second inversion first. Playback order: 1, original, 2, 3.	Plays the third inversion first. Playback order: 2, original, 1, 3.	This variation, which consists of three steps, plays up, and overlaps. Playback order: original, 2, 1, 3.
Down	Plays the original chord, then three inversions in consecutive order and restarts. Playback order: 3, 2, 1, original.	Plays the second step first. Playback order: 2, 3, 1, original.	Plays the third step first. Playback order: 1, 3, 2, original.	This variation, which consists of three steps, plays down, and overlaps. Playback order: 3, 1, 2, original.
Up and down	Plays the original chord, then three inversions in consecutive order, then reverses the order, repeating the first and last. Playback order: original, 1, 2, 3, 3, 2, 1, original.	Plays the second step first. Playback order: 1, original, 3, 2, 2, 3, original, 1.	Once the pattern is played, the order is reversed, then the arpeggio restarts. Playback order: original, 2, 1, 3, 3, 1, 2, original.	Once the pattern is played, the order is reversed, but the third inversion is not repeated. Playback order: original, 1, 2, 3, 2, 1.
Outside-in	Plays the highest inversion, then the original, then plays the second highest and the second lowest inversion, and so on. Playback order: 3, original, 2, 1.	Plays the original, then the highest inversion, then plays the second lowest and the second highest inversion, and so on. Playback order: original, 3, 1, 2.	This is an inside-out variation. Playback order: 1, 2, original, 3.	This is an inside-out variation. Playback order: 2, 1, 3, original.
Random	Played inversion order is randomly generated and can include duplicate chord inversions.	Played inversion order is randomly generated but no chord inversion is played twice.	This variation favors low chord inversions. Played inversion order is randomly generated and can include duplicate chord inversions.	This variation favors high chord inversions. Played inversion order is randomly generated and can include duplicate chord inversions.

Arpeggiator pattern parameters

Arpeggiator pattern parameters overview
Click the Pattern tab to open the Arpeggiator pattern parameters.

The Pattern tab includes two distinct functional modes: Live and Grid. The modes are mutually exclusive, so turning on one turns off the other. It also provides a unique Live Capture to Grid facility.

When Grid mode is active, it controls the arpeggio's velocity, cycle length, rests, ties, and chords. All live input of available grid parameters, such as velocities, is ignored.

When you switch to Live mode, the arpeggio performance is controlled live by your input. For example, the velocities of arpeggiated notes are determined by the way you played them. Any existing grid values are retained but are disabled until you return to Grid mode.

Note: When you capture a live performance, grid values are not retained.

Pattern parameters
- *Live button:* Click to turn on Live mode. See Arpeggiator Live mode.
- *Arrow button (Live mode only):* Click to capture the currently playing velocities, rests, ties, and chords. Grid mode is automatically turned on and the captured performance can be edited in the grid. See Arpeggiator Grid mode.
- *Grid button:* Click to turn on Grid mode.

Arpeggiator Live mode
In Live mode, you can add rests, ties, and chords in real time by clicking the onscreen buttons or by using equivalent MIDI keyboard remote keys. See Arpeggiator keyboard parameters.

The grid acts as a display only. Incoming MIDI velocities, rests, ties, and chords are displayed in real time but cannot be edited in the grid. To edit individual arpeggiator steps, click the Grid button to turn on Arpeggiator Grid mode.

Note: Rests, ties, and chords are active only when Latch mode is turned on. See Arpeggiator control parameters.

Live mode parameters

- *Rest button:* Click to insert a rest at the current arpeggiator step position. A position identification number is assigned to the rest, ensuring that its rhythmic position (step number) within the arpeggio is retained, even when different note order presets are chosen.

  Note: Rests can only be added while building the arpeggio, which means that at least one key must be held if you want to add a rest. Once all keys are released, the Arpeggiator acts in accordance with the rules of the set Latch mode and expects to receive a MIDI note for transposition and so on. In Latch Add mode, this restriction does not apply because it allows you to add MIDI notes, rests, ties, and chords after all keys are released.

- *Tie button:* Click to insert a tie at the current arpeggiator step position. A position identification number is assigned to the tie, ensuring that its rhythmic position (step number) within the arpeggio is retained, even when different note order presets are chosen.

- *Chord button:* Click to insert a chord at the current arpeggiator step position. When the arpeggiator encounters a chord step, it simultaneously plays all notes, including their unique velocities, currently in memory (latched or held). A position identification number is assigned to the chord, ensuring that its rhythmic position (step number) within the arpeggio is retained, even when different note order presets are chosen.

Arpeggiator Grid mode

The grid consists of 16 steps. Each step controls the step velocity as well as its rest, tie, and chord status. In addition, a cycle length can be set. The current grid pattern is automatically saved with the concert. You can also save and load your own or factory grid patterns. The Arpeggiator plug-in assigns incremental position identification numbers to each note in the order they were received, regardless of the initially selected note order preset. These position identification numbers are used to "lock" an event, such as a note, a rest, or a tie to a particular step.

Grid mode parameters

- *Step on/off buttons:* Click to turn each of the 16 available steps on or off.
  - *If a step is turned on:* An arpeggiator note is played at the respective grid position.
  - *If a step is turned off:* The grid position is silent and is perceived as a rest.

  Note: To ensure the integrity of the arpeggio, the note that would have been played—if the step had been active—is moved to the next active grid position.

- *Velocity bars:* Drag the velocity bar vertically to set the velocity for each active step. Where multiple velocity bars exist, click *above* them to draw in the velocity of several steps.

  Drag the velocity bar toward the right to overlap the next step, thus tying it to that step. If the step to the right is a rest (an inactive step) this step is automatically turned on to create the tie. A step can be tied to multiple steps in a row. The original velocity values of tied steps are replaced by the velocity of the first step they are tied to, indicated graphically by the velocity bar extending over all tied steps.

Note: Within an arpeggio, ties are perceived as a rhythmic element rather than a melodic variation. As a consequence, the tied note may change if notes are added after the tie has been entered or when you choose a different note order preset.

- *Chord on/off buttons:* Click the chord symbol to turn on Chord mode for the respective step. When the Arpeggiator encounters a chord step, it simultaneously plays all notes currently in (latched or held) memory on that step. If a chord step is tied to a non-chord step, Chord mode is automatically activated for that step. If a non-chord step is tied to a chord step, Chord mode is automatically turned off for that step. Moving the velocity bar on a chord step changes the overall level of the chord, while retaining relative velocity differences between notes in the chord.

- *Cycle length bar:* Drag the cycle handle at the end of the cycle length bar to change the grid length. The currently playing step is indicated by a light running inside the Cycle length bar.

  Note: The grid length set with the Cycle length bar is independent of the Arpeggio cycle length parameter (which sets the length of the arpeggiated note pattern) described in Arpeggiator options parameters. The grid length cycles independently of the effective note pattern, ensuring that the perceived rhythmic pattern created by the grid is not disrupted by a changing arpeggio length.

- *Pattern pop-up menu:* Choose one of the following items to save or load user grid patterns or to load a factory grid pattern.

  - *Save Pattern as:* Opens a name field. Enter a name, then click the Save button to save your pattern. Click Cancel to exit the "Save Pattern as" name field.

  User patterns are shown in the Pattern pop-up menu.

  Note: Factory grid patterns cannot be overwritten. If you attempt to do so, a Save Pattern as name field appears.

  - *Recall Default:* Deletes all current data and reverts to a "from scratch" state.

  - *Delete User Pattern:* Deletes the current user pattern.

  - *Custom:* This menu item is shown automatically when any pattern changes have been made. It can be considered the "current state" pattern preset.

Arpeggiator options parameters

Click the Options tab to set global Arpeggiator playback parameters, such as note length and velocity.

Options parameters

- *Note Length knob:* Rotate to define the length of the arpeggiated notes. This ranges from 1 to 150%.
- *Random knob:* Rotate to set the amount of random note length variation.
- *Velocity knob:* Rotate to determine the maximum range of possible velocity values for arpeggiated notes. At the far right position (100%), the original velocities of recorded or played notes are retained. At the far left position (0%), the original velocities of recorded or played notes are ignored and all notes are output at a constant velocity.
- *Vel (Velocity Base) field:* Drag vertically to set a minimum velocity value that is used for random velocity modulations and crescendos.
- *Crescendo/Random button:* Click to switch between two modes: Random and Crescendo. The amount of variation is controlled with the Crescendo/Random knob. The range of the crescendo or possible random velocities is set with the Velocity parameters.
- *Crescendo/Random knob:* Rotate to set the strength of the crescendo when the Crescendo/Random button is set to Crescendo. Rotate to set the amount of random velocity variation when the Crescendo/Random button is set to Random.
  - *When set to Crescendo:* The set amount is added to, or subtracted from, the velocity of all notes on each arpeggio repetition, starting with the second cycle.
  - *When set to Random:* The velocity values of all notes are randomized symmetrically by the set amount. At a value of 0% there is no randomization applied. At a value of 100% the velocity values are completely randomized.
- *Swing knob and field:* Rotate to set the strength of note swing. The Swing parameter moves every second note in the arpeggio closer to the nearest downbeat. A value of 0% results in no note movement, whereas a value of 100% results in extreme note movement.
- *Cycle Length knob:* Rotate to set a length for the arpeggio. You can choose from the following:
  - *By Grid:* Matches the arpeggio length to the global division setting. This is useful for rhythmically synchronizing the arpeggio length with other regions.
  - *1 to 32:* Set the arpeggio length to the specified number of notes.
  - *As played:* The arpeggio length is determined by the length of played notes.

Arpeggiator keyboard parameters

Click the Keyboard tab to open the Arpeggiator keyboard parameters. The dots shown on the keyboard represent the output of currently playing notes, including any key and scale adjustments. You can also open the Remote Key editor window from the Keyboard tab. For further details, see Use Arpeggiator keyboard parameters.
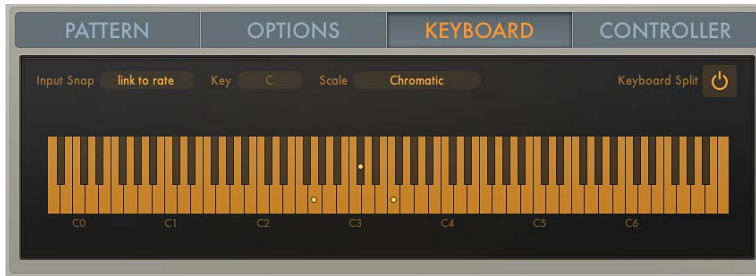
Keyboard parameters

- *Input Snap pop-up menu:* Choose a beat value that is used to "snap" the first incoming note to a position, thus quantizing the start (and playback) of the arpeggio.

  The default Input snap pop-up menu value is "link to rate," which matches the set Arpeggiator Rate (see Arpeggiator note order parameters overview).

- *Key pop-up menu:* Choose a root key for the chosen scale. C is the default key.

- *Scale pop-up menu:* Choose a scale. Played keys are snapped to the nearest note in the chosen scale. Choose from: Off/Chromatic (default), Major, Major Pentatonic, Major Blues, Lydian, Mixolydian, Klezmer, Minor Pentatonic, Minor Blues, Japanese, Minor, Harmonic Minor, Dorian, Phrygian, Lochrian, and South-East Asian.

- *Keyboard Split button:* Click to divide your MIDI keyboard range into three zones.

- *Remote (Key editor) button:* You must first click the Keyboard Split button to make the Remote (Key editor) button visible. The Remote button opens the Remote Key editor window where you can assign a range of MIDI keys to Arpeggiator functions.

Use Arpeggiator keyboard parameters

The Arpeggiator keyboard parameters let you split your keyboard into zones that are used for standard note playback, arpeggio note triggering, and remote control of the Arpeggiator plug-in parameters.

Resize the keyboard display

The default keyboard range spans the 88 notes from C0 to C7.

- Click the keyboard, then drag left or right to reveal additional octaves, in one octave increments.

Set up a keyboard split

By default, the complete MIDI key range (0–127) is used solely for arpeggiating. You can split your MIDI keyboard into several zones to control Arpeggiator plug-in functions with MIDI keys. The onscreen keyboard reflects the split layout.
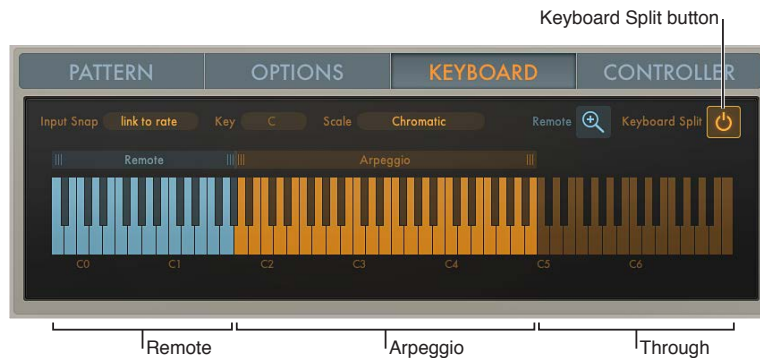
- To divide your MIDI keyboard range into three zones, click the Keyboard Split button.
  - *Arpeggio:* Notes played in this keyboard zone are arpeggiated.
  - *Remote:* Notes played in this zone trigger an Arpeggiator function.
  - *Through:* Notes are passed through the Arpeggiator plug-in unprocessed. This zone covers all keys outside the two zones listed above.

Resize the Arpeggio or Remote zone

- Drag the handles above each end of the Arpeggio or Remote zone to resize it.

Move the Arpeggio or Remote zone

- Click the name of either the Arpeggio or Remote zone and drag to the left or right. If the zones are adjacent, both zones are moved.

The Arpeggio and Remote zones can swap places so that the Remote zone sits above the Arpeggio zone or vice versa, but the two zones cannot overlap.

Remote control the Arpeggiator with a MIDI keyboard
Most Arpeggiator parameters can be remote controlled using a MIDI keyboard. By default,
only a few Remote commands are available. You can resize the Remote zone to make more
commands available.
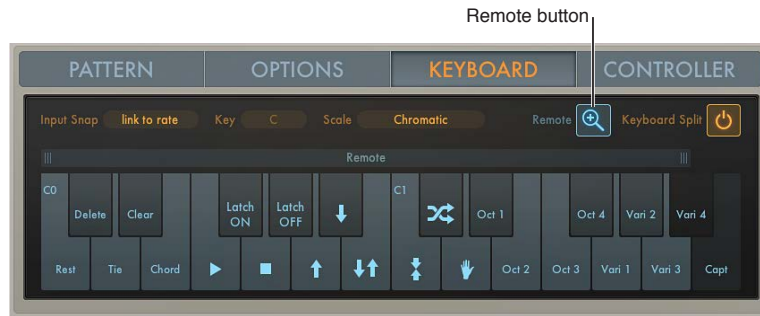
1   You must first click the Keyboard Split button to display the Remote (Key editor) button.

    The type and number of available remote keys is determined by the Remote zone range. Keys
    outside this range are dimmed and assigned functions cannot be remote controlled with a MIDI
    keyboard.

2   Click the Remote button to open the Remote Key editor window.

    A zoomed-in keyboard is displayed, with each key labeled according to its assigned function.
    Click the Remote button a second time to close the Remote Key editor window. You can also
    click the Keyboard Split button to exit the Remote Key editor window.

3   Drag the left or right border of the range bar above the Remote Key editor keyboard to resize
    the Remote zone.

Assign Arpeggiator controller parameters
Click the Controller tab to assign up to four MIDI controllers to Arpeggiator plug-in parameters.

Assign a MIDI controller to an Arpeggiator parameter

1   Choose a MIDI controller from any of the MIDI Controller pop-up menus.

2   Choose a parameter from any of the Destination pop-up menus. Choices are: Note Length, Note
    Length Random, Velocity Range, Velocity Base, and (De-)Crescendo.

Learn a MIDI controller

1   Choose a parameter from any of the Destination pop-up menus.

2   Choose Learn from any of the MIDI Controller pop-up menus.

3   Move a controller on your MIDI keyboard to assign it to the Destination parameter.

    The Learn feature has a 20-second time out facility. If you do not move a controller on your MIDI
    device within 20 seconds, Learn mode is automatically disabled.

Chord Trigger MIDI plug-in

Chord Trigger overview

The Chord Trigger MIDI plug-in lets you trigger chords by playing a single MIDI key. The onscreen keyboards have two functions: the display of incoming and outgoing MIDI notes and the assignment of chords to keys. See Use Chord Trigger.
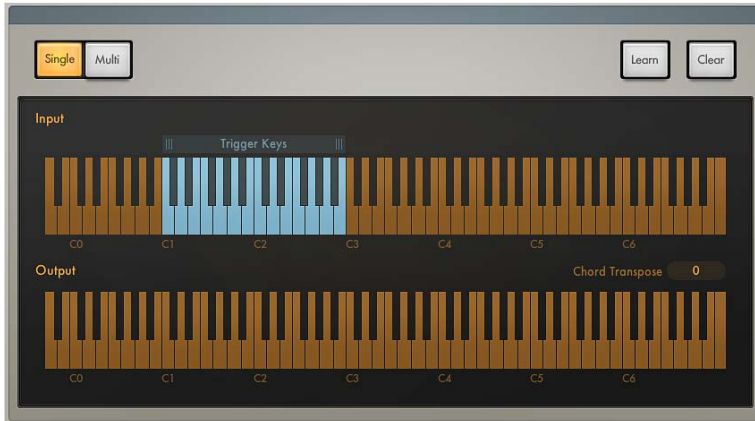
Chord Trigger parameters

- *Single and Multi buttons:* Click either the Single or Multi button to select a mode.
  - *Single Chord mode:* This mode lets you assign a single chord to a trigger key. When playing up and down the keyboard, the memorized chord is transposed in relation to the trigger key. This mimics the Chord Memo or Note Stack feature found on many classic synthesizers.
  - *Multi Chord mode:* This mode lets you assign a different chord to each key on the keyboard.
- *Upper keyboard:* Shows incoming MIDI notes, represented by orange dots and the blue shaded chord trigger range. Drag the handles above the keyboard to set the chord trigger range. Notes that fall within this range are processed. Notes outside the range are not processed. You can also click notes in the chord trigger range to trigger chords.
- *Lower keyboard:* Shows the resulting MIDI output—the chords triggered by incoming MIDI notes. The notes of each chord (in the chord trigger range) are displayed as blue dots. Chords outside the chord trigger range are displayed as orange dots.
- *Learn button:* Click to turn on Learn mode. See Use Chord Trigger for details on using Learn mode.
- *Clear button:* Click to erase a Trigger Key note and the corresponding chord. See Use Chord Trigger for details.
- *Chord Octave pop-up menu:* Choose an octave transposition value for chord playback.

Use Chord Trigger

Chord Trigger is straightforward to use: choose a mode, set a chord trigger range, select a trigger key, then set up a chord. You can also transpose chords and quickly assign multiple chords— onscreen or with your MIDI keyboard.

Define the chord trigger range

The shaded chord trigger range is shown on the upper keyboard. Incoming MIDI notes that fall within this range are interpreted as trigger keys that play the chord (Single Chord mode) or the chords (Multi Chord mode) assigned to them. Chords can be assigned to keys within the chord trigger range. Incoming MIDI notes that fall outside the defined chord trigger range are passed through Chord Trigger unaffected. This allows you to play a melody with the right hand while triggering/transposing memorized chords with the left, for example.

Drag here to define a keyboard range.

Drag here to define a keyboard range.
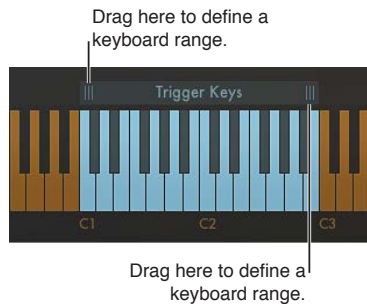
- Drag the handles of the chord trigger range bar above the upper keyboard to define a keyboard range.
  - *In Single Chord mode:* Playing a MIDI note (or clicking the upper keyboard) within the defined chord trigger range plays and transposes a single memorized chord. The transposition is performed in relation to the trigger key the chord is assigned to. For example, if a chord is assigned to C2, playing a D2 transposes the chord upward by two semitones. Playing a B1 transposes the chord down by a semitone.
  - *In Multi Chord mode:* Playing a MIDI note (or clicking the upper keyboard) within the defined chord trigger range triggers the chord that is memorized for the played key. Keys that do not have a chord assigned to them are silent when played.

Note: If the chord trigger range is made shorter, memorized chords that fall outside the range become inaccessible but are not deleted. Lengthening the chord trigger range makes assigned chords accessible again.

Transpose chords in the chord trigger range (Multi Chord mode only)

You may want to transpose triggered chords in some circumstances. For example, in Multi Chord mode you can move the entire chord trigger range upward by two semitones to change a chord progression in C-Major (starting with the C trigger key) into a progression that plays in D-Major, starting with the D trigger key.

Drag left or right to transpose.
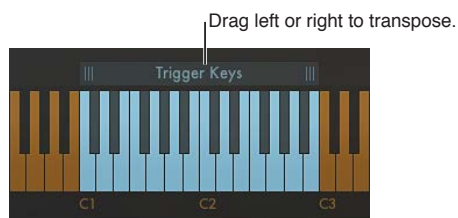
- Drag the center of the chord trigger range left or right.

All memorized chords are moved with the chord trigger range and are automatically transposed.

Transpose chords by octaves

- Choose an octave transposition from the Chord Octave pop-up menu.

All memorized chords can be transposed up or down by up to four octaves.

Assign a chord to a key using the onscreen keyboard

1 Click the Learn button.

The Learn button label changes to "Trigger Key" and the button begins to blink.

2 Click a trigger key—within the chord trigger range—on the upper keyboard.

The trigger key is set up for chord assignment. The Learn (Trigger Key) button label changes to "Chord."

3 Click the note or notes you want to assign to the trigger key on the lower keyboard.

As you click each note, you will hear it and any previously assigned notes in the chord.

Click assigned notes a second time to unassign or remove them from the chord.

4 To end chord assignment, click the Learn button.

You can repeat these steps to assign a different chord to each key in the chord trigger range when in Multi Chord mode. In Single Chord mode, only one chord can be learned.

Assign a chord to a key using a MIDI keyboard
It can be faster to use your MIDI keyboard when assigning chords to trigger keys. The Learn process can be started and stopped by playing an assigned note on your MIDI keyboard.

1 Click the disclosure triangle at the lower left to open the extended parameters.

2 Choose the MIDI note number you want to use as a remote control for the Learn button from the Learn Remote pop-up menu.

Choose Off if you no longer want to use a MIDI note as the Learn button remote control.

3 Play the note selected as the Learn button remote control on your MIDI keyboard.

The Learn button label changes to "Trigger Key" and the button begins to blink.

4 Play a trigger key—within the chord trigger range—on your MIDI keyboard.

This enables the trigger key for chord assignment. The Learn (Trigger Key) button label changes to "Chord."

5 Play the note or notes you want to assign to the trigger key on your MIDI keyboard.

As you play each note, you will hear it and any previously assigned notes in the chord.

Play assigned notes a second time to unassign or remove them from the chord.

6 Play the note selected as the Learn button remote control on your MIDI keyboard to end chord assignment.

You can repeat steps 3–6 to assign a different chord to each key in the chord trigger range when in Multi Chord mode. In Single Chord mode, only one chord can be learned.

Clear a chord assignment

1   Click the Clear button.
   - *In Single Chord mode:* The assigned chord is erased.
   - *In Multi Chord mode:* The button label changes to "Trigger Key" and begins to blink.

2   Click the trigger key that you want to clear on the upper keyboard.

   The chord assigned to the trigger key is erased and the trigger key is dimmed, indicating that no chord is assigned.

Clear all chord assignments
The following applies only to Multi Chord mode.

- Press Option, then click the Clear button.

   All chords on all trigger keys are erased.

Modifier MIDI plug-in

The Modifier MIDI plug-in lets you quickly reassign or filter a single continuous controller (CC) event type. You can also scale or add to event values.

Modifier parameters
- *Input Thru button:* Turn on to define whether the input event is sent to the output *in addition to* the reassignment.
- *Input Event pop-up menu:* Choose the type of MIDI input event that you want to reassign or filter.
- *Re-Assign To pop-up menu:* Choose the type of MIDI output event. If set to Off, the event type chosen from the Input pop-up menu is filtered.
- *Scale slider:* Drag to set the scaling amount for the output event type chosen from the Re-assign To pop-up menu.
- *Add slider:* Drag to set the offset amount for the output event type chosen from the Re-assign To pop-up menu.

Modulator MIDI plug-in

Modulator MIDI plug-in overview

The Modulator MIDI plug-in can generate continuous controller, aftertouch, and pitch bend messages. It consists of one syncable LFO and one Delay/Attack/Hold/Release envelope. See Modulator MIDI plug-in LFO and Modulator MIDI plug-in envelope.

Both the LFO and envelope can be assigned to output any continuous controller, aftertouch, and pitch bend message. You can also specify a step width for the continuous outputs of the LFO and envelope, resulting in modulations that are reminiscent of classic Sample & Hold circuits.

Modulator MIDI plug-in LFO

Modulation LFO parameters

- *LFO on/off button:* Turns the LFO on or off.

- *Waveform Shape buttons:* Click to select a waveform shape. Choose from: triangle, sine, square, and random. Each is suited for different types of modulations.

- *Waveform display:* Shows the LFO waveform shape.

- *Symmetry slider:* Drag to adjust the symmetry of the waveform. This deforms the waveform in the following ways:

  - *Triangle:* Shapes the triangle waveform into either an upward-sawtooth or downward-sawtooth waveform. A symmetry value of 0 results in a perfect triangle waveform.

  - *Sine:* Compresses the sine wave into one narrow and one wide duty cycle. A symmetry value of 0 results in a perfect sine waveform.

  - *Square:* Symmetry acts as a pulse width control. A symmetry value of 0 results in a perfect square waveform.

  - *Random:* Symmetry determines the maximum deviation between two consecutive random values. Low symmetry settings result in random values that are minimally different from one to the next, whereas high symmetry settings result in random values that deviate significantly.

- *Trigger switch:* Choose a switch position to determine how the LFO reacts to incoming MIDI note on messages.

  - *Free:* The LFO ignores MIDI note on messages.

  - *Single:* After all notes have been released, the LFO is reset by the first MIDI note on message it receives.

  Note: This means that legato playing does not reset the LFO, so keep this in mind during performances.

  - *Multi:* The LFO is reset by each received MIDI note on message.

- *Steps per LFO Cycle (Smoothing) slider and field:* Drag to determine the number of steps per LFO cycle.

  By default, the LFO produces a smoothed continuous stream of controller events, but you can use this parameter to create a stepped controller signal that is similar to the output of a Sample and Hold circuit. When you set a manual step rate, the LFO rate can be changed without altering the number of steps.

  Note: If the square or random waveform is selected, the Steps per LFO Cycle slider is renamed to Smoothing. The slider smooths the normally steep slopes of the square and random waveforms.

- *Rate knob:* Rotate to set the cycle speed of the LFO in hertz or in beat values when the Sync button is turned on. The LFO rate can be modulated by the envelope. See Modulator MIDI plug-in envelope.

- *Sync button:* Turn on to synchronize the LFO rate to the song tempo.

- *To pop-up menu:* Choose a continuous controller number, aftertouch, or pitch bend as the LFO output target.

- *Output Level slider:* Move to scale the LFO output level.

- *Oscilloscope:* The Oscilloscope to the left of the Output Level slider displays the shape of the LFO control signal before it is scaled.

- *MIDI Channel pop-up menu (extended parameter):* Click the disclosure triangle at the lower left. Choose a MIDI output channel.

Modulator MIDI plug-in envelope

Modulation Envelope parameters

- *Envelope on/off button:* Turns the envelope on or off.
- *Envelope display:* Shows the current envelope shape. Drag the handles in the display to set the following parameters:
  - *Delay:* Delays the onset of the envelope. Ranges from 0 to 10 seconds.
  - *Attack:* Sets the time required to reach the sustain level. Ranges from 0 to 10 seconds.
  - *Hold:* Sets the sustain level and duration. Ranges from 0 to 10 seconds.
  - *Release:* Sets the time required for the envelope to fall to a value of zero after the sustain phase of the envelope has finished. Ranges from 0 to 10 seconds.
- *Trigger switch:* Choose a switch position to determine if the envelope is triggered by the LFO or by incoming MIDI note on messages.
  - *LFO:* The envelope is retriggered when the LFO reaches its (positive) peak value. See Modulator MIDI plug-in LFO.

  Note: The envelope ignores incoming LFO triggers if it is currently completing an envelope pass.

  - *Single:* After all notes have been released, the envelope is re-triggered by the first MIDI note on message it receives.

  Note: This means that legato playing does not reset the envelope, so keep this in mind during performances.

  - *Multi:* The envelope is re-triggered by each received MIDI note on message.
- *Steps per Env(elope) Pass slider and field:* Drag to determine the number of steps per envelope pass. By default, the envelope produces a smoothed continuous stream of controller events, but you can use this parameter to create a stepped controller signal that is similar to the output of a Sample and Hold circuit. When you set a manual step rate, the envelope time can be changed without altering the number of steps.
- *Env to LFO Rate knob:* Rotate to set the maximum amount of LFO modulation (LFO depth). The LFO rate can be modulated by the Attack, Hold, and Release parameters (see above).

- *Env to LFO Amp knob:* Rotate to set the maximum amount of LFO output modulation. This enables you to fade the LFO in or out with the envelope.

- *To pop-up menu:* Choose a continuous controller number, aftertouch, or pitch bend as the envelope output target.

- *Output Level slider:* Move to scale the envelope output level.

- *Oscilloscope:* The Oscilloscope to the left of the Output Level slider displays the shape of the envelope control signal before it is scaled.

- *MIDI Channel pop-up menu (extended parameter):* Click the disclosure triangle at the lower left. Choose a MIDI output channel.

Note Repeater MIDI plug-in

This plug-in mimics an audio delay by generating repeating MIDI notes.

Note Repeater parameters

- *Input Thru button:* Turn on to pass incoming MIDI note events to the output *in addition to* the delayed note events. Turn off to send only the delayed notes to the output.
- *Delay Sync button:* Turn on to synchronize the plug-in with the host application tempo. Set the delay time with the Delay slider.
- *Delay slider and field:* Drag to set the delay time in milliseconds or in bar/beat values when the Delay Sync button is turned on.

  Note: When the Delay Sync button is turned on, only bar and beat values are available.

- *Display:* Shows the unprocessed input MIDI note (bright bar) and delayed MIDI notes. The height of the bars represents the velocity of each delayed MIDI note.
- *Repeats knob:* Rotate to set the number of delay repetitions.
- *Transpose knob:* Rotate to transpose each delay repeat by the set amount.
- *Velocity Ramp knob:* Rotate to scale the velocity level of each delay repeat by the set amount.
- *Note Range Min and Max sliders (extended parameter):* Click the disclosure triangle at the lower left to open the extended parameters. Move the Note Range Min and Note Range Max sliders to set an input note range. Notes that fall within this range are processed (default range: 1–127). Notes outside the range are not processed.

  Note: You can position the Note Range Min slider above the Note Range Max slider and vice versa, which inverts the input note range behavior: note events that fall within the range are *not* processed and note events outside the range are processed.

Randomizer MIDI plug-in

The Randomizer plug-in randomizes incoming MIDI events in real time.

Randomizer parameters

- *Event Type pop-up menu:* Choose the MIDI event type that you want to randomize.

- *Input Range sliders:* Drag to set the upper and lower limit of the range of values that are affected. Only parameter values that fall within the range are processed. All values outside the range pass through the plug-in.

  Note: You can position the lower Input Range slider above the upper Input Range slider and vice versa, which inverts the input range behavior: events that fall within the range are *not* processed and events outside the range are randomized.

- *Amount slider:* Drag to set the intensity of randomization. The colored box shows the range of possible output values in comparison with the unprocessed input signal shown in the middle.

- *Weight slider:* Drag to increase or decrease the likelihood that an event will be randomized within the set Amount range. The colored box reflects the weight setting: a darker gradient means less chance and a brighter color means more chance to produce values in the respective area.

  - Drag toward the left (Low) to increase the chance of low values being randomized.

  - Drag toward the right (High) to increase the chance of high values being randomized.

  - In the centered position, neither low or high values are favored, resulting in the entire range of values being randomly altered.

- *Output Offset slider:* Drag to offset the (random) MIDI output of the plug-in. Offsets can be negative or positive.

- *Seed slider (extended parameter):* Specifies a starting point (or seed) for randomization.

  An example is when using the Randomizer plug-in to randomize a piano melody. If you bounce the piano part, your randomized melody is saved as an audio file. If you bounce the song again, with Seed set to Random, the two bounces sound different. If Seed is set to the same specific value for both bounces, they are identical.

Scripter plug-in

Use the Scripter plug-in

The Scripter plug-in lets you load and use factory or user-created scripts to process or generate MIDI data in real time. You do not need any programming knowledge to use the plug-ins created in this environment, but you can view and modify them with the built-in script editor. Once authored and stored as a setting or patch or as part of a concert or project file, you can use the plug-in just like any other. A number of pre-built Scripter processors are included.

If you are an advanced user, you can create your own custom MIDI plug-ins. See Use the Script Editor.

The Scripter plug-in has one global parameter. Further parameters, defined by the JavaScript script that is currently running, are shown below the global parameter.
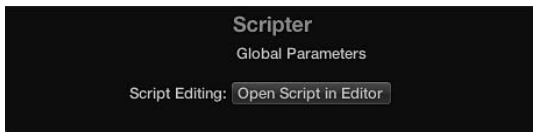
- *Open Script In Editor button:* Click to open the Script Editor window.

    You can write your own scripts or can paste scripts from other sources into this window.

Load a pre-built Scripter processor
Do one of the following:

- Load a setting from the plug-in window header.
- Load a patch from the Library.
- Load a channel strip setting from the channel strip Settings pop-up menu or Library.
- Load a project or concert that contains a Scripter plug-in with a running script.

    You do not need to explicitly save an active script as a setting, patch, and so on. Saving the project or concert retains the script and status of all Scripter plug-ins.

Use the Script Editor

The Script Editor is used to edit JavaScript code, enabling you to write your own MIDI plug-ins. Plug-in creation is in real time, which means that you can change and test your plug-in functions immediately. You can define interface elements, such as sliders and menus, that are shown in the Scripter plug-in window and can create the underlying logic and functions addressed by these onscreen controls. Some examples of utilities you can create with the Scripter plug-in are a chord strummer, a legato processor for stringed instruments, a harp glissando generator, or an algorithmic composer.

Tip: View the supplied scripts in the Script Editor to see how they are constructed. You can modify and re-use the code to change functions, or to create new processors.

See the Scripter API overview for Scripter API documentation and code examples.

Run Script button — Code Editor — Interactive Console

Script Editor parameters

- *Run Script button:* Click to evaluate the script and configure the plug-in and parameters. Output, including errors, is shown in the Interactive Console when you click this button.

- *Code Editor:* Type JavaScript code in this area. The editor provides the following features:
  - Syntax highlighting for JavaScript keywords and the available MIDI API (Application Programming Interface).
  - Live syntax checking, which highlights error lines immediately, making it easier to write your scripts.
  - Line numbers, which are useful for error checking because they are reported by line number in the Interactive Console.

- *Interactive Console:* Displays debugging information and allows you to execute code on the command prompt by typing after the prompt and pressing Return. Type *clear* and press Return to clear the console.

Create and store a Scripter plug-in

1 Open the Scripter plug-in.

2 Click the Open Script In Editor button.

3 Type (or copy and paste existing) JavaScript code in the Code Editor.

4 Click the Run Script button.

5 Test your plug-in to verify it behaves as intended.

6 Assuming no errors are shown in the Interactive Console, save the host document, setting, or patch containing the script.

Scripter API overview

You can create your own MIDI processing plug-ins using the JavaScript API described in these sections.

- MIDI processing functions overview
- JavaScript objects overview
- Create Scripter controls

Tip: View the supplied scripts in the Script Editor to see how they are constructed. You can modify and re-use the code to change functions or to create new processors. See Use the Script Editor.

MIDI processing functions

MIDI processing functions overview

The Scripter plug-in exposes a set of JavaScript functions that you can implement in your script to communicate with the host application. See the linked sections for details about different JavaScript function types and how to use them to create MIDI plug-ins:

- HandleMIDI function
- ProcessMIDI function
- ParameterChanged function
- Reset function

HandleMIDI function

The `HandleMIDI()` function lets you process MIDI events that the plug-in receives. HandleMIDI is called each time a MIDI event is received by the plug-in and is required to process incoming MIDI events. If you do not implement the HandleMIDI function, events pass through the plug-in unaffected.

HandleMIDI is called with one argument, which is a JavaScript object that represents the incoming MIDI event. HandleMIDI and JavaScript Event object use is shown in the examples.

Code example 1
Pass MIDI events through the plug-in.

```
function HandleMIDI(event) {
event.send();  }
```

Code example 2
Log events to the plug-in console and do not send them anywhere.

```
function HandleMIDI(event) {
event.trace();
}
```

Code example 3

Repeat notes up one octave with 100ms delay and pass all other events through.

Text following "//" are comments.

```
function HandleMIDI(event) {
event.send();                        // send original event
if (event instanceof Note) {         // if it's a note
event.pitch += 12;                   // transpose up one octave
event.sendAfterMilliseconds(100);    // send after delay
    }
}
```

ProcessMIDI function

The *ProcessMIDI()* function lets you perform periodic (generally timing-related) tasks. This can be used when scripting a sequencer, an arpeggiator, or another tempo-driven MIDI effect. ProcessMIDI is generally not required for applications that do not make use of musical timing information from the host. ProcessMIDI is called once per "process block," which is determined by the host's audio settings (sample rate and buffer size).

This function is often used in combination with the "JavaScript TimingInfo object" to make use of timing information from the host application. The use of ProcessMIDI and the TimingInfo object is shown in the example. Also see Use the JavaScript TimingInfo object.

Note: To enable the GetTimingInfo feature, you need to add `NeedsTimingInfo = true;` at the global script level (outside of any functions).

Code example

```
// Define NeedsTimingInfo as true at the global scope to enable GetHostInfo()
NeedsTimingInfo = true;
function ProcessMIDI() {
    var info = GetTimingInfo();          // get a TimingInfo object from the host
    if (info.playing) {                  // if the transport is running
        Trace(info.tempo);               // print the tempo in the plugin console
}
}
```

GetParameter function

The `GetParameter()` function retrieves information from parameters defined with `var` PluginParameters.

The GetParameter name argument must match the defined PluginParameters name value.

Code use example

Text following "//" describes the argument function. Open the Mod Wheel Glissando JavaScript in the Script Editor to see how the GetParameter function is used.

```
note.velocity = GetParameter("Note Velocity");   // used within a HandleMIDI
    function, retrieves "Note Velocity" information from the defined "Note
    Velocity" parameter
var PluginParameters = [{name:"name:"Note Velocity", type:"lin", minValue:1,
    maxValue:127, numberOfSteps:126, defaultValue:80"}];   // create a linear
    parameter called "Note Velocity" with a range of 1 to 127, and a default
    value of 80
```

ParameterChanged function

The `ParameterChanged()` function lets you perform tasks triggered by changes to plug-in parameters. ParameterChanged is called each time one of the plug-in's parameters is set to a new value. ParameterChanged is also called once for each parameter when you load a plug-in setting.

ParameterChanged is called with two arguments, first the `parameter index` (an integer number starting from 0), then the `parameter value` (a number).

Code example

Print parameter changes to the plug-in console. This example also creates a slider in the plug-in window and assigns the ParameterChanged function to it.

Text following "//" describes the argument function.

```
var PluginParameters = [{name:"Slider"}];    // create a slider (default range 0.0
    - 1.0)
function ParameterChanged(param, value) {
    if (param == 0)                          // if it's the slider you just
    created
Event.trace(value);                          // print the value to the console
}
```

Reset function

`Reset()` is called when the plugin is reset.

JavaScript objects

JavaScript objects overview

The Scripter plug-in provides JavaScript objects that describe or represent MIDI information and information about the host application, in addition to performing MIDI processing-related functions. See the following sections for details about different object types:

- Use the JavaScript Event object
- Use the JavaScript TimingInfo object
- Use the JavaScript MIDI object

Use the JavaScript Event object

When the HandleMIDI function is called, an Event object represents one MIDI event and implements the following methods you can call in your script:

Event methods

- *Event.send():* Send the event.
- *Event.sendAfterMilliseconds(number ms):* Send the event after the specified value has elapsed (can be an integer or a floating point number).
- *Event.sendAtBeat(number beat):* Send the event at a specific beat (floating point number) in the host's timeline.
- *Event.sendAfterBeats(number beat):* As above, but uses the beat value as a delay in beats from the current position.
- *Event.trace():* Print the event to the plug-in console.
- *Event.toString():* Returns a String representation of the event.
- *Event.channel(number):* Sets MIDI channel 1 to 16.

    Note: `Event.channel` is an event property, rather than a method.

The Event object is not instantiated directly but is a prototype for the following event-specific object types. All of the following types inherit the methods described above and the channel property.

Event types

The event types and their properties are passed to HandleMIDI as follows:

- *NoteOn.pitch(integer number):* Pitch from 1–127.
- *NoteOn.velocity(integer number):* Velocity from 0–127. A velocity value of 0 is interpreted as a note off event, not a note on.
- *NoteOff.pitch(integer number):* Pitch from 1–127.
- *NoteOff.velocity(integer number):* Velocity from 0–127.
- *PolyPressure.pitch(integer number):* Pitch from 1–127. Polyphonic aftertouch is uncommon on synthesizers.
- *PolyPressure.value(integer number):* Pressure value from 0–127.
- *ControlChange.number(integer number):* Controller number from 0–127.
- *ControlChange.value(integer number):* Controller value from 0–127.

    Tip: Use MIDI.controllerName(number) to look up the name of the controller.

- *ProgramChange.number(integer number):* Program change number from 0–127.
- *ChannelPressure.value(integer number):* Aftertouch value from 0–127.
- *PitchBend.value(integer number):* 14-bit pitch bend value from -8192–8191. A value of 0 is center.

Replace every MIDI event received with a modulation control change message

- Type the following in the Script Editor window. Text following "//" describes the argument function.

Tip: You can use the JavaScript "new" keyword to generate a new instance of an Event object of any type.

```
function HandleMIDI() {
    var cc = new ControlChange;     // make a new control change message
    cc.number = 1;                  // set it to controller 1 (modulation)
    cc.value = 100;                 // set the value
    cc.send();                      // send the event
    cc.trace();                     // print the event to the console
}
```

Replace every MIDI event received with a C3 note on/off

- Type the following in the Script Editor window. Text following "//" describes the argument function.

Tip: You can use the JavaScript "new" keyword to generate a new instance of an Event object of any type.

```
function HandleMIDI() {
    var on = new NoteOn;            // make a new note on
    on.pitch = 60;                  // set its pitch to C3
    on.send();                      // send the note
    var off = new NoteOff(on);      // make a note off using the note on to
     initialize its pitch value (to C3)
    off.sendAfterBeats(1);          // send a note off one beat later
}
```

Use the JavaScript TimingInfo object

The TimingInfo object contains timing information that describes the state of the host transport and the current musical tempo and meter. A TimingInfo object can be retrieved by calling GetTimingInfo().

TimingInfo properties

- *TimingInfo.playing:* Uses Boolean logic where "true" means the host transport is running.
- *TimingInfo.blockStartBeat:* A floating point number indicates the beat position at the start of the process block.
- *TimingInfo.blockEndBeat:* A floating point number indicates the beat position at the end of the process block.
- *TimingInfo.blockLength:* A floating point number indicates the length of the process block in beats.
- *TimingInfo.tempo:* A floating point number indicates the host tempo.
- *TimingInfo.meterNumerator:* An integer number indicates the host meter numerator.
- *TimingInfo.meterDenominator:* An integer number indicates the host meter denominator.
- *TimingInfo.cycling:* Uses Boolean logic where "true" means the host transport is cycling.
- *TimingInfo.leftCycleBeat:* A floating point number indicates the beat position at the start of the cycle range.
- *TimingInfo.rightCycleBeat:* A floating point number indicates the beat position at the end of the cycle range.

   Note: The length of a beat is determined by the host application time signature and tempo.

Print the beat position while the transport is running

- Type the following in the Script Editor window:

```
var NeedsTimingInfo = true;
function ProcessMIDI() {
    var info = GetTimingInfo();
    if (info.playing)
        Trace(info.beat)
}
```

Use the JavaScript MIDI object

The MIDI object contains a number of convenient and easy to use functions that can be used when writing your scripts.

Note: The MIDI object is a property of the global object, which means that you do not instantiate it but access its functions much like you would the JavaScript Math object. An example is calling MIDI.allNotesOff() directly.

MIDI object properties

Use the following method names and arguments to perform these functions:

- *noteNumber(string name):* Returns the MIDI note number for a given note name. For example: C3 or B#2.

  Note: You cannot use flats in your argument. Use A#3, not Bb3.

- *noteName(number pitch):* Returns the name (string) for a given MIDI note number.
- *ccName(number controller):* Returns the controller name (string) for a given controller number.
- *allNotesOff():* Sends the all notes off message on all MIDI channels.
- *normalizeStatus(number status):* Normalizes a value to the safe range of MIDI status bytes (128–239).
- *normalizeChannel(number channel):* Normalizes a value to the safe range of MIDI channels (1–16).
- *normalizeData(number data):* Normalizes a value to the safe range of MIDI data bytes (0–127).

Pass events through and send all notes off message when receiving controller 20

Type the following in the Script Editor window:

```
function HandleMIDI(e) {
    e.send();
    if (e instanceof ControlChange && e.number == 20)
        MIDI.allNotesOff();
}
```

Create Scripter controls

The Scripter Script Editor lets you use a simple shorthand to add standard controllers such as sliders and menus for automated or real-time control of your plug-ins. The only mandatory property to define a new parameter is a name, which defaults to a basic slider. In addition, you can add the following properties to change the type and behavior of controls.

Optional properties

- *type:* Type one of the following strings as the value:
  - *"lin":* Creates a linear fader.
  - *"log":* Creates a logarithmic fader.
  - *"menu":* Creates a menu.
  - *"valueStrings":* The menu type requires an additional property that is an array of strings to show in the menu.
- *defaultValue:* Type an integer or floating point number to set a default value. If no value is typed, the default is 0.0.
- *minValue:* Type an integer or floating point number to set a minimum value. If no value is typed, the default is 0.0.
- *maxValue:* Type an integer or floating point number to set a maximum value. If no value is typed, the default is 1.0.

Define MIDI plug-in controls

Type the following in the Script Editor window to create these controller types:

- *Slider 1:* `var PluginParameters = [{name:"Parameter x", defaultValue:0.5}];`

  The code example results in a slider named "Parameter x" with a default range of 0 to 1. It is set to the mid-point of 0.5.

- *Slider 2:* `var PluginParameters = [{name:"Octaves", defaultValue:3, minValue:0, maxValue:5, numberOfSteps:5, unit:"octaves", type:"lin"}];`

  This code example results in a linear slider type, with five possible positions (steps), and a range from 0 to 5.

- *Menu:* `var PluginParameters = [{name:"Range", type:"menu", valueStrings:["Low", "Mid", "High"]}];`

  This code example creates a menu named "Range" with the options "Low", "Mid", and "High".

Retrieve plug-in parameter values

Call *GetParameter()* with the parameter name to return a value (number object) with the parameter's current value. GetParameter() is typically used inside the HandleMIDI function or ProcessMIDI function.

This code example converts modulation events into note events and provides a slider to determine note lengths.

▪ Type the following in the Script Editor window. Text following "//" describes the argument function.

```
var PluginParameters = [{name:"Note Length"}];           // create a slider
     (default range 0.0 - 1.0)
function HandleMIDI(e) {
    if (e instanceof ControlChange && e.number == 1) {   // if event is
     modulation wheel
    var note = new NoteOn;                               // create a NoteOn
     object
    note.pitch = e.value;                               // use cc value as note
     pitch
    note.velocity = 100;                                // use velocity 100
    note.send();                                        // send note on
    var off = new NoteOff(note);                        // create a NoteOff
     object that inherits the NoteOn's pitch and velocity
    var delayInBeats = GetParameter("Note Length") + 0.1; // retrieve the
     parameter value of the slider you created (add 0.1 to guarantee note on and
     off are not simultaneous)
    off.sendAfterBeats(delayInBeats);                   // send note off after
     the length in beats is set via the slider
    }
}
```

Transposer MIDI plug-in

The Transposer MIDI plug-in can transpose incoming MIDI notes in real time and can correct notes to a selected scale.
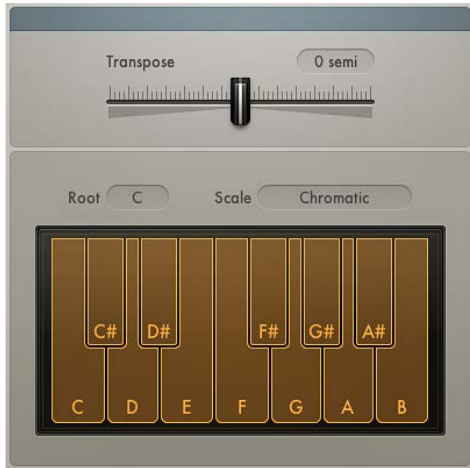
Transposer parameters

- *Transpose slider:* Drag to transpose incoming MIDI Notes by ± 24 semitones.
- *Root pop-up menu:* Choose the root note for the scale.
- *Scale pop-up menu:* Choose one of several preset scales or create your own custom scale (User) with the onscreen keyboard.
- *Keyboard:* Click notes on the Keyboard to switch them on or off. Notes that are turned off are excluded from the User scale.

Velocity Processor MIDI plug-in

Velocity Processor overview

The Velocity Processor MIDI plug-in processes incoming MIDI velocity events—note on and note off—in real time. Among other applications, it allows velocity compression and expansion.

Velocity Processor global parameters

- *Process buttons:* Click either button to process MIDI note on velocity *or* MIDI note off velocity. Click both buttons to process MIDI note on *and* MIDI note off velocity.

- *Mode pop-up menu:* Choose a velocity processing mode. The available parameters change depending on the mode selected.

  - *Compress/Expand:* In Velocity Processor Compress/Expand mode, the plug-in behaves like an audio compressor.

  - *Value/Range:* In Velocity Processor Value/Range mode, the plug-in behaves like an audio limiter.

  - *Add/Scale:* In Velocity Processor Add/Scale mode, the plug-in scales, adds to, or reduces the values of incoming MIDI velocity messages.

  - *Input Min and Input Max sliders (extended parameter):* Click the disclosure triangle at the lower left to open the extended parameters. Move the Input Min and Input Max sliders to set an input note range. Notes that fall within the input note range are processed (default range: 1–127). Notes outside the input note range are not processed.

  Note: You can cross over the Input Min and Input Max sliders, which inverts the input note range behavior: note events that fall within the range are *not* processed and note events outside the range have their velocities processed.

  - *Range Learn checkbox (extended parameter):* Click to turn on Learn mode, then play a (low) key on your MIDI keyboard to set the Input Min value. Play a (high) key to set the Input Max value.

  Once both keys have been played, Learn mode is automatically turned off, and the Range Learn checkbox is cleared.

Velocity Processor Compress/Expand mode

In Compress/Expand mode, the Velocity Processor MIDI plug-in behaves like an audio compressor.

Compress/Expand mode parameters

- *Threshold knob:* Rotate to set a velocity value. Incoming velocities above the threshold are processed. MIDI notes with velocity values below the threshold pass through unaffected.
- *Ratio knob:* Rotate to determine the slope of compression/expansion above the threshold. Processing is done using a "soft knee" characteristic.
  - Ratios smaller than 1 result in an expansion of incoming velocity values.
  - Ratios greater than 1 result in compression of incoming velocity values.
- *Make-up knob:* Rotate to set a velocity offset to compensate for the higher or lower overall velocity caused by compression/expansion. The velocity offset can be positive or negative, either adding to or subtracting from incoming velocity values.
- *Auto (Gain) button:* Turn on to automatically apply a maximum velocity reference level, set with the Make-up knob.

  Note: When the Auto button is active, the Make-up knob changes function: instead of setting the velocity offset value, it sets the maximum velocity reference level.

Velocity Processor Value/Range mode

In Value/Range mode, the Velocity Processor MIDI plug-in can behave like an audio limiter.
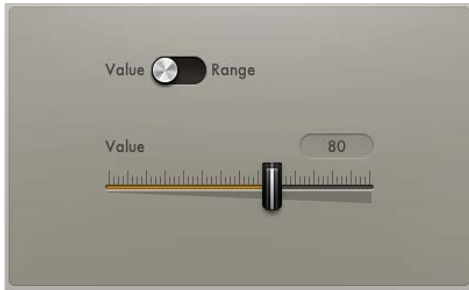
Value/Range mode parameters

- *Value/Range switch:* Set to Value to limit all incoming MIDI velocity values to the value set with the Value slider. Set to Range to limit all incoming MIDI velocity values to the range set with the Min and Max sliders.

- *Value slider:* Drag to set a fixed velocity for all processed notes.

Velocity Processor Add/Scale mode

In Add/Scale mode, the Velocity Processor MIDI plug-in scales, adds to, or reduces the values of incoming MIDI velocity messages.
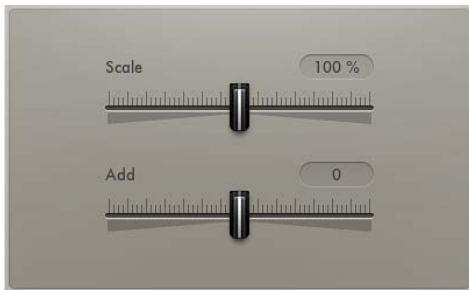
Add/Scale mode parameters

- *Scale slider:* Move to scale all incoming MIDI velocity values by a percentage from zero to 200%.

- *Add slider:* Move to add the set value to, or subtract it from, incoming MIDI velocity values.

Modulation effects

10

Modulation effects overview

Modulation effects—such as chorus, flanging, and phasing—are used to add motion and depth to your sound.

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

You can also control the ratio between 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, which uses an LFO to modulate the frequency of the sound. Unlike other pitch modulation effects, vibrato alters only the delayed signal.

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

Chorus effect

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

Chorus parameters

- *Intensity slider and field:* Drag to set the modulation amount.
- *Rate knob and field:* Rotate to set the frequency, or speed, of the LFO.
- *Mix slider and field:* Drag to determine the balance between dry and wet signals.

Ensemble effect

Ensemble can add richness and movement to sounds, particularly when you use a high number of voices. It is useful for thickening parts, but you can also use it for strong pitch variations between voices, resulting in a detuned quality to processed material. Ensemble combines up to eight chorus effects. Two standard LFOs and one random LFO enable you to create complex modulations. The graphic display visually represents what is happening with processed signals.

Ensemble parameters

- *Intensity sliders and fields:* Drag to set the amount of modulation for LFO 1, LFO 2, and random modulation.

- *Rate knobs and fields:* Rotate to control the frequency of LFO 1, LFO 2, and random modulation.

- *Voices slider and field:* Drag to determine how many individual chorus instances are used. This setting determines the number of voices, or signals, that are generated in addition to the original signal.

- *Graphic display:* Indicates the shape and intensity of the modulations.

- *Phase knob and field:* Rotate to control 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:* Drag to distribute voices across the stereo field. You can set a value of 200% to artificially expand the stereo base. Note that monaural compatibility may suffer if you do this.

- *Mix slider and field:* Drag to set the balance between dry and wet signals.

- *Effect Volume knob and field:* Rotate to set 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 add a spacey or underwater quality to input signals.

Flanger parameters

- *Feedback slider and field:* Drag to set the amount of the effect signal that is routed back into the input. This can change the tonal color and make the sweeping effect more pronounced. Negative Feedback values invert the phase of the routed signal.

- *Speed knob and field:* Rotate to set the frequency (speed) of the LFO.

- *Intensity slider and field:* Drag to determine the modulation amount.

- *Mix slider and field:* Drag to determine the balance between dry and wet signals.

Microphaser

Microphaser enables you to quickly create swooshing, phasing effects.

Microphaser parameters

- *LFO Rate slider and field:* Drag to set the frequency (the speed) of the LFO.

- *Feedback slider and field:* Drag to determine the amount of effect signal routed back into the input. This can change the tonal color and make the sweeping effect more pronounced.

- *Intensity slider and field:* Drag to determine the amount of modulation.

Modulation Delay

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.

Modulation Delay parameters

- *Feedback slider and field:* Drag to set the amount of effect signal routed back to the input. Use a high Feedback value for strong modulations. If you want to double the signal, don't use Feedback. Negative values invert the phase of the feedback signal, resulting in more chaotic effects.

- *Flanger-Chorus knob and field:* Rotate to set 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:* Turn on to make sure the pitch of the modulated signal remains constant.

- *Const(ant) Mod(ulation) button:* Turn on to make sure the modulation width remains constant, regardless of the modulation rate.

    Note: When Const Mod is enabled, higher modulation frequencies reduce the modulation width.

- *Mod(ulation) Intensity slider and field:* Drag to set the modulation amount.

- *LFO Mix slider and fields:* Drag to determine the balance between the two LFOs.

- *LFO 1 and LFO 2 Rate knobs and fields:* Rotate to set the modulation rate for the left and right stereo channels.

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

- *LFO Left Right Link button:* Turn on to link the modulation rates of the left and right stereo channels. Adjustment of either Rate knob will affect the other channel in stereo instances.

- *LFO Phase knob and field:* Rotate to control the phase relationship between channel modulations. Available only in stereo instances.

    - At 0°, the extreme values of the modulation are achieved simultaneously for both channels.

    - At 180° or −180°, you achieve 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:* Drag to determine the impact of LFO modulation on the amplitude of the effect signal.
- *Output Mix slider and field:* Drag to determine the balance between dry and wet signals.
- *All Pass button (Extended Parameters area):* Turn on to introduce 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):* Drag to determine the frequency at which the phase shift crosses 90°—the halfway 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 parameters

- *Filter button:* Turns the filter section on or off, which processes the feedback signal.
- *LP and HP knobs and fields:* Rotate to set the cutoff frequency of the lowpass (LP) and highpass (HP) filters.
- *Feedback slider and field:* Drag to determine the amount of effect signal routed back into the input.
- *Ceiling and Floor sliders and fields:* Drag to determine the frequency range affected by the LFO modulations. Drag the blue area to move the entire range.
- *Order slider and field:* Drag to choose phaser algorithms. The more orders, the stronger the effect.
  - The 4, 6, 8, 10, and 12 settings switch between five different phaser algorithms. All are modeled on analog circuits, with each designed for a specific application.
  - The odd-numbered settings (5, 7, 9, and 11) don't generate actual phasing effects. The more subtle comb filtering effects produced by odd-numbered settings can, however, be useful.
- *Env Follow slider and field:* Drag to determine the impact of incoming signal levels on the frequency range (set with the Ceiling and Floor controls).
- *LFO 1 and LFO 2 Rate knobs and fields:* Rotate to set the speed for each LFO.
- *LFO Mix slider and fields:* Drag to determine the ratio between the two LFOs.
- *Env Follow slider and field:* Drag to determine the impact of incoming signal levels on the speed of LFO 1.
- *Phase knob and field:* Rotate to control the phase relationship between the individual channel modulations. Available only in stereo instances. At 0°, the extreme values of the modulation are achieved simultaneously for both channels. At 180° or −180°, there is the greatest possible distance between the modulation phases of the channels.
- *Output Mix slider and field:* Determines the balance of dry and wet signals. Negative values result in a phase-inverted mix of the effect and direct (dry) signal.
- *Warmth button:* Click to turn on a distortion circuit, which is suitable for warm overdrive effects.

Ringshifter

Ringshifter overview

Ringshifter combines a ring modulator with a frequency shifter effect. Both effects were popular during the 1970s and are currently experiencing 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 of, and the difference between, the frequency content in the two original signals. Its sound is often described as *metallic* or *clangorous*.

The frequency shifter moves the frequency content of the input signal by a fixed amount and thereby alters the frequency relationship of the original harmonics. The resulting sounds range from sweet and spacious phasing effects to robot-like timbres.

Note: Do not confuse frequency shifting with pitch shifting. Pitch shifting transposes the original signal, leaving its harmonic frequency relationship intact.

Ringshifter interface

The Ringshifter interface consists of six main sections.

- *Mode buttons:* Determine whether Ringshifter operates as a frequency shifter or a ring modulator. See Set the Ringshifter mode on page 195.
- *Oscillator parameters:* Configure the internal sine wave oscillator, which modulates the amplitude of the input signal in both of the frequency shifter modes as well as in the ring modulator OSC mode. See Ringshifter oscillator parameters on page 196.
- *Delay parameters:* Delay the effect signal. See Ringshifter delay parameters on page 197.
- *Envelope follower parameters:* Modulate the oscillator frequency and output signal with an envelope follower. See Ringshifter envelope follower modulation on page 197.
- *LFO parameters:* Modulate the oscillator frequency and output signal with an LFO. See Ringshifter LFO modulation on page 198.
- *Output parameters:* Set feedback, stereo width, and the amount of dry and wet signals. See Ringshifter output parameters on page 199.

Set the Ringshifter mode

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

Ringshifter mode parameters

- *Single Freq(uency) Shift(er) button:* The frequency shifter generates a single, shifted effect signal.

  The oscillator Frequency control determines whether the signal is shifted to a positive value on the right side of the Frequency knob or to a negative value on the left side.

- *Dual Freq(uency) Shift(er) button:* 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 towards the left or the right channel.

- *OSC Ring Mod(ulator) button:* The ring modulator uses the internal sine wave oscillator to modulate the input signal.

- *Side Chain Ring Mod(ulator) 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 disabled when Side Chain mode is active.

Ringshifter oscillator parameters

In both of the frequency shifter modes as well as in 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, either up or down, applied to the input signal.
- In the ring modulator OSC mode, the Frequency parameter controls the frequency content, or timbre, of the resulting effect. This timbre can range from subtle tremolo effects to clangorous metallic sounds.

Oscillator parameters

- *Frequency control:* Rotate to set the frequency of the sine oscillator.
- *Lin(ear) and Exp(onential) buttons:* Switch the scaling of the Frequency control:
  - *Lin(ear):* Linear scaling is even across the entire control range.
  - *Exp(onential):* Exponential scaling offers extremely small increments around the 0 point, which is useful for programming slow-moving phasing and tremolo effects.
- *Env(elope) Follow slider and field:* Drag to determine the impact of incoming signal levels on the oscillator modulation depth.
- *LFO slider and field:* Drag to determine the amount of oscillator modulation by the LFO.

Ringshifter delay parameters

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

Delay parameters

- *Time knob and field:* Rotate to set the delay time—in Hz when running freely, or in note values, including triplet and dotted notes, when the Sync button is selected.
- *Sync button:* Turn on to synchronize the delay to the host application tempo. You can choose musical note values with the Time knob.
- *Level knob and field:* Rotate to set 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).

Ringshifter modulation

Ringshifter envelope follower modulation

The oscillator Frequency and Dry/Wet parameters can be modulated with the internal envelope follower—and the LFO (see Ringshifter LFO modulation on page 198). 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 to create a continuously changing control signal—a dynamic volume envelope of the input signal. This control signal can be used for modulation purposes.

Envelope Follower parameters

- *Power button:* Turns the envelope follower on or off. When it is turned on, you can access the following parameters:
- *Sens(itivity) slider and field:* Drag to determine 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:* Drag to set the response time of the envelope follower.
- *Decay slider and field:* Drag to set the time it takes the envelope follower to return from a higher to a lower value.

Ringshifter LFO modulation

The oscillator Frequency and Dry/Wet parameters can be modulated with the LFO—and the envelope follower (see Ringshifter envelope follower modulation on page 197). The oscillator frequency even allows modulation through the 0 Hz point, thus changing the oscillation direction. The LFO produces continuous, cycled control signals.

Ringshifter LFO parameters

- *Power button:* Turns the LFO on or off. When it is turned on, you can access the following parameters.
- *Symmetry and Smooth sliders and fields:* Drag to change the shape of the LFO waveform.
- *Waveform display:* Provides visual feedback about the LFO waveform shape.
- *Rate knob and field:* Rotate to set the waveform cycle speed of the LFO.
- *Sync button:* Turn on to synchronize the LFO rate with the host application tempo, using musical note values.

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 level of the Ringshifter.

Ringshifter Output parameters

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

- *Feedback knob and field:* Rotate to set the amount of signal routed back to the effect input. Feedback adds an edge to the Ringshifter sound and is used for a variety of special effects.

  - Feedback, in combination with a slow oscillator sweep, produces a rich phasing sound.

  - A high Feedback setting with a short delay time of less than 10 ms produces comb-filtering effects.

  - A high Feedback setting with a longer delay time produces continuously rising and falling frequency shift effects.

- *Stereo Width knob and field:* Rotate to determine 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(elope) Follower slider and field:* Drag to determine the amount of Dry/Wet parameter modulation by the input signal level.

- *LFO slider and field:* Drag to set the LFO modulation depth of the Dry/Wet parameter.

Rotor Cabinet effect

Rotor Cabinet effect overview

The Rotor Cabinet effect emulates the rotating loudspeaker cabinet of a Hammond organ. Also known as the *Leslie effect*, it simulates both the rotating speaker cabinet, with and without deflectors, and the microphones that pick up the sound.

Basic Rotor Cabinet parameters

- *Rotation switch:* Move to change the rotor speed between Slow, Brake, or Fast.
- *(Cabinet) Type pop-up menu:* Click to choose a cabinet model:
  - *Wood:* Mimics a Leslie with a wooden enclosure, sounding like the Leslie 122 or 147 model.
  - *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, sounding like the Leslie 825 model.
  - *Split:* Routes the bass rotor's signal slightly to the left and the treble rotor's signal toward the right.
  - *Wood & Horn IR:* Uses an impulse response of a Leslie with a wooden enclosure.
  - *Proline & Horn IR:* Uses an impulse response of a Leslie with a more open enclosure.
  - *Split & Horn IR:* 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.
- *Deflector switch:* Emulates a Leslie cabinet with the horn deflectors removed or attached. A Leslie cabinet contains a double horn, with a deflector at the horn mouth. This deflector makes the Leslie sound. You can remove the deflector to increase amplitude modulation and decrease frequency modulation.

Rotor Cabinet effect motor parameters

The Rotor Cabinet effect provides the following motor control parameters.


Motor parameters

- *Acceleration knob:* Rotate to set the time it takes to get the rotors up to the speed set with the Max Rate knob, and the length of time it takes for them to slow down. The Leslie motors need to physically accelerate and decelerate the speaker horns in the cabinets, and their power to do so is limited. Turn Acceleration to the far left position to switch to the preset speed immediately. As you rotate the knob to the right, it takes more time to hear the speed changes. At the default, centered, position the behavior is Leslie-like.

- *Max Rate knob:* Rotate to set the maximum possible rotor speed.

- *Motor Control pop-up menu:* Set different speeds for the bass and treble rotors in the pop-up menu. Use the Rotation switch to choose slow, brake, or fast mode.

  - *Normal:* Both rotors use the speed determined by the Rotation switch position.

  - *Inv (inverse):* In fast mode, the bass compartment rotates at a fast speed, while the horn compartment rotates slowly. This is reversed in slow mode. In brake mode, both rotors stop.

  - *910:* The 910 (also known as "Memphis"), stops the bass drum rotation at slow speed, while the speed of the horn compartment can be switched. This is useful when 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 drums are locked, but the effect is clearly audible only during acceleration or deceleration.

Note: If you choose Single Cabinet from the (Cabinet) Type pop-up menu, the Motor Control setting is not relevant because there are no separate bass and treble rotors in a single cabinet.

Rotor Cabinet effect microphone types

The Rotor Cabinet effect provides modeled microphones that pick up the sound of the Leslie cabinet. You can specify the microphone type with these parameters. Also see Rotor Cabinet effect mic processing controls.

Mic Position switch

Click to choose a microphone type.

Click to choose a microphone type.
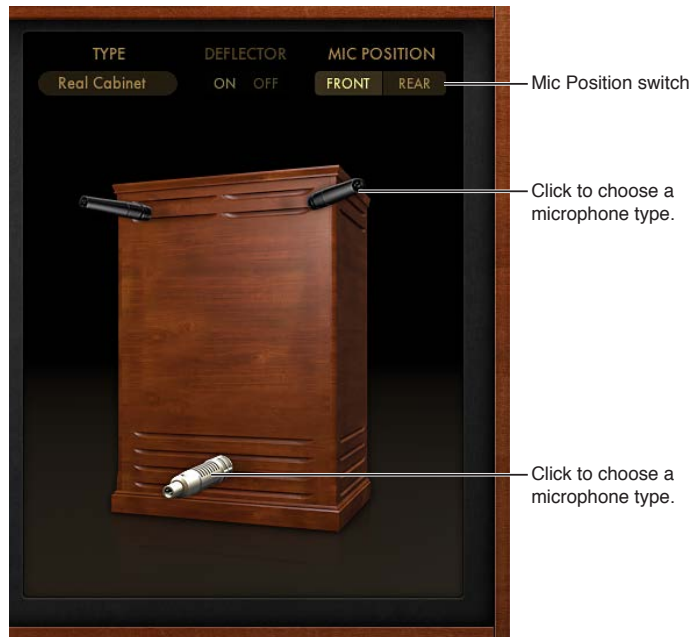
Click the microphone icons to choose a microphone type for the horn and drum speakers when Real Cabinet is chosen in the Type pop-up menu. See Rotor Cabinet effect overview.

- *Dynamic:* Emulates the sound of a dynamic cardioid microphone. This microphone type sounds brighter and more cutting than the Condenser mic.

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

- *Mid-Side Mic:* A Middle and Side (MS) configuration where two microphones are positioned closely together. One is a cardioid (or omnidirectional) microphone that directly faces the cabinet—in a straight alignment. The other is a bidirectional microphone, with its axes pointing to the left and right of the cabinet at 90° angles. The cardioid microphone captures the middle signal to one stereo side. The bidirectional microphone captures the side signal to the other stereo side.

Rotor Cabinet effect mic processing controls

The Rotor Cabinet effect provides the following microphone processing parameters.

Mic processing parameters

- *Mic Position switch:* Choose either the front or rear position for the virtual microphone. See Rotor Cabinet effect microphone types.
- *When Real Cabinet is chosen in the Type pop-up menu:*
  - *Horn knob:* Rotate to define the stereo width of the Horn deflector microphone.
  - *Drum knob:* Rotate to define the stereo width of the Drum deflector microphone.
- *When other cabinets are chosen in the Type pop-up menu:*
  - *Distance knob:* Rotate to determine the distance of the virtual microphones (the listening position) from the emulated speaker cabinet. Turn to the right for a darker and less defined sound.
  - *Angle knob:* Rotate to define the stereo image, by changing the angle of the simulated microphones between 0 and 180 degrees.
- *Balance knob:* Rotate to set the balance between the horn and drum microphone signals.

Scanner Vibrato effect

Scanner Vibrato simulates the scanner vibrato section of a Hammond organ. Scanner Vibrato is based on an analog delay line consisting of several lowpass filters. The delay line is scanned by a multipole capacitor that 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.

Scanner Vibrato parameters

- *Type knob:* Rotate to choose from three Vibrato positions (V1, V2, and V3) or three Chorus positions (C1, C2, and C3).
  - In each of the Vibrato positions, only the delay line signal is heard. Each vibrato type has a different intensity.
  - In the three Chorus positions (C1, C2, and C3), the signal of the delay line is mixed 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 Chorus plug-in.
  - In the C0 setting, neither the chorus nor the vibrato is enabled.
- *Depth knob:* Rotate to set the intensity of the chosen chorus effect type. If a vibrato effect type is chosen, this parameter has no effect.
- *Stereo Phase knob:* Rotate to determine the phase relationship between left and right channel modulations. If you set the knob to free, you can set the modulation speed of the left and right channels independently.
- *Rate Left knob:* Rotate to set 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. The Rate Right knob has no function when in this mode.
- *Rate Right knob:* Rotate to set the modulation speed of the right channel when Stereo Phase is set to free.

Spreader

Spreader widens the stereo spectrum of a signal by periodically shifting the frequency range of the original signal, thus changing the perceived width of the signal. You can also use Spreader to specify the delay between channels in samples, which adds to the perceived width and channel separation of a stereo input signal.

Spreader parameters

- *Intensity slider and field:* Drag to determine the modulation amount.
- *Speed knob and field:* Rotate to set the speed of the built-in LFO.
- *Channel Delay slider and field:* Drag to set the delay time in samples.
- *Mix slider and field:* Drag to determine the balance between the effect and input signals.

Tremolo effect

Tremolo modulates the amplitude of the incoming signal, resulting in periodic volume changes. Tremolo is commonly used in vintage guitar combo amps, where it is sometimes incorrectly referred to as *vibrato*. The graphic waveform display shows all the parameters except Rate.

Tremolo parameters

- *Depth slider and field:* Drag to determine the modulation amount.
- *Waveform display:* Shows the resulting waveform.
- *Rate knob and field:* Rotate to set the frequency of the LFO.
- *Symmetry and Smoothing knobs and fields:* Rotate to change the shape of the LFO waveform.

  If Symmetry is set to 50% and Smoothing to 0%, the LFO waveform becomes rectangular. The timing of the highest volume signal is then equal to the timing of the lowest volume signal, with the switch between both states occurring abruptly.

- *Phase knob and field:* Rotate to control the phase relationship between individual channel modulations in stereo signals. At 0, modulation values are reached simultaneously for both channels. At values of 180 or −180, there is the greatest possible distance between the modulation phases of the channels.
- *Offset slider and field (Extended Parameters area):* Drag to set the amount that the modulation (cycle) is shifted to the left or right, resulting in subtle or significant tremolo variations.

Pitch effects

11

Pitch effects overview

You can use the pitch effects 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.

You can also define a scale to automatically correct some, but not all, sung notes in a vocal performance, for example. This enables you to effectively perfect an imperfect vocal take.

You can also use pitch correction effects creatively, modifying all pitched notes in a performance to a single pitch or a particular key.

Pitch Correction effect

Pitch Correction effect overview

You can use the Pitch Correction effect to fix 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 are almost silent when making moderate corrections.

Pitch correction works by accelerating and slowing down the audio playback speed, matching the input signal (sung vocal) with 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*). Improperly intonated notes are corrected in accordance with this scale.

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 with the Scale parameter may suffice. Less precise intonations may need more significant changes to the Scale and Root parameters.

Note: Polyphonic recordings, such as choirs, and highly percussive signals with prominent noisy portions cannot be corrected to a specific pitch. Despite this, you may want to try the plug-in on some drum sounds, such as toms and congas because it can deliver interesting results.

Pitch Correction effect parameters

Pitch Correction parameters

- *Use Global Tuning button:* Turn on to use project Tuning settings for the pitch correction process. Turn off to set the reference tuning with Ref. Pitch. See Use Pitch Correction effect reference tuning on page 211.
- *Normal and low buttons:* Click to set the pitch range that is scanned (for notes that need correction). See Pitch Correction effect quantization grid on page 209.
- *Ref. Pitch field:* Drag to set the reference tuning in cents (relative to the root).
- *Root pop-up menu:* Choose the root note of the scale.
- *Scale pop-up menu:* Choose a pitch 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 Exclude notes from pitch correction on page 210.
- *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.
- *Bypass All button:* Click to quickly compare the corrected and original signals or to audition automation changes.
- *Show input and show output buttons:* Click to show 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:* Drag to determine 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:* Drag to detune the output signal by the set value.

Pitch Correction effect 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 default setting is the chromatic scale.

The *user scale* is the scale that is set manually with the onscreen keyboard in the plug-in window. If you're unsure of the intervals used in any given scale, choose it from the Scale pop-up menu and look at the onscreen keyboard. 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, but you should try them if aiming for interesting effects.

Choose the root note of the scale from the Root pop-up menu. You can freely transpose the major and minor scales and scales named after chords.

Note: If you choose user "scale" or "chromatic" from the Scale pop-up menu, the Root pop-up menu entries are dimmed.

Exclude 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 is 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 do not 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 is corrected to an E.

Note: The settings made with the Pitch Correction effect's onscreen keyboard are valid for all octave ranges. Individual settings for different octaves are not provided.

Bypass individual notes in a scale
- To exclude notes from correction, click the small bypass buttons ("byp") above the green (black) and below the blue (white) keys.

   This is particularly 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.

   Tip: You will 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 limits correction to the root note, the fourth, and the fifth of the key scale. Turn on the bypass buttons for all other notes and only the most important and sensitive notes are corrected, while all other singing remains untouched.

Bypass all pitch correction
Not all audio material can be effectively pitch corrected.

- Click the bypass all button, to pass the input signal through unprocessed and uncorrected.

   This is useful for spot corrections to pitch through use of automation. Bypass all is optimized for near-instant, seamless operation in all situations.

Use Pitch Correction effect reference tuning

Turn on the Use Global Tuning button to use your host application Tuning settings for the pitch correction process. This ensures that all software instruments and your tuned vocal part will be in tune with each other.

If Use Global Tuning is turned off, you can use the Ref. Pitch field to set the reference tuning to the root key or note.

As an example of where Ref. Pitch can be effective, consider that the intonation of a vocal line is often slightly sharp or flat throughout an entire song. You can use Ref. Pitch to address this issue at the input of the pitch detection process by setting it 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 are individually and perfectly corrected to the same pitch, the choir effect is partially lost. You can prevent this by (de)tuning the pitch corrections individually.

Set your host application reference tuning
Do one of the following:

- *In Logic Pro:* To determine the tuning reference for all software instruments, choose File > Project Settings > Tuning.

- *In MainStage:* To determine the tuning reference for all software instruments, choose MainStage > Preferences > Tuning, then open the General tab.

Pitch Shifter

Pitch Shifter overview

Pitch Shifter provides a simple way to combine a pitch-shifted version of the signal with the original signal. Use pitch shifting to achieve the best results.

- *Semi Tones slider and field:* Drag to set the pitch shift value in semitones.
- *Cents slider and field:* Drag to control detuning of the pitch shift value in cents (1/100th of a semitone).
- *Drums, Speech, and Vocals buttons:* Set one of three optimized algorithms for common types of audio material. Choose from:
  - *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:* Drag to set the balance between the effect and original signals.
- *Timing pop-up menu (Extended Parameters area):* Choose how timing is derived:
  - *Preset:* Follows the selected algorithm.
  - *Auto:* Analyzes incoming audio.
  - *Manual:* Uses the settings of the Delay, Crossfade, and Stereo Link parameters.

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

- *Delay slider and field (Extended Parameters area):* Drag to set 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—to effectively pitch shift the signal.
- *Crossfade slider and field (Extended Parameters area):* Drag to set the range (expressed as a percentage of the original signal) used to analyze the input signal.
- *Stereo Link buttons (Extended Parameters area):* Click Inv. to invert the stereo channel's signals, with processing for the right channel occurring on the left, and vice versa. Click Normal to leave the signal as it is.

Use Pitch Shifter

Pitch Shifter is used most effectively when you take a structured approach.

Use pitch shifting

1 To set the amount of transposition, or pitch shift, drag the Semi Tones slider.

2 To set the amount of detuning, drag the Cents slider.

3 To select the algorithm that best matches the material you are working with, click the Drums, Speech, or Vocals button.

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

   Tip: While auditioning and comparing different settings, temporarily set the Mix parameter to 100% because this makes Pitch Shifter II artifacts easier to hear.

Vocal Transformer

Vocal Transformer overview

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.

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

Vocal Transformer includes the following parameters.

Vocal Transformer parameters

- *Pitch knob and field:* Rotate to determine the amount of transposition applied to the input signal. See Use the Vocal Transformer Pitch and Formant parameters.

- *Robotize button:* Turns Robotize mode on or off. This mode is used to augment, diminish, or mirror the melody. See Use Vocal Transformer's Robotize mode.

- *Pitch Base slider and field:* Available only in Robotize mode. Drag to transpose the note that the Tracking parameter is following.

- *Tracking slider, field, and buttons:* Available only in Robotize mode. Drag to control how the melody is changed.

- *Mix slider and field:* Drag to define the level ratio between the original (dry) and effect signals.

- *Formant knob and field:* Rotate to shift the formants of the input signal. See Use the Vocal Transformer Pitch and Formant parameters.

- *Glide slider and field (Extended Parameters area):* Drag to determine the time 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):* Choose how Vocal Transformer processes formants.

  - *Process always:* All formants are processed.

  - *Keep unvoiced formants:* Only voiced formants are processed. This retains sibilant sounds in a vocal performance and produces a more natural-sounding transformation effect with some signals.

- *Detune slider and field (Extended Parameters area):* Drag to detune the input signal by the set value. This parameter is of particular benefit when automated.

Use Vocal Transformer

You can change the pitch of performances, inclusive of, or independent from, formants. Robotize mode enables you to augment or diminish the melody.

Use the Vocal Transformer Pitch and Formant parameters

- *To transpose the pitch of the signal upward or downward:* Rotate the Pitch knob. 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.

 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.

- *To shift the formants while maintaining—or independently altering—the pitch:* Rotate the Formant knob. 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.

Use Vocal Transformer's Robotize mode

1 Click the Robotize button to turn on Robotize mode. In this mode, Vocal Transformer can augment or diminish the melody.

You can control the intensity of this distortion with the Tracking parameter.

2 Click one of the following buttons to immediately set the Tracking slider to one of these most useful values:

- *−1 button:* Sets the slider to −100%. All intervals are mirrored.
- *0 button:* 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 button:* Sets the slider to 100%. The range of the melody is maintained. Higher values augment, and lower values diminish, the melody.
- *2 button:* 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. For example, with Tracking set to 0%, the pitch of the (spoken) note is transposed to the chosen base pitch value.

Reverb effects

<div style="text-align: right">

12

</div>

Reverb effects overview

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.
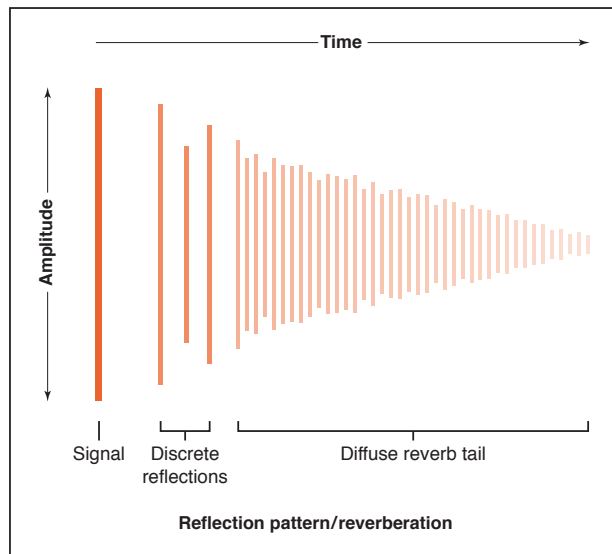
Plates, digital reverb effects, and convolution reverb

The first form of reverb used in music production was actually a purpose-built 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. See Space Designer overview on page 226.

EnVerb

EnVerb overview

EnVerb is a versatile reverb effect with a unique feature: it allows you to adjust the envelope—the shape—of the diffuse reverb tail.

EnVerb is divided 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 on page 219.
- *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 on page 220.
- *Mix slider:* Drag to determine the balance between the effect (wet) and direct (dry) signals.

EnVerb time parameters

EnVerb provides the following time parameters.

EnVerb time parameters

- *Dry Signal Delay slider and field:* Drag to determine 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%.
- *Graphic display:* Shows changes to the reverb shape when knobs below the display are adjusted.
- *Predelay knob and field:* Rotate to set the time between the original signal and the start point of the reverb attack phase—the very beginning of the first reflection.
- *Attack knob and field:* Rotate to set the time it takes for the reverb to climb to its peak level.
- *Decay knob and field:* Rotate to set the time it takes for the level of the reverb to drop from its peak to the sustain level.
- *Sustain knob and field:* Rotate to set 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:* Rotate to set the duration of the reverb sustain phase.
- *Release knob and field:* Rotate to set the time it takes for the reverb to fade out completely, after the sustain phase.

EnVerb sound parameters

EnVerb provides the following sound parameters that change the tonal color of the reverb effect.
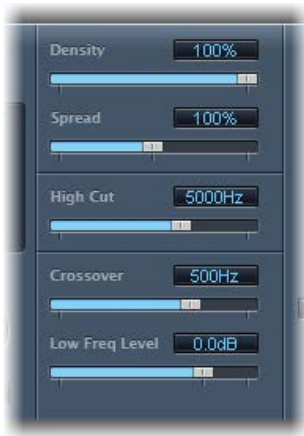
EnVerb sound parameters

- *Density slider and field:* Drag to set the reverb density.
- *Spread slider and field:* Drag to control the width of the reverb's stereo image. At 0% the effect generates a monaural reverb. At 200% the stereo base is artificially expanded.
- *High Cut slider and field:* Drag to filter frequencies above the set value out of the reverb tail.
- *Crossover slider and field:* Drag to set the frequency used to split the input signal into two frequency bands for independent processing.
- *Low Freq Level slider and field:* Drag to set 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.

PlatinumVerb

PlatinumVerb overview

PlatinumVerb allows you to edit both the early reflections and diffuse reverb tail separately, making it easy to precisely emulate real rooms. PlatinumVerb splits the incoming signal into two bands: each is processed and can be edited separately.

PlatinumVerb is divided into four parameter areas:

- *Early reflections parameters:* Emulate the original signal's first reflections as they bounce off the walls, ceiling, and floor of a natural room. See PlatinumVerb early reflections parameters on page 222.

- *Reverb parameters:* Control the diffuse reverberations. See PlatinumVerb reverb parameters on page 223.

- *Output parameters:* Determine the balance between the effected (wet) and direct (dry) signals. See PlatinumVerb output parameters on page 224.

- *Balance ER/Reverb slider:* Drag to control the balance between the early reflections and reverb signal. When the slider is set to either extreme position, the other signal is not heard.

PlatinumVerb early reflections parameters

PlatinumVerb provides the following early reflections parameters.

PlatinumVerb early reflections parameters

- *Predelay slider and field:* Drag to set the time between the start of the original signal and the arrival of the early reflections.

  - *Extremely short:* Predelay setting can color the sound and make it difficult to pinpoint the position of the signal source.

  - *Very long:* Predelay setting 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 side effects, such as an audible echo. When you reach this point, reduce the Predelay setting slightly.

- *Room Shape slider and field:* Drag to define the geometric form (the shape) of the room. The numeric value (3 to 7) represents the number of corners in the room.

- *Room Size slider and field:* Drag to determine the dimensions of the room. The numeric value indicates the length of the room's walls—the distance between two corners.

- *Graphic display:* Shows changes to Room Size and Room Shape parameters.

- *Stereo Base slider and field:* Drag to set the distance between the two virtual microphones that capture the simulated room signal.

  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):* Drag to scale early reflections along the time axis. This slider influences the Room Shape, Room Size, and Stereo Base parameters simultaneously.

PlatinumVerb reverb parameters

PlatinumVerb provides the following reverb parameters.

PlatinumVerb reverb parameters

- *Initial Delay slider and field:* Drag to set the time between the original signal and the diffuse reverb tail.

- *Spread slider and field:* Drag to control the width of the reverb's stereo image. At 0%, the effect generates a monaural reverb. At 200%, the stereo base is artificially expanded.

- *Crossover slider and field:* Drag to set the frequency used to split the input signal into two frequency bands, for separate processing.

- *Low Ratio slider and field:* Drag to determine the relative reverb times of the bass and high frequency 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:* Drag to set 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:* Drag to filter frequencies above the set value 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 sounds as if it is reflecting off stone or glass.

- *Density slider and field:* Drag to set 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 sounds grainy.

- *Diffusion slider and field:* Drag to set 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:* Drag to determine the reverb time of the high frequency 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

PlatinumVerb provides the following output parameters.

PlatinumVerb output parameters

- *Dry slider and field:* Drag to control the amount of the original signal.
- *Wet slider and field:* Drag to control the amount of the effect signal.

SilverVerb

SilverVerb provides a low frequency oscillator (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 frequency transients in reverb signals can sound unpleasant, can hamper speech intelligibility, or mask the overtones of the original signal. Long reverb tails with a lot of bass generally result in an indistinct mix.

SilverVerb parameters

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

Space Designer convolution reverb   13

Space Designer overview

Space Designer is a *convolution* reverb effect that you can use to place your audio signals in exceptionally realistic recreations of real-world acoustic environments. Space Designer generates reverb by convolving, or combining, an audio signal with an impulse response reverb sample. An impulse response is a recording of a room's reverb characteristics, or more precisely, a recording of all reflections in a given room following an initial signal spike. The actual impulse response file is a standard audio file.

To understand how this works, imagine a situation where Space Designer is used on a vocal track. An impulse response file recorded in an actual opera house is loaded into Space Designer. This impulse response file is convolved with your vocal track, placing the singer inside the opera house.

Convolution can be used to place your audio signal inside any space, including a speaker cabinet, a plastic toy, a cardboard box, and so on. All you need is an impulse response recording of the space.

In addition to loading impulse responses, Space Designer includes an on-board impulse response synthesis facility. This enables you to create completely unique effects, particularly when the synthesized impulse response doesn't represent a real space.

Space Designer also offers features such as envelopes, filters, EQ, and stereo balance controls, which provide precise control over the dynamics, timbre, and length of the reverberation.

Space Designer can operate as a mono, stereo, or true stereo (meaning each channel is processed discretely) effect.

Automation and Space Designer

Space Designer cannot be fully automated—unlike most other plug-ins. This is because Space Designer needs to reload the impulse response and recalculate the convolution before audio can be routed through it.

You can, however, record, edit, and play back any movement of the following Space Designer parameters in a suitable host application:

- Stereo Crossfeed
- Direct Output
- Reverb Output

Space Designer interface

The Space Designer interface consists of the following main sections:
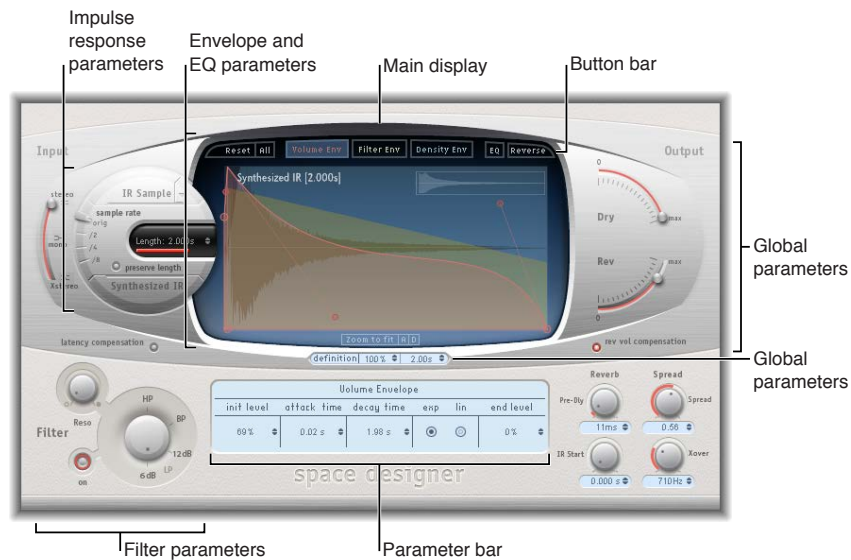
- *Impulse response parameters:* Use to load, save, or manipulate recorded or synthesized impulse response files. The chosen impulse response file determines what Space Designer will use to convolve with your audio signal. See Use impulse responses on page 228.

- *Envelope and EQ parameters:* Use the view buttons in the button bar to switch the main display and parameter bar between envelope and EQ views. Use the main display to edit the displayed parameters graphically, and use the parameter bar to edit them numerically. See Space Designer envelopes and EQ overview on page 232.

- *Filter parameters:* Use to modify the timbre of the Space Designer reverb. You can choose from several filter modes, adjust resonance, and also adjust the filter envelope dynamically over time. See Space Designer filter parameters on page 237.

- *Global parameters:* After your impulse response is loaded, use these parameters to determine how Space Designer operates on the overall signal and impulse response. Included are input and output parameters, delay and volume compensation, predelay, and so on. See Space Designer global parameters overview on page 239.

Use impulse responses

Space Designer can use either recorded impulse response files or synthesized impulse responses. The circular area to the left of the main display contains the impulse response parameters. These are used to determine the impulse response mode (IR Sample mode or Synthesized IR mode), to load or create impulse responses, and to set the sample rate and length.

Impulse response parameters

- *IR Sample button and pop-up menu:* Click the IR Sample button to switch to IR Sample mode. In IR Sample mode, an impulse response sample is used to generate reverberation. Click the down arrow next to the IR Sample button to open the IR Sample pop-up menu.
- *Synthesized IR button:* Click to turn on Synthesized IR mode. A new synthesized impulse response is generated, which is derived from the values of the Length, envelope, Filter, EQ, and Spread parameters.
- *Sample rate slider:* Move to determine the sample rate of the loaded impulse response.
- *Preserve length button:* Click to preserve the original length of the impulse response when changing the sample rate with the sample rate slider.
- *Length field:* Move to adjust the length of the impulse response.

Important: To convolve audio in real time, Space Designer must first calculate any parameter adjustments to the impulse response. This requires a moment or two following parameter edits and is indicated by a blue progress bar. During this parameter edit processing time you can continue to adjust the parameter. When calculation starts, the blue bar is replaced by a red bar, indicating that a calculation is occurring.

Turn on IR Sample mode

In IR Sample mode, Space Designer loads and uses an impulse response recording of an acoustic environment. This is convolved with the incoming audio signal to place it in the acoustic space provided by the impulse response.

1 Click the IR Sample button in the circular area to the left of the main display.

2 Select an impulse response file from any folder.

   Note: If you have already loaded an impulse response file, clicking the IR Sample button switches the mode from Synthesized IR to IR Sample mode.

Manage the loaded impulse response file

■ Click the down arrow next to the IR Sample button to open a pop-up menu containing the following commands:

   • *Load IR:* Loads an impulse response sample without changing the envelopes.

   • *Load IR & Init:* Loads an impulse response sample and initializes the envelopes.

   • *Show in Finder:* Opens a Finder window that shows the location of the currently loaded impulse response file.

   All impulse responses that ship with MainStage are installed in the /Library/Audio/Impulse Responses/Apple folder. Deconvolution files have an .sdir file extension.

   Any mono, stereo, AIFF, SDII, or WAV file can be used as an impulse response. In addition, surround formats up to 7.1, discreet audio files, and B-format audio files that consist of a single surround impulse response can also be used.

Use Synthesized IR mode

In Synthesized IR mode, Space Designer generates a synthesized impulse response based on the values of the Length, envelope, Filter, EQ, and Spread parameters.

Note: You can switch between a loaded impulse response sample and a synthesized impulse response without losing the settings of the other.

■ Click the Synthesized IR button in the Impulse Response Parameters section.

   Repeated clicks of the *active* Synthesized IR button will randomly generate new impulse responses with slightly different reflection patterns. The current impulse response state (including parameter and other values that represent the reflection patterns and characteristics of the synthetic impulse response) is saved with the setting file.

Note: Clicking the Synthesized IR button while you are in IR Sample mode switches you back to the synthesized impulse response stored with the setting.

Set the impulse response sample rate and preserve length

Changing the sample rate upward increases—or changing it downward decreases—the frequency response (and length) of the impulse response, and to a degree the overall sound quality of the reverb. Upward sample rate changes are of benefit only if the original impulse response sample actually contains higher frequencies. When reducing the sample rate, use your ears to decide if the sonic quality meets your needs.

Note: Natural room surfaces—except concrete and tiles—tend to have minimal reflections in the higher frequency ranges, making the half-rate and full-rate impulse responses sound almost identical.

- *To determine the sample rate of an impulse response:* Move the sample rate slider.

- *Orig:* Space Designer uses the current project sample rate. When loading an impulse response, Space Designer automatically converts the sample rate of the impulse response to match the current project sample rate, if necessary. For example, this allows you to load a 44.1 kHz impulse response into a project running at 96 kHz, and vice versa.

- */2, /4, /8:* These settings are half-divisions of the preceding value—one-half, one-quarter, one-eighth. For example:
  - If the project sample rate is 96 kHz, the options are 48 kHz, 24 kHz, and 12 kHz.
  - If the project sample rate is 44.1 kHz, the options are 22.05 kHz, 11.025 kHz, and 5512.5 Hz.

  When you select half the sample rate, the impulse response becomes twice as long. The highest frequency that can be reverberated is halved. This results in a behavior that is much like doubling every dimension of a virtual room—multiplying a room's volume by eight. The lower sample rates can also be used for interesting tempo, pitch, and retro digital sounding effects. Another benefit of reducing the sample rate is that processing requirements drop significantly, making half–sample rate settings useful for large, open spaces.

- *To retain the original length of the impulse response when the sample rate is changed:* Click the preserve length button. Manipulating this parameter in conjunction with the sample rate parameter can lead to interesting results.

If you are running Space Designer in a project that uses a higher sample rate than the impulse response, you may also want to reduce the impulse response sample rate. Make sure the preserve length button is turned on. This cuts CPU processing time without compromising reverb quality.

Tip: You can make similar adjustments while running in Synthesized IR mode. Most typical reverb sounds don't have an excessive amount of high frequency content. If your project is running at 96 kHz, for example, you would need to use lowpass filtering to obtain the mellow frequency response characteristics of many reverb sounds. A better approach would be to first reduce the high frequencies by 1/2 or even 1/4 using the "sample rate" slider, followed by using the lowpass filter, thus conserving significant CPU resources.

Set impulse response lengths

- Move the Length parameter to set the length of the impulse response—sampled or synthesized.

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

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

Space Designer envelopes and EQ

Space Designer envelopes and EQ overview
Space Designer's main interface area is used to show and edit envelope and equalizer (EQ) parameters. It consists of the button bar at the top, the main display, and the parameter bar.

- The button bar is used to choose the current view/edit mode.
- The main display shows, and allows you to graphically edit, either the envelope or the EQ curve.
- The parameter bar displays, and allows you to numerically edit, either the envelope or the EQ curve.
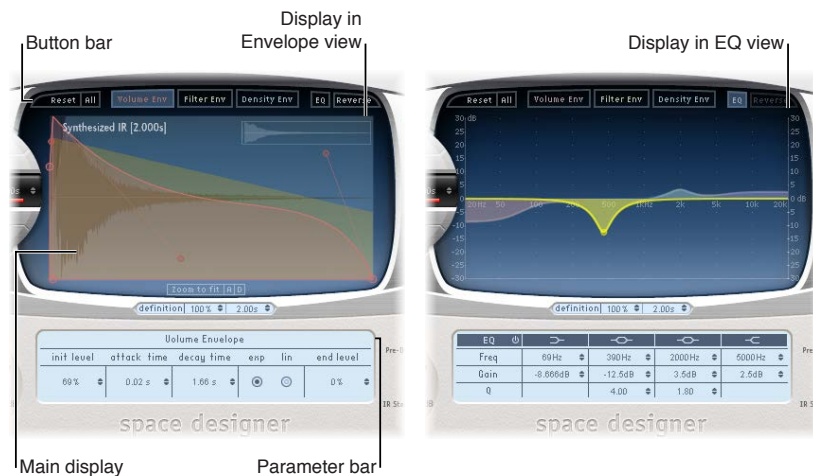
Space Designer button bar
The button bar is used to switch the main display and parameter bar between envelope and EQ views. It also includes buttons that reset the envelopes and EQ or reverse the impulse response.

Button bar parameters
- *Reset button:* Click to reset the currently displayed envelope or EQ to default values.
- *All button:* Click to reset all envelopes and the EQ to default values.
- *Volume Env button:* Click to show the volume envelope in the foreground of the main display. The other envelope curves are shown as transparencies in the background. See Space Designer volume envelope on page 234.
- *Filter Env button:* Click to show the filter envelope in the foreground of the main display. The other envelope curves are shown as transparencies in the background. See Space Designer filter parameters on page 237.
- *Density Env button:* Click to show the density envelope in the foreground of the main display. The other envelope curves are shown as transparencies in the background. See Use impulse responses on page 228.
- *EQ button:* Click to show the four-band parametric EQ in the main display. See Use Space Designer EQ parameters on page 236.

- *Reverse button:* Click to reverse the impulse response and envelopes. When the impulse response is reversed, you are effectively using the tail rather than the front end of the sample. You may need to change the Pre-Dly and other parameter values when reversing.

Edit Space Designer envelope parameters

You can edit the volume and filter envelopes of all impulse responses and the density envelope of synthesized impulse responses. All envelopes can be adjusted both graphically in the main display as well as numerically in the parameter bar.

Whereas some parameters are envelope-specific, all envelopes consist of the Attack Time and Decay Time parameters. The combined total of the Attack Time and Decay Time parameters is equal to the total length of the synthesized or sampled impulse response, unless the Decay time is reduced. See Use impulse responses on page 228.

The large nodes are value indicators of the parameters shown in the parameter bar below the main display—Init Level, Attack Time, Decay Time, and so on. If you edit any numerical value in the parameter bar, the corresponding node moves in the main display.

When displaying envelopes, the main display offers the following zoom and navigation parameters (not shown in EQ view).
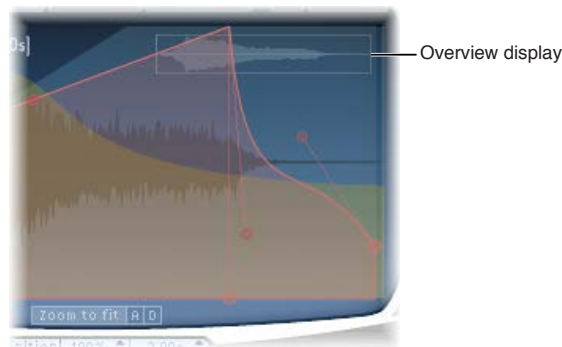
Envelope navigation parameters
- *Overview display:* Indicates which portion of the impulse response file is currently visible in the main display, helping you to orient yourself when zooming.
- *Zoom to Fit button:* Click to display the entire impulse response waveform in the main display. Any envelope length changes are automatically reflected.
- *A and D buttons:* Click to limit the Zoom to Fit function to the attack and decay portions of the currently selected envelope shown in the main display. The A and D buttons are available only when you are viewing the volume and filter envelopes.

Move an envelope node graphically in Space Designer
- Drag the node in one of the available directions.

Two arrows are shown when you move the pointer over any node in the main display, indicating possible movements.

Change Space Designer's envelope curve shape graphically

1 Drag the envelope curve in the main display.

2 Drag the small nodes attached to a line for fine adjustments to envelope curves. These nodes are tied to the envelope curve itself, so you can view them as envelope handles.

Move the nodes vertically or horizontally to change the shape of the envelope curve.
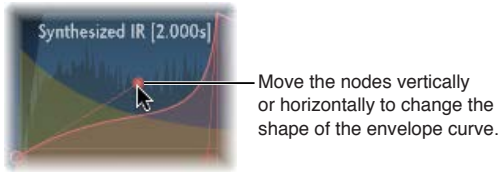
Space Designer volume envelope

The volume envelope is used to set the reverb's initial level and adjust how the volume will change over time. You can edit all volume envelope parameters numerically, and you can also edit many of them graphically (see).
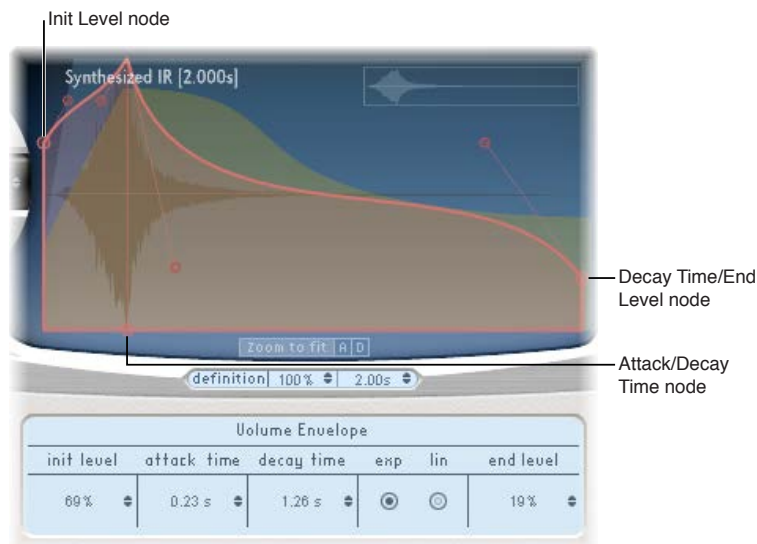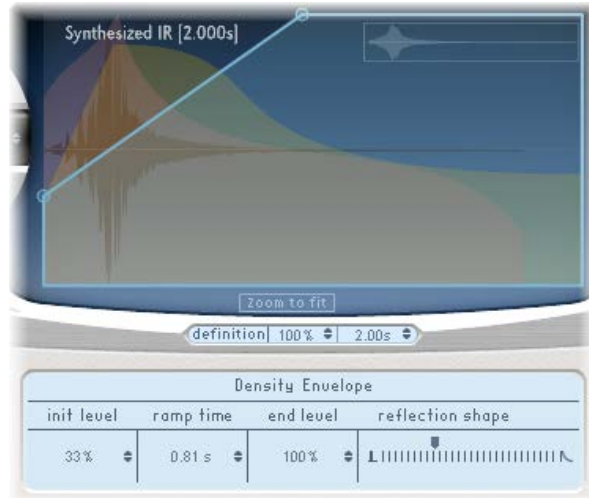

Volume envelope parameters

- *Init Level field:* Sets the initial volume level of the impulse response attack phase. It is expressed as a percentage of the full-scale volume of the impulse response file. The attack phase is generally the loudest point of the impulse response. Set Init Level to 100% to ensure maximum volume for the early reflections.
- *Attack Time field:* Determines the time before the decay phase of the volume envelope begins.
- *Decay Time field:* Sets the length of the decay phase.
- *Volume decay mode buttons:* Set the volume decay curve type.
    - *Exp:* The output of the volume envelope is shaped by an exponential algorithm, to generate the most natural-sounding reverb tail.
    - *Lin:* The volume decay will be more linear, and less natural sounding.
- *End Level field:* Sets the end volume level. It is expressed as a percentage of the overall volume envelope.
    - If set to 0%, you can fade out the tail.
    - If set to 100%, you can't fade out the tail, and the reverb stops abruptly if the end point falls within the tail. (If the end time falls outside the reverb tail, End Level has no effect.)

Space Designer density envelope

The density envelope allows you to control the density of the synthesized impulse response over time. You can adjust the density envelope numerically in the parameter bar, and you can edit the Init Level, Ramp Time, and End Level parameters using the techniques described in Edit Space Designer envelope parameters on page 233.

Note: The density envelope is available only in Synthesized IR mode.


Density envelope parameters

- *Init Level field:* Sets the initial density (the average number of reflections in a given period of time) of the reverb. Lowering the density levels results in audible reflections patterns and discreet echoes.

- *Ramp Time field:* Adjusts the time between the Initial and End Density levels.

- *End Level field:* Sets the density of the reverb tail. If you select an End Level value that is too low, the reverb tail sounds grainy. The stereo spectrum might also be affected by lower values.

- *Reflection Shape slider:* Determines the steepness (shape) of early reflection clusters as they bounce off the walls, ceiling, and furnishings of the virtual space. Small values result in clusters with a sharp contour, and large values result in an exponential slope and a smoother sound. This is handy when recreating rooms constructed of different materials. Reflection Shape, in conjunction with suitable settings for the envelopes, density, and early reflection, will assist you in creating rooms of almost any shape and material.

Use Space Designer EQ parameters

Space Designer has a four-band EQ consisting of two parametric mid-bands plus two shelving filters (one low shelving filter and one high shelving filter). You can edit the EQ parameters numerically in the parameter bar or graphically in the main display.
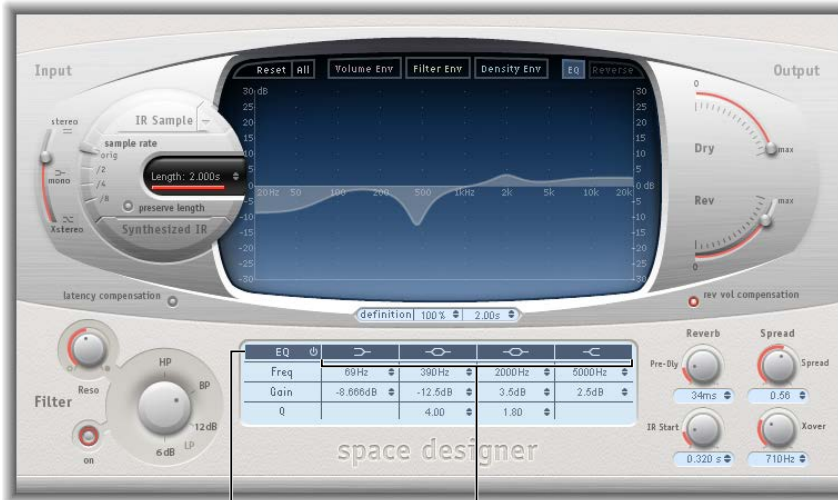

EQ On/Off button   Individual EQ band buttons

- *EQ On/Off button:* Click to turn the entire EQ section on or off.
- *EQ band buttons:* Click to turn individual EQ bands on or off.
- *Frequency fields:* Set the frequency for the selected EQ band.
- *Gain fields:* Cut or boost the selected EQ band.
- *Q fields:* Set the Q factor for the two parametric bands. The Q factor can be adjusted from 0.1 (very narrow) to 10 (very wide).

Graphically edit an EQ curve in Space Designer

1 Enable the EQ and one or more bands with the EQ On/Off and EQ band buttons in the top row of the parameter bar.

2 Move the cursor horizontally over the main display. When the cursor is in the access area of a band, the corresponding curve and parameter area are automatically highlighted and a pivot point is displayed.
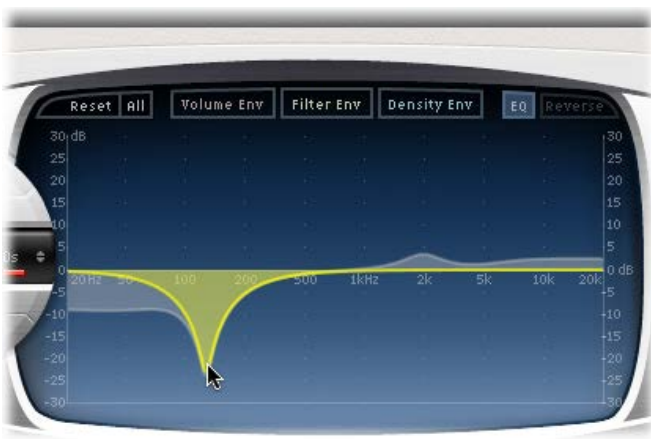
3 Drag horizontally to adjust the frequency of the band.

4 Drag vertically to increase or decrease the Gain of the band.

5 Vertically drag the highlighted pivot point of a parametric EQ band to raise or lower the Q value.

Space Designer filter

Space Designer filter parameters

Space Designer's filter provides control over the timbre of the reverb.

You can select from several filter types and also have envelope control over the filter cutoff, which is independent of the volume envelope. Changes to filter settings result in a recalculation of the impulse response rather than a straight change to the sound as it plays through the reverb.

The main filter parameters are located in the lower-left corner of the interface.
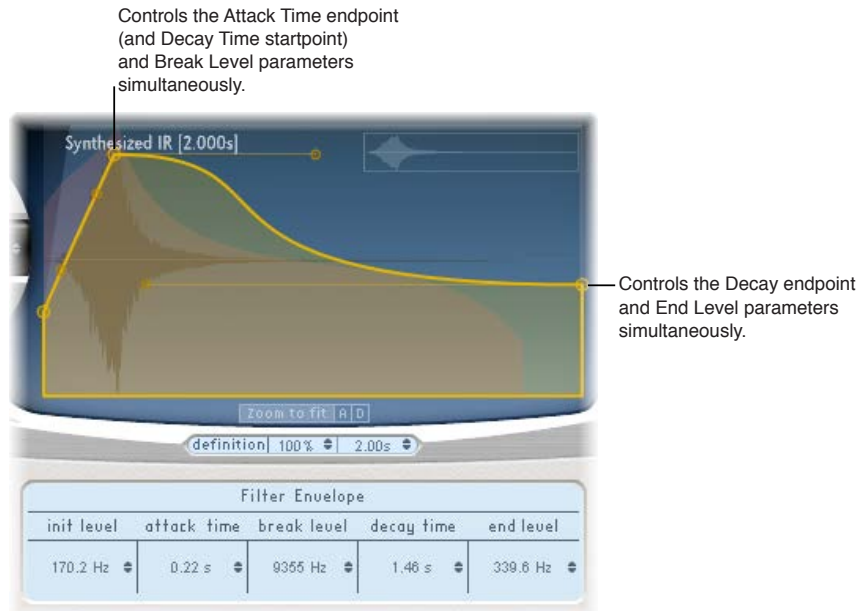
Main filter parameters

- *Filter On button:* Click to switch the filter section on or off.
- *Filter mode knob:* Rotate to set the filter mode.
  - *6 dB (LP):* Bright, general-purpose lowpass filter mode that retains the top end of most material while still providing some filtering.
  - *12 dB (LP):* Warm, lowpass filter mode without drastic filter effects that smooths out bright reverbs.
  - *BP:* 6 dB per octave bandpass design that reduces the low and high ends of the signal, leaving the frequencies around the cutoff frequency intact.
  - *HP:* 12 dB per octave/two-pole highpass design that reduces the level of frequencies that fall below the cutoff frequency.
- *Reso(nance) knob:* Rotate to emphasize frequencies above, around, or below the cutoff frequency. The impact of the resonance knob on the sound is highly dependent on the chosen filter mode, with steeper filter modes resulting in more pronounced tonal changes.

Space Designer filter envelope

The filter envelope appears in the main display when you click the Filter Env button. You can use it to control the filter cutoff frequency over time. You can adjust all filter envelope parameters either numerically in the parameter bar or graphically in the main display using the techniques discussed in Edit Space Designer envelope parameters on page 233.

Note: Activation of the filter envelope automatically enables the main filter.

Controls the Attack Time endpoint (and Decay Time startpoint) and Break Level parameters simultaneously.

Controls the Decay endpoint and End Level parameters simultaneously.

Filter envelope parameters

- *Init Level field:* Sets the initial cutoff frequency of the filter envelope.
- *Attack Time field:* Determines the time required to reach the Break Level.
- *Break Level field:* Sets the maximum filter cutoff frequency that the envelope reaches. This setting also acts as the separation point between the attack and decay phases of the overall filter envelope. In other words, when the selected level has been reached after the attack phase, the decay phase begins. You can create interesting filter sweeps by setting the Break Level to a value lower than the Init Level.
- *Decay Time field:* Determines the time required after the Break Level point to reach the End Level value.
- *End Level field:* Sets the cutoff frequency at the end of the filter envelope decay phase.

Space Designer global parameters

Space Designer global parameters overview

Space Designer's global parameters affect the overall output or behavior of the effect. See Use Space Designer global parameters and Use Space Designer output parameters.

The global parameters are divided into two sections—those around the main display and those below the main display.

Input slider   Latency Compensation button   Definition area   Rev Vol Compensation button
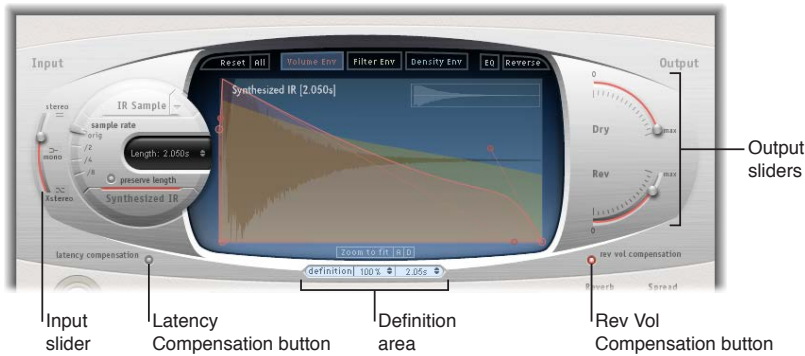
Global parameters (upper)

- *Input slider:* Move to determine how Space Designer processes a stereo input signal.
- *Latency Compensation button:* Click to switch Space Designer's internal latency compensation feature on or off.
- *Definition fields:* Drag to select a less-defined impulse response set, to emulate reverb diffusion and save CPU resources.
- *Rev(erb) Vol(ume) Compensation button:* Click to turn on Space Designer's internal impulse response volume matching function.
- *Output sliders:* Move to adjust output levels.

Global parameters (lower)

- *Pre-Dly knob and field:* Rotate to set the reverb's predelay time, or time between the original signal and the first reflections from the reverb.
- *IR Start knob and field:* Rotate to set the playback start point in the impulse response sample.
- *Spread and Xover knobs and fields:* Rotate the Spread knob to adjust the perceived width of the stereo field, and rotate the Xover knob to set the crossover frequency. Any synthesized impulse response frequency that falls below this value is affected by the Spread parameter. (These knobs are available only for synthesized impulse responses.)

Use Space Designer global parameters

Space Designer's global parameters affect the overall output or behavior of the effect. See Space Designer global parameters overview.

The tasks below cover the use of Space Designer's global parameters.

Use the Space Designer Input slider

The Input slider behaves differently in stereo configurations. (The slider does not appear in mono or mono to stereo instances of the effect.)

- *In stereo instances:* Drag the Input slider to determine how a stereo signal is processed.
  - *Stereo setting (top of slider):* The signal is processed on both channels, retaining the stereo balance of the original signal.
  - *Mono setting (middle of slider):* The signal is processed in mono.
  - *XStereo setting (bottom of slider):* The signal is inverted, with processing for the right channel occurring on the left, and vice versa.
  - *In-between positions:* A mixture of stereo to mono crossfeed signals is produced.

Use Space Designer's latency compensation feature

The complex calculations made by Space Designer take a small amount of time, which results in a processing delay, or *latency*, between the direct input signal and the processed output signal.

Note: This compensation feature is not related to latency compensation in the host application; it occurs *entirely within* Space Designer.

- Click the Latency Compensation button to turn it on, which delays the direct signal in the Output section so that it matches the processing delay of the effect signal.


Space Designer's processing latency is 128 samples at the original sample rate, and it doubles at each lower sample rate division. The processing latency increases to 256 samples if you set Space Designer's sample rate slider to "/2." Processing latency does not increase at sample rates above 44.1 kHz.

Use Space Designer's Definition parameter

The Definition parameter emulates the diffusion of natural reverb patterns. When used at values of less than 100% it also reduces CPU processing requirements.

Natural reverbs contain most of their spatial information in the first few milliseconds. Toward the end of the reverb, the pattern of reflections—signals bouncing off walls, and so on—becomes more diffuse. In other words, the reflected signals become quieter and increasingly nondirectional, containing far less spatial information. To emulate this phenomenon, use the full impulse response resolution only at the onset of the reverb, then use a reduced impulse response resolution toward the end of the reverb.
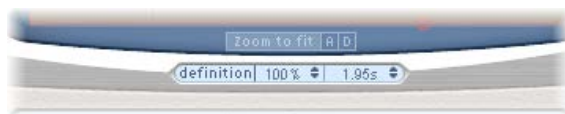
- Drag either of the Definition fields vertically to set the crossover point—where the switch to the reduced impulse response resolution occurs.

  The first Definition field is shown as a percentage, where 100% is equal to the length of the full resolution impulse response. The second field is shown in milliseconds, which indicate the exact crossover point position. (These two fields are linked, so making a change in one automatically changes the other.)

  Note: The Definition fields are visible below the main display only when you have loaded CPU-intensive synthesized impulse responses.

Use Space Designer's reverb volume compensation feature
The reverb volume compensation feature attempts to match the perceived—not the actual—volume differences between impulse response files. The rev vol compensation button is turned on by default and should generally be left in this mode, although it may not work with all types of impulse responses.

- If you have an impulse response that is of a different level, turn off reverb volume compensation and adjust input and output levels accordingly.

Use Space Designer's predelay feature
Predelay is the amount of time that elapses between the original signal and the initial early reflections of the reverberation. For a room of any given size and shape, predelay is determined by the distance between the listener and the walls, ceiling, and floor. Space Designer allows you to adjust this parameter over a greater range than what would be considered natural.

- To set a suitable predelay time, rotate the Pre-Dly knob.

  The ideal predelay setting for different sounds depends on the properties of—or more accurately, the envelope of—the original signal. Percussive signals generally require shorter predelays than signals where the attack fades in gradually, such as strings. A good rule of thumb is to use the longest predelay possible before undesirable side effects, such as an audible echo, begin to materialize.

  In practice, an extremely short predelay tends to make it difficult to pinpoint the position of the signal source. It can also color the sound of the original signal. On the other hand, an excessively long predelay can be perceived as an unnatural echo. It can also divorce the original signal from its early reflections, leaving an audible gap between the original and reverb signals.

  These guidelines are intended to help you design realistic-sounding spaces that are suitable for various signals. If you want to create unnatural sound stages or otherworldly reverbs and echoes, experiment with the Pre-Dly parameter.

Change the impulse response start point
Using the IR Start parameter affords a number of other options that can be quite creative, particularly when combined with the Reverse function. See Space Designer button bar on page 232.

Note: The IR Start parameter is neither available nor necessary in Synthesized IR mode, because the Length parameter provides identical functionality.

- Rotate the IR Start knob to shift the playback start point of the impulse response.

  This effectively cuts off the beginning of the impulse response, which can be useful for eliminating level peaks at the start of the impulse response sample.

Use Space Designer output parameters

Space Designer's global parameters affect the overall output or behavior of the effect. See Space Designer global parameters overview.

The tasks below cover the use of Space Designer's output parameters.

Set Space Designer mono/stereo output parameters

Use the output parameters to adjust the balance between the direct, or *dry*, signals and the processed signals. The parameters that are available depend on Space Designer's input configuration.

Space Designer provides two output sliders—the Dry slider for the direct signal, and the Rev slider for the reverb signal—when you insert it as a mono, mono to stereo, or stereo effect.
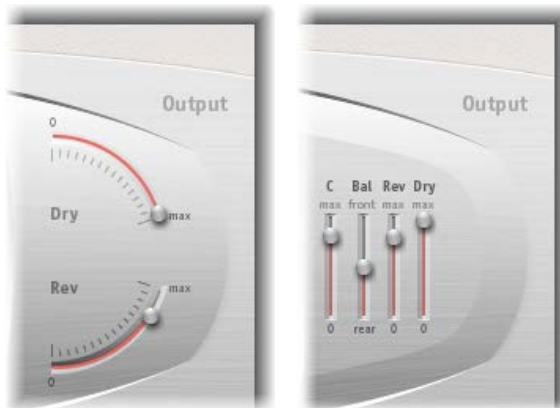
Mono/Stereo   **Surround**

- *To set the level of the Dry slider:* Move to set the level of the non-effect, or dry, signal. Move the slider to a value of 0 (mute) if Space Designer is inserted in a bus channel or you are using modeling impulse responses, such as speaker simulations.
- *To set the level of the Rev(erb) slider:* Move to adjust the output level of the effect, or wet, signal.

Use the Space Designer Spread parameters

The Spread and Xover (crossover) knobs enhance the perceived width of the signal without losing the directional information of the input signal normally found in the higher frequency range. Low frequencies are spread to the sides, reducing the amount of low frequency content in the center—allowing the reverb to encompass the mix.

Note: The Spread and Xover knobs function *only* in Synthesized IR mode.

- *To set the level of the Spread knob and field:* Rotate to extend the stereo base to frequencies that fall below the frequency determined by the Xover parameter.
  - At a Spread value of 0.00, no stereo information is added. (The inherent stereo information of the source signal and reverb, however, is retained.)
  - At a Spread value of 1.00, the left and right channel divergence is at its maximum.
- *To set the level of the Xover knob and field:* Rotate to set the crossover frequency in Hertz. Any synthesized impulse response frequency that falls below the value you set will be affected by the Spread parameter (at values over 0).

Note: Because these parameters adjust stereo processing, they have *no* impact when you use Space Designer as a mono plug-in.

Specialized effects and utilities

14

Specialized effects overview

MainStage includes a bundle of specialized effects and utilities designed to address tasks often encountered during audio production:

- Denoiser eliminates or reduces noise below a threshold level.
- Exciter adds life to your recordings by generating artificial high frequency components.
- Grooveshifter creates rhythmic variations in your recordings.
- Speech Enhancer improves speech recordings made with your computer's internal microphone.
- SubBass generates an artificial bass signal that is derived from the incoming signal.

Denoiser

Denoiser overview

Denoiser eliminates or reduces any noise below a threshold volume level. It does this by using Fast Fourier Transform (FFT) analysis to identify frequency bands of lower volume and less complex harmonic structure, which it then reduces to the defined decibel level.

If you use Denoiser too aggressively, you may encounter artifacts that are less pleasant than the existing noise. Use the three Smoothing knobs to reduce or eliminate these artifacts. See Denoiser smoothing parameters.
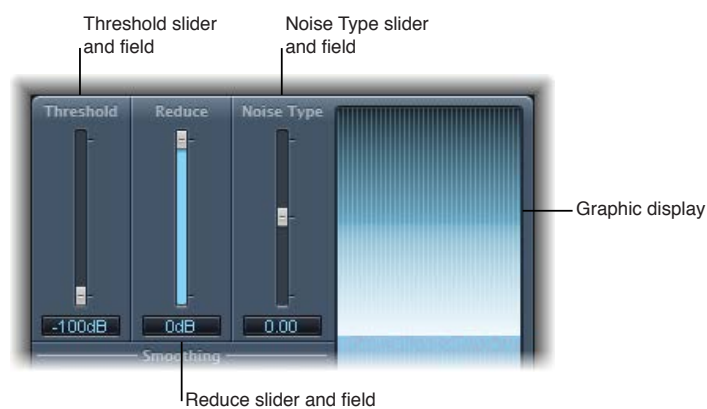
Denoiser main parameters

- *Threshold slider and field:* Drag to set the threshold level below which the noise signals are reduced.

  Locate a section of the audio where only noise is audible, then set the Threshold slider to a dB value that filters only signals at or below this level.

- *Reduce slider and field:* Drag to set the amount of noise reduction applied to signals that fall below the threshold. Aim for a Reduce slider value where noise reduction is optimal but little of the music or vocal signal is reduced. Each 6 dB reduction halves the volume level, and each 6 dB increase doubles it.

  Note: If the noise level of your recording is very high (more than −68 dB), reducing it to a level of −83 to −78 dB should suffice, provided no audible side effects are introduced. This reduces the noise by more than 10 dB, to less than half its original volume.

- *Noise Type slider and field:* Drag to determine the type of noise you want to reduce.
  - A value of 0 equals white noise—equal frequency distribution.
  - Positive values change the noise type to pink noise—harmonic noise; greater bass response.
  - Negative values change the noise type to blue noise—hissy tape noise.

- *Graphic display:* Shows how the lowest volume level signals in your audio material, which are mostly or entirely noise, are reduced.

Denoiser smoothing parameters

Denoiser has the following smoothing parameters:
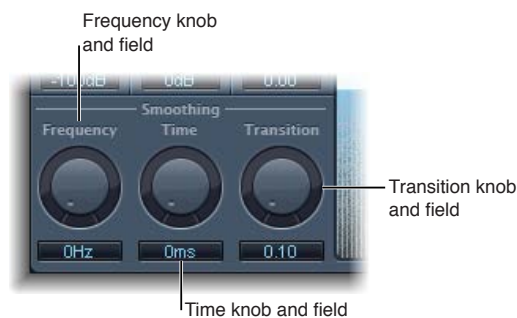
Frequency knob and field

Transition knob and field

Time knob and field

Denoiser smoothing parameters

- *Frequency knob and field:* Rotate to adjust how smoothing is applied to neighboring frequencies. If Denoiser recognizes that only noise is present on a certain frequency band, use the Frequency parameter to smooth the neighboring frequencies to avoid artifacts. The higher you set the Frequency parameter, the more the neighboring frequency bands are changed.

- *Time knob and field:* Rotate to set the time required to reach maximum noise reduction. This is the simplest form of smoothing.

  Note: The Time parameter also sets a release time, which is the time required for the signal to revert to its normal level from the maximum noise reduction level. As with all Denoiser parameters, the Threshold value determines the level that triggers the noise reduction process.

- *Transition knob and field:* Rotate to adjust how smoothing is applied to neighboring volume levels. If Denoiser recognizes that only noise is present in a certain volume range, use the Transition parameter to smooth the neighboring volume levels to avoid artifacts. The higher you set the Transition parameter, the more similar-level values are changed.

Exciter

Exciter generates high frequency components that are not part of the original signal. It does this by utilizing a nonlinear distortion process that resembles the one used to produce overdrive and distortion effects.

Unlike this process, however, Exciter's distortion process involves passing the input signal through a highpass filter before feeding it into the harmonics (distortion) generator. Artificial harmonics are thus added to the original signal, and 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 Exciter to add life to recordings, particularly audio tracks with a weak treble frequency range. You can also use Exciter to enhance guitar tracks.

Exciter parameters

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

  Note: In most cases, it is preferable to select higher Frequency and Harmonics values, 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, and Color 2 generates a more intense harmonic distortion.

  Note: Color 2 also introduces more intermodulation distortions, which can result in unpleasant artifacts.

Grooveshifter

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. Grooveshifter can make this straightforward solo swing.

Grooveshifter automatically follows all changes to the project tempo, which it uses as the reference tempo.

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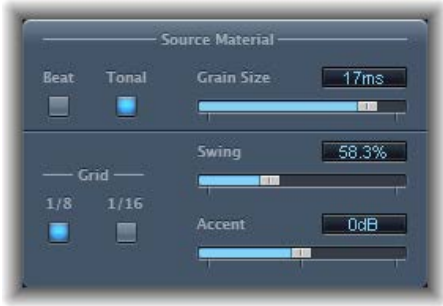
Grooveshifter source material parameters

- *Beat and Tonal buttons:* Click to select the type of source, or input, material you are using.
  - *Beat button:* The beat algorithm is optimized for percussive input material. The Grain Size slider is disabled when you select Beat.
  - *Tonal button:* The tonal algorithm is optimized for tonal input material. Because this algorithm is based on granular synthesis, the Grain Size slider is available when you select Tonal.
- *Grain Size slider and field:* Drag to set the size of the grains—from 1 ms to 20 ms. Technically, this determines the analysis precision. The default Auto setting at the left end of the slider automatically assigns a suitable grain size value based on the incoming signal.

Grooveshifter swing parameters

- *Grid buttons:* Click to set the beat division used as a timing reference when analyzing audio material.
  - *1/8 button:* Select if the audio material contains primarily eighth notes.
  - *1/16 button:* Select if the audio material consists mostly of sixteenth notes.
- *Swing slider and field:* Drag to set the amount that even beats are delayed—from 50% to 75%. A value of 50% means there is 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:* Drag to set the level of even beats—from –12 dB to +12 dB— suppressing or 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.

Speech Enhancer parameters

- *Denoise slider and field:* Drag to determine the noise floor in the recording—from –60 dB to –20 dB—and thus the amount of noise reduction required. Settings toward –60 dB allow more noise to pass; settings toward –20 dB increasingly suppress background noise but also proportionately increase artifacts.

- *Mic Correction Off/On buttons:* Click the On button to improve the frequency response of recordings made with your built-in microphone, thus creating the impression that an expensive microphone was used.

- *Mic Model pop-up menu:* Choose a microphone model to compensate for tonal characteristics of specific built-in Mac microphones.

  Note: You can use the Speech Enhancer effect with other microphones as well, but microphone correction models are provided only for built-in Mac and iSight microphones. If you are using a non-Apple microphone, choose Generic from the Mic Correction pop-up menu.

- *Voice Enhance Off/On buttons:* Click to turn the Voice Enhance multiband compression circuit off or on.

- *Enhance Mode pop-up menu:* When Voice Enhance is active, choose an appropriate setting to make the recorded voice louder and more intelligible:
  - *(Female or Male) Solo:* Choose when the recorded signal consists of a vocal only.
  - *(Female or Male) Voice Over:* Choose when the recorded signal contains both a vocal performance and a musical or atmospheric bed.

SubBass

SubBass overview

SubBass generates frequencies below those of the original signal, resulting in artificial bass content. The simplest use for 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.

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 on page 249.

> *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

SubBass offers the following parameters.

SubBass parameters

- *High Ratio knob and field:* Rotate to adjust the ratio between the generated signal and the upper frequency band of the original signal.
- *High Center knob and field:* Rotate to set the center frequency of the upper frequency band.
- *High Bandwidth knob and field:* Rotate to set the width of the upper frequency band.
- *Graphic display:* Shows the selected upper and lower frequency bands.
- *Freq. Mix slider and field:* Drag to adjust the mix ratio between the upper and lower frequency bands.
- *Low Ratio knob and field:* Rotate to adjust the ratio between the generated signal and the lower frequency band of the original signal.

- *Low Center knob and field:* Rotate to set the center frequency of the lower frequency band.
- *Low Bandwidth knob and field:* Rotate to set the width of the lower frequency band.
- *Dry slider and field:* Drag to set the amount of dry (non-effect, original) signal.
- *Wet slider and field:* Drag to set the amount of wet (effect) signal.

SubBass use tips

Unlike a pitch shifter, SubBass generates a waveform that is not based on the waveform of the input signal; instead it uses a sine wave. Given that pure sine waves rarely work well in complex arrangements, make sure to use the Wet and Dry sliders to control the amount of—and balance between—the generated and original signals.

Use the High parameters and the Low parameters to define the two frequency bands that 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, a Ratio value of 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 minimize frequency intermodulations which can lead to unpleasant artifacts. Set the High Center knob value a fifth higher than Low Center, 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, using a Ratio of 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 over-enhance the low end of some tracks, resulting in an unbalanced mix.

Utilities and tools

15

Utilities and tools overview

The tools found in the Utility category can help with routine tasks and situations you may encounter during production. Examples include the Gain plug-ins, which you can use to adjust the level or phase of input signals, and I/O Utility, which you can use to integrate external audio effects into your host application mixer.

Gain plug-in

Gain amplifies (or reduces) the signal by a specific decibel amount. It is very useful for quick level adjustments when you work 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 plug-in parameters

- *Gain slider and field:* Drag to set the amount of gain.
- *Phase Invert Left and Right buttons:* Turn on to invert the phase of the left and right channels, respectively.

  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.

- *Balance knob and field:* Rotate to adjust the balance of the incoming signal between the left and right channels.
- *Swap L/R (left/right) button:* Turn on to swap the left and right output channels. The swapping occurs after the Balance parameter in the signal path. The Swap L/R button is disabled when the Mono button is turned on.
- *Mono button:* Turn on to output 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. Only one Phase Invert button is available in mono and mono to stereo modes. In mono mode, the Balance, Swap L/R, and Mono parameters are disabled as well.

Use I/O utility

I/O utility enables you to use external audio effects units, similar to using internal effects.

Note: I/O utility is not practical unless you are using an audio interface that provides discrete inputs and outputs, either analog or digital, that are used to send signals to and from the external audio effects unit.
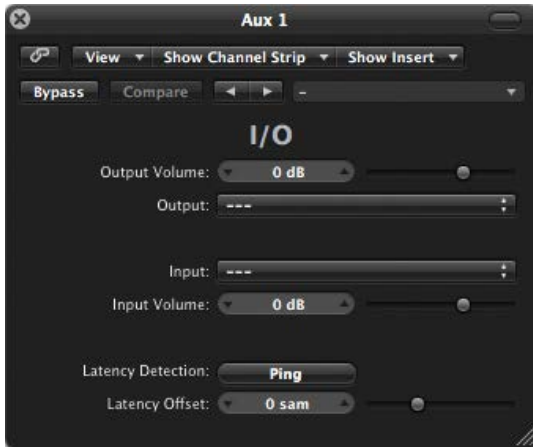
I/O utility parameters

- *Output Volume field and slider:* Drag to adjust the level of the output signal.
- *Output pop-up menu:* Choose the output, or output pair, of your audio hardware.
- *Input pop-up menu:* Choose the input, or input pair, of your audio hardware.

  Note: The Input pop-up menu is visible only when an audio interface with multiple inputs is active.

- *Input Volume field and slider:* Drag to adjust the level of the input signal.
- *Latency Detection (Ping) button:* Click to detect the delay between the selected output and input. Following detection, any delay is automatically compensated for.

  Note: You can obtain the most accurate reading by bypassing any latency-inducing plug-ins on the track.

- *Latency Offset field and slider:* Displays the value for the detected latency between the selected output and input in samples. You can also use this slider to offset the latency manually.

Use an external effects unit with I/O utility

1 Connect an output of your audio interface with the input on your effects unit, and connect the output of your effects unit with an input on your audio interface.

  Note: These can be either analog or digital connections, depending on the features of your audio interface and effects unit, and each connection can be either an output or an output pair.

2 Click an Insert slot of an aux channel strip that is being used as a bus send/return, and choose Utility > I/O.

3 In the I/O utility window, choose the Outputs and the Inputs of the 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 Volume and Output Volume sliders as required in the I/O utility 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 are processed by the external effects unit.

Test Oscillator

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, which is the 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, which is activated by clicking the Sine Sweep button, Test Oscillator generates a user-defined frequency spectrum tone sweep when triggered with the Trigger button.

Test Oscillator parameters

- *Waveform buttons:* Select the waveform to use 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; the Needle Pulse waveform is a single needle impulse waveform.

  Note: The Waveform buttons are disabled when the Sine Sweep button is turned on.

- *Frequency knob and field:* Rotate to set the frequency of the oscillator (the default is 1 kHz).

- *Sine Sweep button:* Turn on to generate a sine wave sweep of the frequency spectrum you set with the Start Freq and End Freq fields.

- *Time field:* Drag to set the duration of the sine wave sweep.

- *Start Freq(uency) and End Freq(uency) fields:* Drag to set the oscillator frequency for the beginning and end of the sine sweep.

- *Sweep Mode pop-up menu (Extended Parameters area):* Choose a type of sweep curve—Linear or Logarithmic.

- *Trigger button:* Turn on to start the sine sweep.

- *Trigger pop-up menu:* Choose the sine sweep mode:
  - *Single:* Triggers the sweep once.
  - *Continuous:* Triggers the sweep indefinitely.

- *Level slider and field:* Drag to set the overall output level.

Legacy effects

Legacy effects overview

Legacy effects are included for project compatibility. These plug-ins are inserted when you load a project (that contains these plug-ins) created with an older MainStage version.

You can use these plug-ins or you can replace them with other effect plug-ins available in MainStage.

You cannot directly insert these plug-ins in MainStage unless you override the effects plug-in menu.

Display and insert legacy plug-ins in MainStage

1 Press Option, then click a plug-in slot on a channel strip.

The plug-in menu opens, with a Legacy submenu shown below the Utility plug-in submenu.

2 Choose the legacy plug-in that you want to insert from the Legacy submenu.

AVerb

AVerb is a basic 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.


AVerb parameters
- *Predelay slider and field:* Drag to determine the time between the source signal and the early reflections of the reverb signal.
- *Reflectivity knob and field:* Rotate to define how reflective the imaginary walls, ceiling, and floor are. In other words, how hard the walls are and what they are made of. Glass, stone, timber, carpet, and other materials have a dramatic impact on the tone of the reverb.
- *Room Size knob and field:* Rotate to define the dimensions of simulated rooms.
- *Density/Time slider and field:* Drag to determine both the density and duration of the reverb.
  - Low values generate clearly discernible early reflection clusters, resulting in an echo.
  - High values result in a more reverb-like effect.
- *Mix slider and field:* Drag to set the balance between the effect (wet) and direct (dry) signals.

Bass Amp

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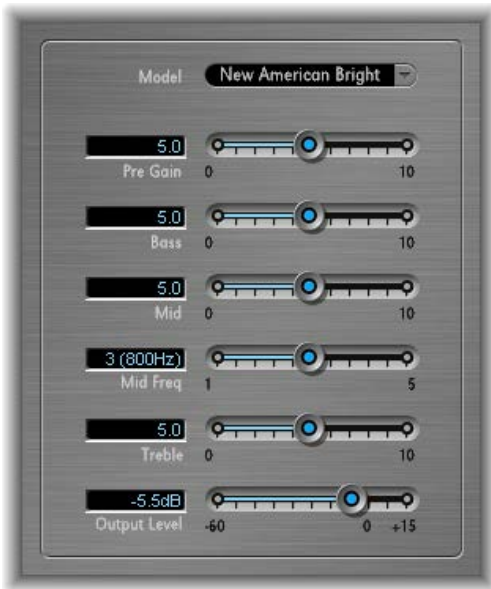

Bass Amp Parameters

- *Model pop-up menu:* Choose one of the following amplifier models:
  - *American Basic:* 1970s-era American bass amp, equipped with eight 10" speakers. Suitable for blues and rock recordings.
  - *American Deep:* Based on the American Basic amp, but with strong lower-mid frequency (from 500 Hz on) emphasis. Suitable 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. Suitable 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, suitable for blues and rock recordings.
  - *New American Bright:* Based on the New American Basic amp, this model emphasizes the frequency range above 2 kHz. Suitable for rock and heavy metal.
  - *Top Class DI Warm:* Famous DI box simulation, suitable 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 suitable 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.

EQ

DJ EQ

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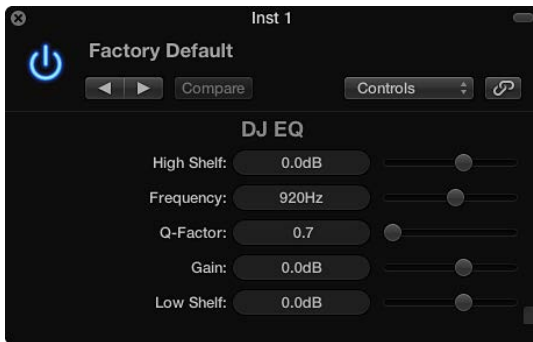
DJ EQ parameters
- *High Shelf slider and field:* Drag to set the amount of gain for the high shelving filter.
- *Frequency slider and field:* Drag to set the center frequency of the parametric EQ.
- *Q-Factor slider and field:* Drag to set the range (bandwidth) of the parametric EQ.
- *Gain slider and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

Fat EQ

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

Fat EQ parameters

- *Band Type buttons:* For bands 1, 2, 4, and 5, click one of the paired buttons to select the EQ type. Band 3 is parametric.
    - *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. The scale is shown in dB.
- *Frequency fields:* Drag to set the frequency for each band.
- *Gain knobs and fields:* Rotate to set the amount of gain for each band.
- *Q fields:* Drag to set 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 are working with a narrow frequency band, you will usually 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:* Turn the corresponding band on or off.
- *Master Gain slider and field:* Drag to set the overall output level of the signal. Use it after boosting or cutting individual frequency bands.

Single-Band EQs

The single-band EQs are used for different types of equalization tasks.

- *High Cut or Low Cut:* High Cut attenuates the frequency range above the selected frequency. Low Cut attenuates the frequency range that falls below the selected frequency.

- *High Pass or Low Pass Filter:* High Pass Filter affects the frequency range below the set frequency. Higher frequencies pass through the filter. You can use High Pass Filter to eliminate the bass below a selectable frequency. Low Pass Filter affects the frequency range above the selected frequency.

- *High Shelf or Low Shelf EQ:* Low shelving EQ affects only the frequency range that falls below the selected frequency. High shelving EQ affects only the frequency range above the selected frequency.

- *Parametric EQ:* 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.

High Cut and Low Cut parameters

- *Frequency slider and field:* Drag to set the cutoff frequency.

High Pass and Low Pass parameters

- *Frequency slider and field:* Drag to set the cutoff frequency.

- *Order slider and field:* Drag to set the filter order. The more orders used, the stronger the filtering effect.

- *Smoothing slider and field:* Drag to adjust the amount of smoothing, in milliseconds.

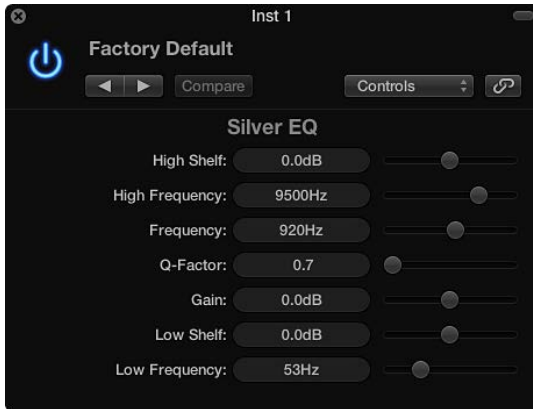### High and Low Shelving EQ parameters

- *Gain slider and field:* Drag to set the amount of cut or boost.

- *Frequency slider and field:* Drag to set the cutoff frequency.

Parametric EQ parameters

- *Gain slider and field:* Drag to set the amount of cut or boost.

- *Frequency slider and field:* Drag to set the cutoff frequency.

- *Q-Factor slider and field:* Drag to set the Q (bandwidth).

Silver EQ

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 factor of the parametric EQ.

Silver EQ parameters

- *High Shelf slider and field:* Drag to set the level of the high shelving EQ.
- *High Frequency slider and field:* Drag to set the cutoff frequency for the high shelving EQ.
- *Frequency slider and field:* Drag to set the center frequency of the parametric EQ.
- *Q-Factor slider and field:* Drag to set the range (bandwidth) of the parametric EQ.
- *Gain slider and field:* Drag to set the amount of cut or boost for the parametric EQ.
- *Low Shelf slider and field:* Drag to set the level of the low shelving EQ.
- *Low Frequency slider and field:* Drag to set the cutoff frequency for the low shelving EQ.

GoldVerb

GoldVerb overview

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

GoldVerb is divided 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 on page 261.
- *Reverb parameters:* Control the diffuse reverberations. See GoldVerb reverb parameters on page 262.
- *Balance ER/Reverb slider:* Drag to set the balance between the early reflections and the 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 provides the following Early Reflections parameters.

GoldVerb early reflections parameters

- *Predelay slider and field:* Drag to set the 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.

    - *Very 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 side effects, such as an audible echo. When you reach this point, reduce the Predelay setting slightly.

- *Room Shape slider and field:* Drag to define the geometric form (the shape) 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:* Drag to determine the dimensions of the room. The numeric value indicates the length of the room's walls—the distance between two corners.

- *Graphic display:* Shows changes to Room Size and Room Shape parameters.

- *Stereo Base slider and field:* Drag to set the distance between the two virtual microphones that capture the simulated room signal.

    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

GoldVerb provides the following reverb parameters.

GoldVerb reverb parameters

- *Initial Delay slider and field:* Drag to set the time between the original signal and the diffuse reverb tail. If you are trying to attain 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:* Drag to control the width of the reverb's stereo image. At 0%, the effect generates a monaural reverb. At 200%, the stereo base is artificially expanded.

- *High Cut knob and field:* Rotate to filter frequencies above the set value 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 to its maximum value, the reverb will sound as if it is reflecting off stone or glass.

- *Density knob and field:* Rotate to control 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:* Rotate to set the 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.
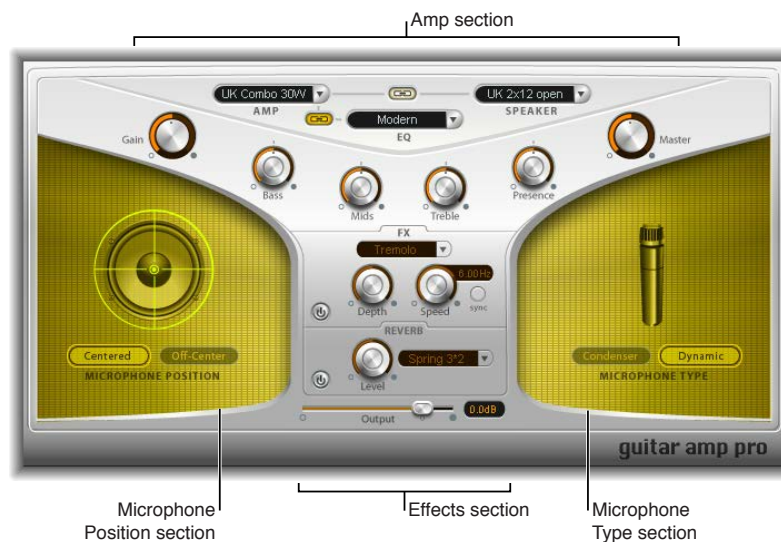
Guitar Amp Pro

Guitar Amp Pro overview

Guitar Amp Pro simulates 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 use it with other instruments, applying the sonic character of a guitar amp to a trumpet or vocal part, for example.

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. Guitar Amp Pro also emulates classic guitar amplifier effects, including reverb, vibrato, and tremolo.

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

The Guitar Amp Pro window is organized into several parameter sections.

- *Amp section:* The model parameters at the top are used to choose the type of amp, EQ model, and speaker. See Guitar Amp Pro amplifier models, Guitar Amp Pro cabinet models, and Guitar Amp Pro EQ. The knobs in the V-shaped section are used to set tone, gain, and level. See Guitar Amp Pro amplifier models on page 264.
- *Effects section:* Provides parameters to control the built-in tremolo, vibrato, and reverb effects. See Guitar Amp Pro effects on page 267.
- *Microphone Position and Type sections:* These sections enable you to set the position and type of the microphone. See Guitar Amp Pro microphone parameters on page 268.
- *Output slider:* 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.

Guitar Amp Pro amplifier models

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

Amp models

- *UK Combo 30W:* Neutral-sounding amp, suitable for clean or crunchy rhythm parts.
- *UK Top 50W:* Quite aggressive in the high frequency range, suitable for classical rock sounds.
- *US Combo 40W:* Clean sounding amp model, suitable 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 suitable 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).

Guitar Amp Pro cabinet models

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 cabinet parameters

- *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 suitable 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 bypasses 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.

Guitar Amp Pro EQ

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

EQ parameters

- *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 re-create a well-known guitar sound. You can combine any speaker or EQ model with any amp by turning off the two Link buttons.

Guitar Amp Pro amplifier 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.

Amplifier parameters

- *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.

Guitar Amp Pro effects

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 and vibrato parameters

- *On/off button:* Turns the tremolo/vibrato effect on or off.
- *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.

Reverb parameters

- *On/off button:* Turns the reverb effect on or off.
- *Reverb pop-up menu:* Choose one of the three types of spring reverb.
- *Level knob:* Rotate to set the amount of reverb applied to the pre-amplified amp signal.

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 also choose to vary any other parameters.

Silver Compressor

Silver Compressor is a simplified version of the Compressor plug-in. See Use Compressor on page 85.

Silver Compressor parameters

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

Silver Gate

Silver Gate is a simplified version of the Noise Gate plug-in. See Use Noise Gate on page 97.

Silver Gate parameters

- *Lookahead slider and field:* Drag to set how far ahead Silver Gate analyzes the incoming signal, allowing it to respond more quickly to peak levels.
- *Threshold slider and field:* Drag to set the threshold level. Signals that fall below the threshold are reduced in level.
- *Attack knob and field:* Rotate to set the time it takes to fully open the gate after the signal exceeds the threshold.
- *Hold knob and field:* Rotate to set the time the gate remains open after the signal falls below the threshold.
- *Release knob and field:* Rotate to set the time it takes to fully close the gate after the signal falls below the threshold.