Fender®

PLUG-INS

USER GUIDE

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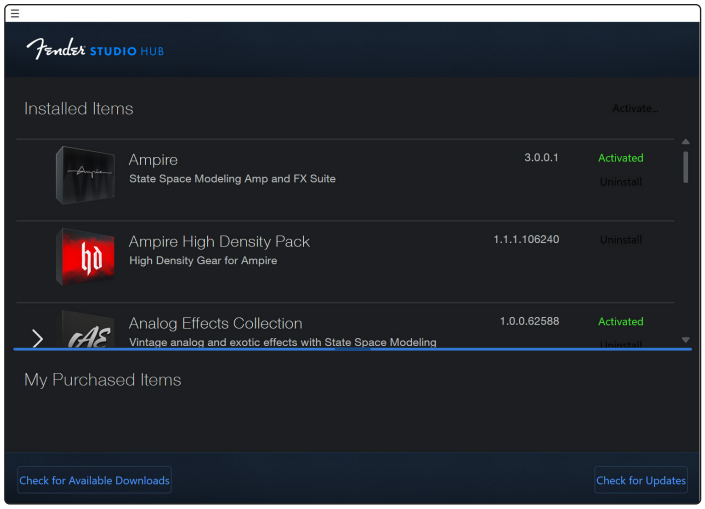
1 Introduction

Thank you for purchasing a Fender® plug-in. Originally designed for Fender Studio Pro, Fender plug-ins offer state-of-the-art sound quality and functionality in Fender Studio Pro Native Effects™, VST3, AU, and AAX formats. This user manual covers the features and functions of Ampire, Fat Channel XT, Fender Virtual Instruments, and the PreSonus Channel Strip collection, as well as the Fat Channel Add-on packs and installation using Fender Studio Hub.

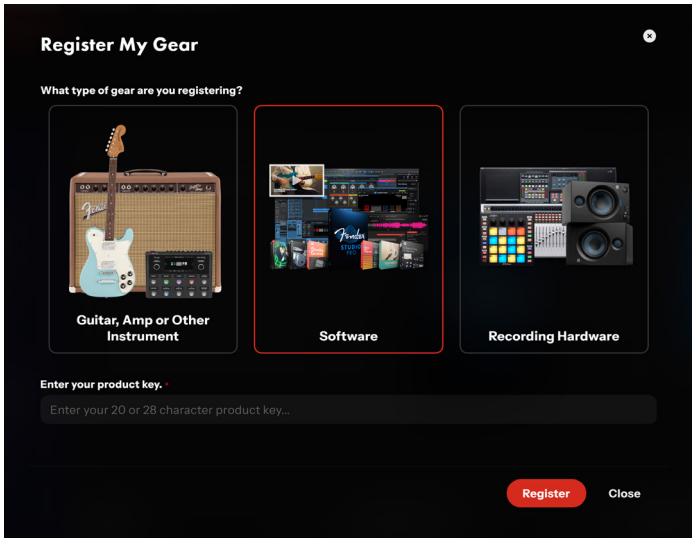
1.1 Fender Studio Hub



Fender Studio Hub is a universal manager application for all Fender and PreSonus plug-ins. After you purchase any Fender plug-in, you will find the Fender Studio Hub application in your MyFender user account. Please follow these instructions to register, download, install, and activate your Fender plug-ins.

1.2 Registration

When you purchase a plug-in on the Fender online shop, the product will automatically be added to your MyFender account. If you purchased your product from your favorite Fender retailer, please log into your MyFender account, click "Register new product..." and enter the Product Key to your account:



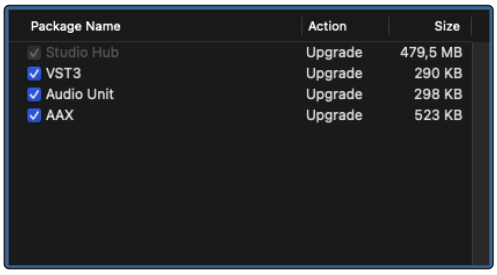
1.3 Installation and Activation

The easiest way to install and activate your plug-ins is using the Fender Studio Hub application. Fender Studio Hub is available as a free download from MyFender. As soon as a Fender plug-in is registered to your MyFender account, Fender Studio Hub will be automatically added as well.

Note: *If you're running Fender Studio Pro on the same computer, be sure to quit Fender Studio Pro before running Fender Studio Hub to install new plug-ins.*

1. Download and install Fender Studio Hub.

2. During the installation you will be asked to select what formats you would like to install for your Fender plug-ins. Select all that apply. Check your DAW or host applications if you are unsure which format is supported.

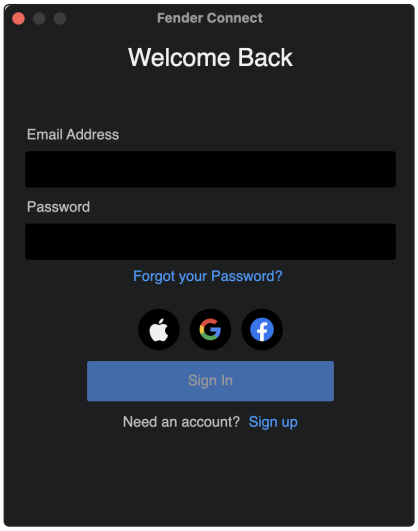


Package Name	Action	Size
Studio Hub	Upgrade	479,5 MB
VST3	Upgrade	290 KB
Audio Unit	Upgrade	298 KB
AAX	Upgrade	523 KB

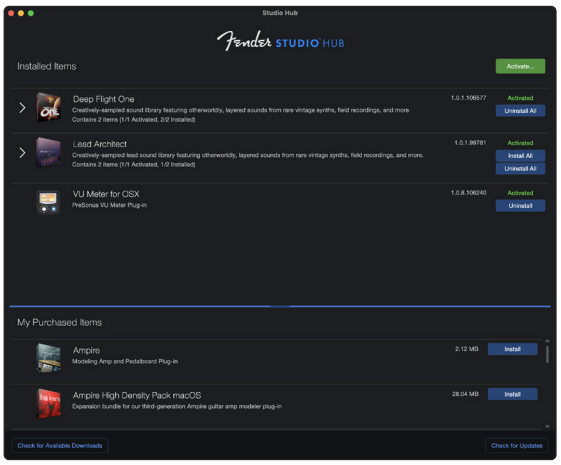
3. Launch Fender Studio Hub.

4. Click Authorize to activate Hub on your computer.

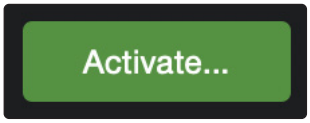
5. Log in with your MyFender credentials.



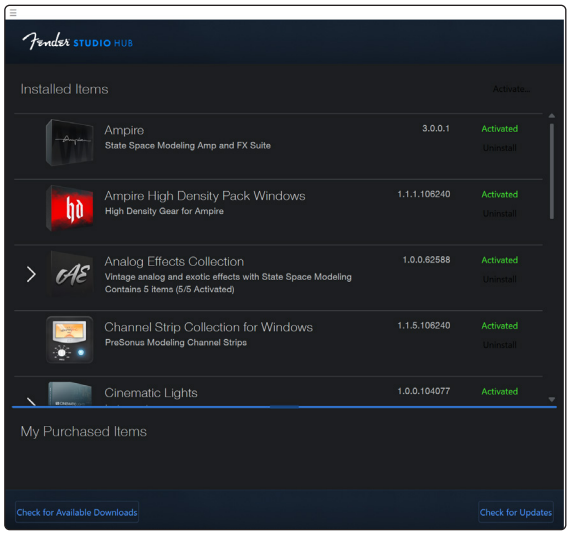
6. Once Fender Studio Hub has been authorized, you will find your plug-ins under "My Purchased Items." Click Install for each plug-in you would like to install on your computer. This will download and install the latest version from MyFender.



7. Click Activate.

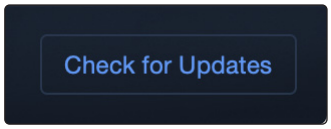


Once installed and activated, your plug-ins will be listed in the Installed Items section.

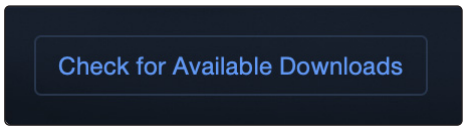


Power User Tip: *If you ever need to uninstall and remove a plug-in from your computer, launch Fender Studio Hub and click Uninstall next to the plug-in.*

Use Fender Studio Hub to keep your Fender plug-ins up to date. Simply click the Check for Updates button to verify that the latest version is installed on your computer.



If your Fender plug-in is not displayed in the My Purchased Items section, click the Check for Available Downloads button.



Fender plug-ins can also be installed by downloading the installation package from your MyFender account. Launch Fender Studio Hub and drag the .install file to the Fender Studio Hub application window.

2 Fender Plug-ins

2.1 Analog Effects Collection

Analog Effects Collection is a collection of five modulation, delay, distortion, and dynamics effects available for the first time in VST3, AU and AAX format—previously available only in Fender Studio Pro!

Analog Chorus, Analog Delay, Rotor, Red Light Distortion and Tricomp represent the pinnacle of Fender plug-in design, art, and craftsmanship. With the addition of State Space Modeling technology and a brand new look and feel, the Analog Effects Collection is the perfect complement to any DAW.

2.1.1 Chorus

Chorus is a 1-3 voice chorus processor with optional LFO delay time modulation and stereo width control. It offers a wide range of effects—from subtle to extreme—with its roots in the 70s era of analog guitar pedals and studio effects processors.

- **Delay.** Adjusts the delay of the Chorus voices. The value you set is the delay time between voices.
- **LFO Shape.** Choose between the four waveforms for the LFO: Triangle, Sine, Sawtooth, or Square.
- **Voices.** Adjusts the number of added voices in the Chorus; select from 1, 2, or 3.
- **LFO.** The LFO modulates the Spacing parameter.
- **Depth.** Adjusts the mix of the processed Chorus output with the dry input signal. Variable from 0 to 100%.
- **St. Width.** Adjusts the spreading of the Chorus voices in the stereo field. Click the Spread button to engage/disengage the Stereo Width feature.
- **LFO Speed.** Adjusts the speed of the LFO.
- **LFO Width.** Adjusts the range of the LFO modulation of Spacing. Variable from 0 to 100%. A value of 100%would modulate the Spacing parameter from 0 to 2x Spacing.
- **Low Freq.** Sets the corner frequency of the low-cut filter.
- **High Freq.** Sets the corner frequency of the high-cut filter.
- **Mode.** Choose between Doubler mode and Chorus mode, which employs inverse all-pass movement, for truer chorus effect.

2.1.2 Analog Delay

Analog Delay emulates a BBD with optional tempo sync, LFO, filtered feedback, and other features. It can be used to create deteriorating echoes, echoes with changing delay and pitch, and flanging/chorus effects. These types of sounds are often used in dub music or 70s rock.

2.1 Analog Effects Collection

The addition of a State Space Modeling based Drive stage adds analog warmth and saturation to the signal for even more tonal control.

- **Delay**
  - » **Time.** The base delay time.
  - » **Sync.** Optional Sync mode for Time. Sync disengaged results in speed expressed as time from 1 ms to 3 s. Sync engaged results in time expressed as beats from 4/1 to 1/64, with triplets.
  - » **Feedback.** Feedback percentage; that is, the amount of delayed signal to be fed back into the delay input. Variable from 0 to 100%.
  - » **Boost.** Enable this parameter to boost feedback levels.
- **LFO**
  - » **Speed.** The base LFO speed.
  - » **Sync.** Optional Sync mode for LFO Speed. Sync disengaged results result in Speed expressed as frequency, from 0.01 Hz to 5 Hz. Sync engaged results in Speed expressed as beats, from 4/1 to 1/64, with triplet and dotted-time variants.
  - » **Amount.** Modifies the effect of the LFO on delay speed. Variable from -50%to 50%.
  - » **Type.** The shape of the LFO waveform; select from triangle, sine, sawtooth, and square.
- **Color**
  - » **Low Cut.** Filters frequencies below this value from the delayed signal. Variable from Off to 20 Hz to 3.2 kHz. Filter is 6 dB per octave.
  - » **High Cut.** Filters frequencies above this value from the delayed signal. Variable from 400 Hz to 16 kHz to Off. Filter is 6 dB per octave.
  - » **Drive.** Emulates tape saturation with State Space Modeling. Variable percentage from 0 to 100%.
- **Motor**
  - » **Factor.** Modifies tape speed. Variable from 0.5 (double the delay length) to 2 (half the delay length).
  - » **Inertia.** Modifies speed of changes over time, based on Factor. Variable from 0 to 5.

  Synced LFO and synced delay with LFO slower than delay means that modulation is not perceptible (except with high inertia).
- **Width.** Regulates the stereo width of the delay feedback. Variable from mono to full stereo width.
- **Ping-Pong Mode**
  - » **Off.** Select this setting to shut off Ping-Pong Mode. This also can be used to freeze the delay effect to one side, for example.
  - » **Sum.** Select this setting to feed a mono-summed mix of both channels into the delay. Try higher settings of the Width control to achieve the full ping-pong effect.
  - » **2-Ch.** This option feeds the stereo mix into the delay. Try lower settings of the Width control to “monoize” the delay effect.
- **Swap.** Click this to swap the left and right sides of the Ping-Pong effect. If Ping-Pong Mode is Mono and Width is 100%, for example, this determines whether the delay effect starts on the left or right sides.
- **Global**
  - » **Dry/Wet.** Adjusts the mix of processed signal and the original dry signal. Variable from 0 to 100%. A setting of 100%is likely to increase feedback.
  - » **Locked.** Engage this switch to prevent changes to the Dry/Wet balance.

2.1.3 Red Light Distortion

Red Light Distortion is an analog distortion emulator with several selectable distortion models. Choose between different types of distortion by selecting Soft Tube, Hard Tube, Bad Tube, Transistor, Fuzz, or OpAmp emulation. A diverse control scheme allows you to create everything from subtle harmonics that help an instrument cut through a mix to raging walls of noise.

- **In.** Input gain to the distortion. Variable from -12 dB to 24 dB.
- **Distortion.** Only for Hard and Bad Tube types, this is the tube working-point adjustment (bias). Variable from 0 to 10.00.
- **Low Freq.** Filters frequencies below this value from the distorted signal. Variable from 20 Hz to 5 kHz, depending on the High Freq setting.
- **High Freq.** Filters frequencies above this value from the distorted signal. Variable from 800 Hz to 16 kHz.
- **Drive.** Amplification during overdrive. Variable from 0 to 11; drive increases a lot between 10 and 11 for really distorted sounds.
- **Out.** Adjust the output gain of Red Light Distortion. Variable from -12 dB to 24 dB.
- **Stages.** Sets the number of overdrive stages used serially in the signal path (including filters). Select from 1, 2, or 3 with the horizontal fader.
- **Type.** Select the type of distortion emulation by clicking on the display and selecting Soft Tube, Hard Tube, Bad Tube, Transistor, Fuzz, or OpAmp from the list.
- **Mix.** Mix lets you set a mix between the wet (affected) and dry (unaffected) signals running through the plug-in, allowing for parallel processing effects.

2.1.4 Rotor

Rotor is a rotary speaker effect that simulates the sound of a tube-powered amplifier with independently rotating high-mid horn and bass woofer, as you might find attached to a classic electric organ. Historically, rotary speakers have been notoriously difficult to emulate without owning the real thing, but we feel we've done it justice.

Rotor excels at adding a sense of motion and unique tonal character to organ sounds, guitars, or anything you want to try. Each speaker's rotation can be set to a range of speeds, with realistic braking and acceleration effects when changing speeds. State Space Modeling technology provides authentic tube emulation for extra warmth and character.

- **Off/On.** Toggles the rotating action of the virtual speakers on and off, with a smooth speed transition between states.
- **Slow/Fast.** Toggles between the two preset speeds for the woofer and horn, with a smooth transition between speeds.

- **Amp**
  - » **Drive.** Add the desired amount of amp drive to the tone using State Space Modeling technology. Lower settings are cleaner, higher settings are more overdriven.
  - » **Horn Q.** Blends in a midrange peak that emulates the resonance of rotating horn speakers. Lower settings are more flat,higher settings have a more pronounced resonance.
  - » **A/B.** Toggles between two amplifier models that are recreated using State Space Modeling technology.
- **Position**
  - » **Distance.** Lets you choose the position of the virtual microphone that picks up the rotating speaker. At low settings, the mic is close to the speaker, and the stereo sweeping effects are more pronounced. At higher settings, the mic is further away from the speaker, and the effect is subtler and more diffuse.
  - » **Balance.** Lets you blend between the woofer and horn, to achieve the desired tonal balance. Fully down, you mainly hear the woofer. Fully up, you mainly hear the horn.
  - » **Spread.** Controls the stereo width of the rotating speaker elements. At low settings, the part of the signal signifying the front of each rotating element moves in a tight, distinct band across the stereo spectrum. At higher settings, the rotating element appears wider and more diffuse.
- **Trim**
  - » **Woofer Speed** and **Horn Speed.** These dual-slider controls let you set the speed at which the woofer and horn spin at slow and fast speed settings. You can set them to identical values for more coordinated rotation between woofer and horn, or to differing values that set up contrasting rotations.

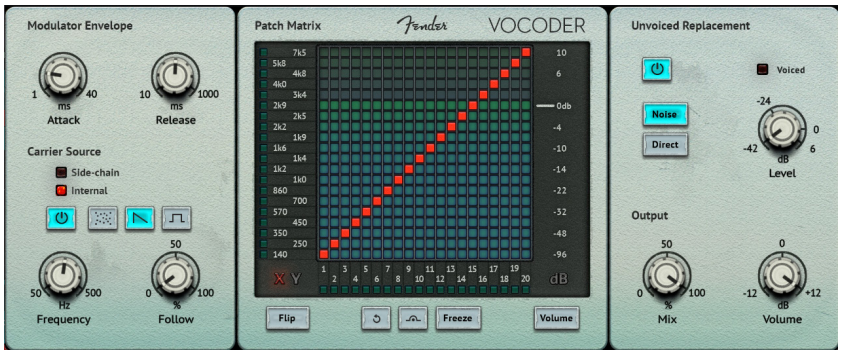
2.1.5 Tricomp

Finalize your mix or add brilliance or punch to frequency-rich signals with this three-band compressor with automatic threshold and ratio settings, plus a relative control for the low and high bands and switchable attack and release controls.

- **Input Gain.** Set overall Input Gain to the compressor
- **Mix.** Adjust the mix between dry (unaffected) and wet (effected) signals, for parallel compression effects.
- **Low.** Adjusts the relative amount of compression to be applied to the Low compression band. Variable from -5 to 5 depending on the Compress setting.
- **High.** Adjusts the relative amount of compression to be applied to the High compression band. Variable from -0.50 to 0.50, depending on the Compress setting.
- **Low Freq.** Adjusts the upper corner frequency of the Low compression band. Variable from 80 Hz to 480 Hz.
- **High Freq.** Adjusts the lower corner frequency of the High compression band. Variable from 800 Hz to 12 kHz.
- **In.** Input Meter Displays Tricomp's input level.
- **Reduction Meter.** Displays the amount of signal reduction.
- **Compression Amount.** The relative amount of compression to be applied to all three compression bands. Variable from 0 to 10.
- **Output Meter.** Displays Tricomp's output level.
- **Knee.** Adjusts the distance/curve of the compressor knee. Variable from 0 dB (hard knee) to 6 dB (soft knee).
- **Gain.** Set overall output Gain. Variable from -18 dB to +18 dB.
- **Attack.** When Auto is not engaged, these buttons select the compressor attack time. Fast Attack is 0.1 ms; Slow Attack is 10 ms.
- **Release.** When Auto is not engaged, this adjusts the compressor release time. Release is variable from 1 ms to 300 ms.
- **Auto.** Click to engage adaptive settings for the compressor attack and release times, based on signal content.
- **Saturation.** Saturation recreates the saturation sound of famous leveling amplifiers using State Space Modeling. Variable from 0 to 100%.

- **Sidechain.** Engage by clicking the [Sidechain] button at the top of the effect window to allow specific sources to control the Tricomp processor.
- **Sources.** Click to display a list of potential sidechain channel sources. A checked box indicates the current source. Multiple selections can be made.

2.2 Vocoder



The Vocoder is a creative effects processor that combines two input signals to create an entirely new sound. One signal is the **modulator**, and the other is the **carrier**.

Vocoder works by splitting the sound of the modulator signal into 20 frequency bands, each with its own bandpass filter and envelope follower. The **carrier** signal is sent through the filterbank, and the levels adjust to match the frequency of the **modulator**. The resultant sound is that of the carrier signal being filtered in such a way that its harmonic content resembles that of the modulator signal. When you start manipulating the mapping of frequency band Inputs and Outputs... that's where the real fun starts!

While typically used to give the human voice the texture (and polyphony!) of a synthesized instrument, Vocoder is by no means limited to this application. You'll find Vocoder to be highly rewarding of experimentation.

Fun fact: hardware vocoders were originally created to encrypt radio transmissions, way back in 1928. They didn't find use in creative musical applications until the 1970s.

2.2.1 Assigning input sources

Drag and drop Vocoder onto the track you want to process. This Track will serve as Vocoder's **modulator** signal.

Using the sidechain menu at the top of the Instrument Editor, assign the second audio source you want. This will be the carrier signal (You can also use an internal source from Vocoder itself as the carrier; more on that below).

We recommend familiarizing yourself with sidechaining before using Vocoder.

The following parameters are available for Vocoder:

Modulator Envelope

- **Attack** controls the attack speed of the modulator envelope.
- **Release** controls the release time of the modulator envelope.

Carrier

Carrier Source: This section is used to set up Vocoder for use with an external (sidechain) carrier signal or an internal carrier signal generated by Vocoder itself. The LEDs will reflect whether a side chain is assigned or not.

Source select (power button) toggles between sidechain and internal carrier sources. **Sidechain must be enabled at the top of the Instrument Inspector for this feature to be available.**

The following options apply only when using the Internal carrier:

- **Waveform:** Select Noise, Saw or Rect waveform. This is for the internal carrier generator only.
- **Frequency:** Controls the frequency of the internal carrier signal. When the carrier source is set to Noise, Frequency controls a blend between pink noise (left) and white noise (right).
- **Follow:** Controls modulation amount of the internal carrier pitch as determined by following the input (modulator) level. Essentially an envelope follower that can be used to add some pitch movement to an artificially static voice.

Patch Matrix

The patch matrix is a creative tool which allows you to specify routing from the vocoder's analyzer section into the synthesizer section. Doing so can help to improve intelligibility of a vocoded voice, or it can be used to push sounds from natural to more unnatural. The possibilities are almost endless!

The outputs of the 20 analyzer bands, identified at the bottom of the matrix, can be patched using "pins" to the inputs of the 20 synthesizer bands, identified on the left of the matrix. The default mapping is a diagonal (1:1).

Routings can be clicked in one at a time, or click and drag to draw a routing configuration quickly.

There are two modes (X and Y) for these routings:

X mode: there is one pin per output band, thus it is possible to assign for example one input band to one or more output bands. Click to activate X mode.

Y mode: there is one pin per input band, thus it is possible to assign or merge multiple input bands into one output band. Click to activate Y mode.

- **Flip** toggles between X and Y modes (see above).
- **Reset** resets all pins and volume sliders to the 1:1 (diagonal) configuration.
- **Bypass** bypasses the patching matrix.
- **Freeze** When activated, the contour of the synthesizer bands will be frozen (sample & hold) until the button is disengaged. When you store a preset with Freeze activated, the structure will be stored and reloaded with the preset.
- **Volume**: Click to toggle the matrix display to show the volume of each of the 20 synthesized output bands. Click or click-and-drag within the matrix to set pins at the desired volume level of each band.
- **Analyzer LEDs** indicate the level per analyzer band on the left of the matrix.
- **Synthesizer LEDs** indicate the level per synthesizer band at the bottom of the matrix.

Unvoiced Replacement

Mode: Vocoding of unvoiced speech sounds (f, s, etc.,) is often unsatisfactory. Vocoder offers a function to detect these unvoiced sounds and then replace them with other signals, including noise or a mix of the direct signal.

- **Off:** No replacement is performed.
- **Noise:** Unvoiced parts are replaced by a noise signal.
- **Direct:** A filtered portion of the original carrier signal input is directly mixed into the output.

Speech intelligibility is increased significantly with Noise or Direct mode active.

- **Level**: Controls the level of the replacement signal.
- **Voiced**: Indicates whether the input signal is rated as voiced or unvoiced.

Output

- **Mix**: Wet / Dry mix balance of the carrier input signal and the synthesized output signal.
- **Volume**: Overall output volume.

2.3 VU Meter

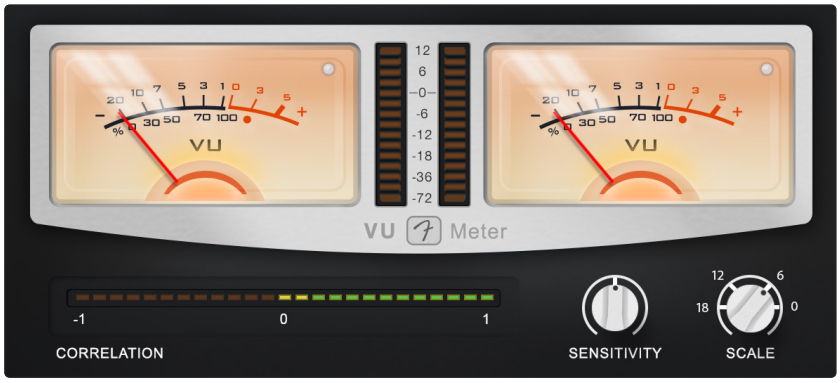
The VU Meter plug-in provides the best of both worlds for your signal metering needs: Analog-style VU meters (with clip indicators), and LED-segment-style peak meters. VU meters are most useful for showing overall signal level, whereas peak meters are most useful for keeping track of peak signal levels (like those found in the mix consoles of most DAW applications). By referencing both of these, you can see a more nuanced picture of your signal — overall level and peak level. A stereo correlation meter rounds out the trio, giving you insight into the left/right phase relationship of your signal.

Click on the Presets menu to recall factory presets. These presets cannot be overwritten.

Click on the Info menu to open Fender Studio Hub and this user manual.

2.3 VU Meter

2.3.1 VU Meter Controls



The following controls are available for you to better adjust the VU Meter to suit your needs:

- **Correlation**: This meter shows you the phase relationship of your stereo signal. When the meter is at 1, left and right channels are completely in phase. When the meter approaches –1, divergence between left and right signals is quite high, and your signal may suffer phase cancellation effects if it is later summed to mono. A correlation value between 0 and 1 is generally the goal when you are trying to avoid phase issues upon mono summing.
- **Sensitivity:** Sets the speed at which the VU meter needles move to follow changes in signal level. At lower settings, the needles move slowly, giving you a more generalized representation of signal output level. At higher settings, the needles react more quickly, giving you a more detailed view of signal level changes over time.
- **Scale**: Sets the digital signal level (in dBFS) that corresponds to 0 dBVU on the VU meters. At a setting of 0, the two scales are equivalent. For example, an input level of 0 dBFS is shown as 0 dBVU on the VU meter. With Scale set to 6, 0 dBVU now corresponds to –6 dBFS, and so on. This lets you calibrate the meter to match the amount of headroom you prefer.

3 PreSonus Plug-ins

3.1 Ampire

Available in Fender Studio Pro Native Effects™, VST3, AU and AAX formats, Ampire is a powerful and versatile collection of guitar- and bass-amplifier models based on our proprietary State Space Modeling technology, with precise emulation of every component in the signal path. Convolution-based speaker cabinets and microphones play a vital role, with variable mic positioning and phase. Signals can be processed pre- and/or post-amplifier by a bevy of effect stompboxes, some of which employ State Space Modeling of their components. Use Ampire with guitars, basses, or any audio signal to create spot-on emulation of guitar amps and cabinets and a rougher, harmonically-enriched sound.

The Ampire window is divided into four main sections:

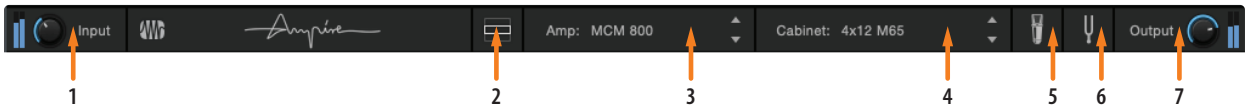
- **Toolbar.** This runs along the top of the Ampire window, and provides the basic functions such as input and output levels, window configuration, amp and cabinet selectors, and access to the microphone settings and a tuner.
- **Gallery.** The first section under the toolbar allows visual selection and display of the amps and cabinets models. It can be hidden to reduce the overall size of the Ampire window.
- **Edit Section.** The next section houses the controls for the selected amplifier model. It is always visible.
- **Pedalboard.** The bottom of the Ampire window is home to the Pedalboard, which allows you to add up to eight effects units to the signal path and configure them however you want. These effects units are also known as Stomps.

Click on the Presets menu to recall factory presets. These presets cannot be overwritten.

Click on the Info menu to open Fender Studio Hub and this user manual.

3.1.1 Toolbar

The gateway to Ampire is the toolbar, where component selections are made and basic levels are set. Viewing options are available here too, along with access to a handy tuner and other features.

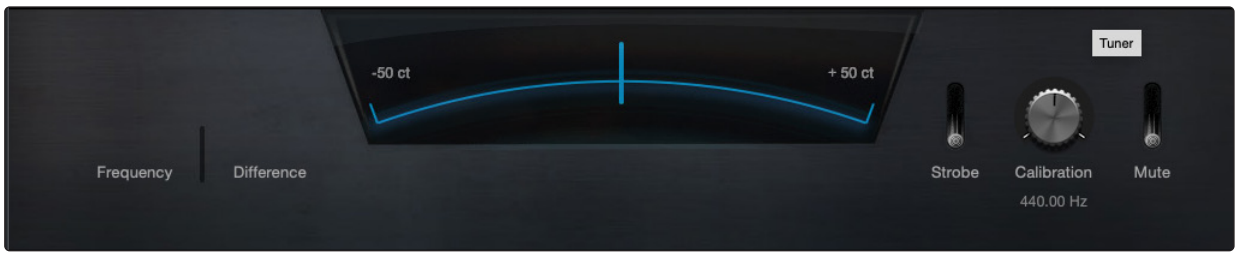


1. **Input Level.** Use the input level directly at Ampire's input to pull up weak instrument signals or to attenuate loud, processed signals. With a good input level, the signal is in the optimal range for dynamic distortion within Ampire. Variable from -12 dB to 24 dB. This parameter is not saved with each preset. It stays at a static setting for each instance of Ampire until you change it.
2. **Show / Hide Views.** This button hides or reveals different sections of the Ampire window so you only see what you want. Click above the button to toggle the Stage view; click below the button to toggle the Stomps view.
3. **Amplifier Model.** Select an amplifier model by clicking on the Amp name in the toolbar and choosing an amplifier from the Gallery. Move the cursor over the Gallery images to read a brief description of each amp. You can also use the up / down arrows in the toolbar or on the left side of the Stage view to select an adjacent amplifier.

   Changing the amp model changes the characteristics of the entire amplifier, including preamp and power amp distortion and amplification, the differences between channels 1, 2, and 3 (where applicable), and tone stack behavior. Select "None" if you want to bypass the amp head and run the signal directly through the cabinet.
4. **Cabinet Model.** Click the Cabinet Model selection box to choose a cabinet from the Gallery. Move the cursor over the Gallery images to read a brief description of each cabinet. You can also use the up / down arrows in the toolbar or on the left side of the Stage view to select an adjacent amplifier. If you want to take the direct output of the amp head, choose the Bypass icon

   You can also drag and drop Impulse Response files into the Mic A, B, and C slots for the User Cabinet. WAV and AIFF files are supported.
5. **Mic Edit Controls.** Click the Mic Edit Controls button to access the settings for the microphones that were used to capture the sound of each cabinet. *For more information,* ***please review "Mic Edit Controls" on "Mic Edit Controls" on page 16.***
6. **Show / Hide Tuner.** This button reveals or hides the Tuner window. *For more information,* ***please review "Tuner" on page 12.***
7. **Output Level.** Distortion and amplifier emulation may result in massive level changes. Use Output Level to adjust the signal to normal levels. Unlike the Input Level setting, the Output Level value is saved with each preset. Variable from -24 dB to 12 dB.

3.1.2 Tuner



- **Frequency/Difference.** When a pitched input signal is detected, the name and octave number of the nearest fundamental pitch is shown below the tuning meter (i.e., D#2), with arrows on either side to indicate whether the note is flat, sharp, or in tune (both arrows lit).
- **Frequency.** The fundamental frequency of the input signal is displayed in Hertz (Hz).
- **Difference.** The amount of deviation from center is shown as a positive number (sharp) or a negative number (flat).
- **Strobe.** Click this switch to toggle the tuner between the standard and strobe tuner views.
- **Calibration.** Click and drag the knob to set the tuning reference frequency, or enter a value manually in the number field. [Ctrl]/[Cmd]-click to reset the value to A=440.00.
- **Mute.** Activate the Mute switch to defeat the input signal temporarily while the instrument is being tuned. The signal is restored when the Tuner window is hidden. However, if the Ampire Editor window is closed while the Mute switch is still engaged, the input signal is not restored.

3.1.3 State Space Amplifier Models

Using our State Space Modeling technology to perform a component-level analysis of every circuit in these sought-after amplifiers enables Ampire to recreate the dynamic behavior and non-linearities that are critical to the distinctive tone and character of tube and semiconductor-based analog circuits.

After an amplifier is selected its controls appear in the middle of the Ampire window (the Edit section). This section will describe the controls for each of the five amplifier models.

3.1.3.1 MCM 800

This high-gain 80s classic is so loud, you can hear it from Mars—all your neighbors will hear it, too.

- **Tone Controls.** Presence controls the amount of negative feedback; increasing the value changes the high frequencies and harmonics. Bass, Middle, and Treble are basic tone controls for their respective frequency ranges.
- **Gain.** Controls Master Volume controls the final output volume. Pre-Amp Volume controls the level of the input signal before it enters the amplifier circuitry.
- **Sensitivity.** Click one of the connectors to route the signal into the High sensitivity input or the Low sensitivity input.

3.1.3.2 Dual Amplifier

This three-channel heavy metal standard is ideal when high-gain isn't enough gain. The channels of the Dual Amplifier model are numbered from right to left. They have similar EQ and tone controls, but each has different gain and circuit characteristics.

- **Channel selector.** This knob selects channel 1, 2, or 3. You can also use the small selector buttons between the toggle switches to select a channel. The channel indicators have different colors (green, orange, and red) to help you see which is active. The green channel is "clean", while the orange and red channels are more distorted.
- **Main Power.** There are two options for the power supply characteristics: Spongy or Bold.
- **Tone Controls.** These are identical for channels 1-3. Bass, Mid, and Treble are basic tone controls for their respective frequency ranges.
- **Gain Controls.** These are identical for channels 1-3. Presence controls the amount of negative feedback; increasing the value changes the high frequencies and harmonics. Master controls the final output volume for the channel. Gain controls the pre-amp input level.
- **Mode switch: Channel 1.** This switch as two positions: Clean and Pushed, each with different sound characteristics. Pushed has a boost, so the tone controls contribute less to the sound of Channel 1 in that position.
- **Mode switch: Channels 2 and 3.** This switch has three positions: Raw, Vintage, and Modern. Each provides sound characteristics that were modeled after different amplifiers from the same manufacturer.
- **Rectifier switch: All Channels.** This switch has two positions: Diode and Tube. These determine the basic character of the selected channel, which changes the output volume, the harmonic response, and the amount of headroom for that channel.

3.1.3.3 VC30

The VC30 is modeled from a versatile 70s British invader that you May associate with a regal guitarist. Served with sides of tremolo and vibrato.

- **Channel Inputs.** The VC30 has three channels: Vib-Trem, Normal, and Brilliant. Each channel has two inputs on the front panel, arranged vertically. The low-sensitivity inputs are on the top row, and the high-sensitivity inputs are on the bottom row. Each channel corresponds to a section of controls on the right side. The active channel is determined by which input is used.

- **Vibrato.** The controls in this channel are used when one of the Vib-Trem inputs is connected. Use the Speed selector to choose one of the three speeds. The Vib-Trem selector also has three settings (Vibrato, Off, or Tremolo). Adjust the level with the Vib-Trem control in the Volume section.
- **Volume.** Adjust the gain of each channel with the corresponding control in the Volume section. Usually only one channel is active, depending on which input is connected. But when the Vib-Trem and Brilliant channels are "jumpered" as shown below, the gain controls for both channels can be used to achieve the perfect blend.
- **Tone.** The Treble and Bass controls are dedicated to the Brilliant channel. The Cut control attenuates the high frequencies, and is active for all three channels. 0% = no cut, 100% clockwise = maximum cut.
- **Vib-Trem > Brilliant.** The Vib-Trem and Brilliant channels can be patched together and used at the same time. To do this, click input 2 of the Vib-Trem channel, then click it again and a patch cable appears between Vib-Trem input 1 and Brilliant input 2. This shows that the connection has been made.

  This "jumpered connection" sends the signal through both the Vib-Trem and Brilliant circuits, so you have the dirt and tone control from the Brilliant channel and the vibrato / tremolo effect from the Vib-Trem channel available. The relevant controls in each channel can be used to modify and blend the two outputs.

3.1.3.4 Blackface Twin

This loud American's benchmark cleans and lush reverb have made it a studio staple since 1960-something. It has two channels (normal and vibrato). Each channel has two input jacks that differ in input volume, as well as a Bright switch.

- **Normal.** Channel A simple channel with three EQ knobs and a Bright switch. The EQ knob range is from 1 (maximum cut) to 10 (maximum boost); 5 is neutral (no change). With Bright switched on, high frequencies can pass through the volume stage more easily. The effect depends on the Volume setting; when the volume is low, the Bright switch makes a bigger difference.
- **Vibrato Channel.** The features are identical to the Normal Channel, with the addition of Reverb and Vibrato controls. The Reverb knob controls the amount of spring reverb, with a setting of 1 shutting off the reverb. The Speed and Intensity knobs control the "Vibrato" effect (it's actually tremolo), with a minimum to maximum range of 1 to 10. An Intensity setting of 1 shuts off the Vibrato. The Reverb and the Vibrato Intensity can be toggled between their current value and "Off" by clicking below the appropriate knob.

3.1.3.5 Amp STV

Bass in your face with a gritty tube midrange. Stands out in a mix. Unsubtle. This amplifier was designed for bass instruments. It has two independent channels, with two inputs per channel (Bright and Normal). The difference between the Normal and Bright inputs is that the Bright input attenuates the low and mid frequencies.

- **Channel One.** Standard volume and boost/cut EQ controls (treble, midrange, bass), with a set of selector switches that influence the results in each EQ range. Click the desired switch to change its setting.
  - » **ultra hi** is a two-position switch: flat (0) or boost (+). It has a significant impact on the high frequencies, especially in combination with the other tone controls.
  - » **mid frequency** is a three-position switch: 220 Hz, 800 Hz, or 3000 Hz. This sets the center frequency of the midrange EQ band.
  - » **ultra lo** is a three-position switch: cut (-), flat (0), or boost (+). It has a significant impact on the low frequencies, especially in combination with the other tone controls.

Channel Two is identical to Channel One, only without the midrange controls.

3.1.4 Cabinets

Clicking the Cabinet Select field in the Toolbar will open the Cabinet Selection drop-down window. From here, you can choose between the following cabinets:

- **4x12 M65.** The holy grail of hard rock cabinets, this 4x12 classic from the 80s might give you an appetite for a little destruction.
- **4x12 MFB.** The perfect 4x12 cab for modern high-gain guitar sounds, oversized to handle extra volume—when you're ready to reaffirm your commitment to rock.
- **2x12 VC 30.** Don't let its size fool you, too: this 2x12 cabinet can deliver scintillating tone at high volume—you've finally found what you're looking for.
- **2x12 Blackface.** The dark 2x12 cabinet delivers fuzz and overdriven tones at lower volumes while still capable of sparkling, clean sounds—making it a go-to vintage standard.
- **8x10 STV.** Big bass tones that have shaken the radio for over 45 years need a cabinet the size of a refrigerator. Get the tone you want, no chiropractor required.
- **1x12 American.** Don't let its compact size fool you—this 1x12 sealed cabinet packs a serious punch with a gut-rumbling low end.
- **2x12 American.** This classic American 2x12 cabinet can sing sweet, pristine tones and switch to a swampy blues straight out of the delta.
- **1x12 British.** With mellow high frequencies and a classic British chime, this compact 1x12 cabinet is known for its versatile tone.
- **1x12 Boutique.** This 1x12 cabinet will take you right back to the glory days of rock and roll with classic American tones.
- **1x12 British II.** Portable and affordable, this '70s British bulldog was a working-class hero to a thousand bands—and a million fans.

- **2x12 Boutique.** A 2x12 cabinet with peerless tone and warmth, ready to rock with versatility.
- **4x10 American.** This 4x10 cabinet is the sound of an American legend; perfect for larger-than-life clean tones and bluesy crunch.
- **4x12 British.** The 4x12 cabinet that has lorded over countless arenas and stadiums the world over with its giant wall of sound.
- **4x12 American.** Raise your fist and get ready to shred. This 4x12 only has two volumes: loud and louder.
- **2x10 Bass.** An aggressive 2x10 powerhouse that guarantees your bass will cut through the mix.
- **8x10 Bass.** Punchy, tightly-focused bass is the hallmark of this 8x10 monster. It's a sound that you *feel*.
- **User Cabinet.** Build your own cabinet by loading impulse response files for up to three microphone positions here. A wide variety of formats are supported.

3.1.5 Mic Edit Controls

1. **Mic Mix Link.** Activate this button to link the microphone channel levels. When linked, moving one fader adjusts all three proportionately, for a combined level of 100%. If a mic level is at 0, moving the other faders adjusts only those channels. Disengage this button for independent control of each microphone level.
2. **Mic Mute.** Each channel has a Mute button. Use the appropriate button to mute only that microphone. Note that this does not affect the Link status of that channel.
3. **Mic Polarity.** Each channel has a Polarity switch. Use the appropriate button to invert the phase of that microphone.
4. **Mic B/C Delay.** These controls simulate moving Mic B and Mic C further from the cabinet. Variable from 0 to 2.9 ms.
5. **Mic A/B/C Mix.** Adjust these faders to achieve the desired level for each mic. When linked (as described above), adjusting one adjusts all three.

3.1.6 Pedalboard and Stomp Boxes

The Ampire Pedalboard features a wide variety of Stomp Box effects, including distortion and modulation of all sorts. Up to eight Stomps can be used at once and placed in any order you like (pre- or post-amplifier). Several have sync-to-tempo capabilities. You can even use the same Stomp in several slots if you want. Flexibility, thy name is Pedalboard.

If you want to use the Ampire pedalboard without the amps, a separate Pedalboard plug-in is included. This section covers the features of both (except where noted).

***Note** that all effects are placed prior to the cabinets in the audio chain.*

Before describing each parameter, let's take a look at the row of buttons beneath the Stomps window.

The numbered slots show the names of the selected Stomps in the order they appear in the Stomps window. An unused slot has no name. Each slot has an on/off switch, and on the far left is one switch to rule them all: it toggles the entire Pedalboard on and off.

Pre-/post amp placement (Ampire only)

Notice the double blue line between two of the Stomp slots; that's the dividing line between pre- and post-amplifier effects. Click-drag the blue line to place it between any two Stomps, or even before or after the entire Pedalboard. As it moves you'll see a vertical, silver switch move to the same location in the Stomps window.

Mirrored actions

Many of the actions performed in the Stomps window are mirrored in the Stomps tray, and vice versa. For example:

- Click-drag the silver switch in the Stomps window and the blue line moves too.
- Click-drag a Stomp to a new location and its numbered slot does the same below. Remember that this works both ways.
- Toggle the foot switch of an effect in the Stomps window and the on/off switch in the Stomps tray does it too.
- Hover over a number in the Stomp tray to identify that effect in the Stomps window. Hover over the top of a Stomp to see its number.

This is useful if you're using more than one instance of the same Stomp, for example, and you want to know which is which. It also helps if the Stomps are not located directly above their position numbers in the Stomp tray. Some of them are fat in more ways than one.

Add or remove a Stomp box

To insert a Stomp, right-click in an empty space in the Stomps window or in the Stomps tray. To remove a Stomp, right-click on the Stomp itself or on its numbered slot below. Note that a right-click on a Stomp control opens the Macro/Automation menu instead.

Select a Stomp box

Click the name area of any slot in the Stomps tray to open the Gallery and select an effect for that slot. There are too many to fit in the Gallery window, so be sure to scroll up or down if you don't seen the one you're looking for at first.

Bypass

This selection leaves the Stomp slot empty.

3.1.6.1 Big Fuzz

This State Space modeled distortion pedal is responsible for so many classic riffs, it would take a supermassive black hole to hold them all.

- **Volume.** Controls the level into the Tone circuit.
- **Tone.** Turn to left to attenuate high frequencies; turn to the right for more treble, less bass. Tone control is post the two clipping stages.
- **Sustain.** A pre-pre-amp stage into the Volume circuit. Adds sustain and fuzz.

3.1.6.2 Compressor

This compressor pedal reduces the dynamic range of your signal for additional shaping of your sound. With its two different modes you can dial in the desired amount of sustain precisely.

- **Gain.** Sets the input gain to the compressor.
- **Peak.** Controls the amount of peak reduction. Higher peak settings result in more gain reduction and more compression.
- **Mode: limit / compress.** Toggle between two different dynamic ratios. When in limiter mode, peaks are limited more aggressively.

3.1.6.3 Delay

- **LC.** Sets the frequency of the low-cut filter from 20 Hz to 1 kHz.
- **HC.** Sets the frequency of the high-cut filter from 1 kHz to 20 kHz.
- **Speed.** Adjusts the delay speed from 0.01 to 10 Hz.
- **Beats.** Selects a beat value for the synced delay speed.
- **FB.** Adjusts the amount of feedback from 0 to 100%.
- **Mix.** Adjusts the mix of the delayed signal with the original signal from 0 to 50%.
- **Sync.** Engage this if you want to sync the delay speed to tempo.

3.1.6.4 Equalizer

- **Guitar/Bass.** Selects the appropriate style, which adjusts the frequency values for each band of the graphic EQ.
- **Band Sliders.** Adjusts the level of each EQ band up or down to achieve the desired EQ setting. Click-drag across them to set a quick curve.

3.1.6.5 Fat

This State Space modeled 1980s popular pedalboard staple provides three controls to shape your tone for just the right amount of aggression.

- **Distortion.** Turn to the right to increase the amount of distorton.
- **Filter.** With this control fully to the left, the Filter is open and high frequencies can pass; fully to the right, the high frequencies are attenuated. Tone control is post-Distortion.
- **Volume.** Controls the final output level.

3.1.6.6 Gate

This gate pedal completely removes unwanted noise in your signal. No more buzzing and humming in your signal chain. This One-Knob model has the following controls

- **Threshold.** Adjust the threshold where the Gate will start to work in a wide range from -96dB ... 0dB. All signals above the threshold setting are passed through unaffected, whereas signals below the threshold (e.g. pick-up hum or noise) are muted.

3.1.6.7 Modulation

- **Chorus/Flanger/Phaser.** Selects the type of modulation.

Chorus

- **Delay.** Adjusts the delay of the chorus signal from 2 to 20 ms.
- **Speed.** Adjusts the chorus speed from 0.01 to 10 Hz.
- **Width.** Adjusts depth of delay line modulation, from 0 to 100%.
- **Depth.** Adjusts the chorus depth from 0 to 100%.

Flanger

- **Feed.** Adjusts the amount of feedback from 0 to 100%.
- **Delay.** Adjusts the delay of the flanged signal from .2 to 4 ms.
- **Speed.** Adjust the flanger speed from 0.01 to 10 Hz.
- **Beats.** Selects a beat value for the synced flanger speed.
- **Width.** Adjusts the flanger LFO width from 0 to 100%.
- **Depth.** Adjusts the flanger depth from 0 to 100%.
- **Sync.** Engage this if you want to sync the flanger speed to tempo.

Phaser

- **Feed.** Adjusts the amount of feedback from 0 to 100%.
- **Phase.** Adjusts the frequency of the phaser from 240 Hz to 8 kHz.
- **Speed.** Adjusts the speed of the phaser from 0.01 to 10 Hz.
- **Beats.** Selects a beat value for the synced phaser speed.
- **Width.** Adjusts the phaser LFO width from 0 to 100%.
- **Depth.** Adjusts the phaser depth from 0 to 100%.
- **Sync.** Engage this if you want to sync the phaser speed to tempo.

3.1.6.8 MP Ninety

This State Space-modeled classic phase-shifter has a singular focus and a truly unique sound.

- **Speed.** Controls the phaser speed.

3.1.6.9 PAE Chorus 1

With the flexibility of four pedals in one, this State Space-modeled chorus is the standard to which every other chorus is compared.

- **High/Low.** Selects the input sensitivity.
- **Level.** Control Adjusts the input level.
- **Chorus Intensity.** Controls the rate and depth of the Chorus effect for Chorus mode.
- **Vibrato Depth.** Controls the depth of the Vibrato effect for Vibrato mode.
- **Vibrato Rate.** Controls the rate of the Vibrato effect for Vibrato mode.
- **Normal/Effect.** This foot switch toggles the effect on and off.
- **Mode: Vibrato/Chorus.** This foot switch toggles the effect between Vibrato and Chorus modes.

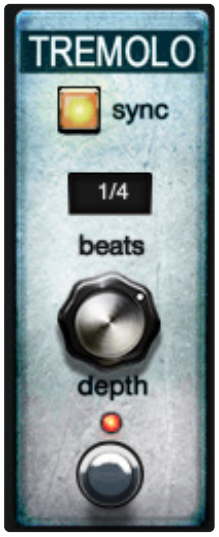
3.1.6.10 Pan

- **Sync.** Engage this if you want to sync the pan speed to tempo.
- **Speed.** Adjust the pan speed from 0.01 to 10 Hz.
- **Beats.** Selects a beat value for the synced pan speed.
- **Depth.** Adjusts the pan depth from 0 to 100%.

3.1.6.11 Reverb

- **Size.** This control affects several parameters to approximate an overall room size. It adjusts the size of the reverberated signal from 0 to 100%, with lower percentages representing smaller rooms and therefore shorter reverb tails and higher percentages representing larger rooms and longer tails.
- **Mix.** Adjusts the mix of the reverberated signal with the original signal from 0 to 50%.
- **LC.** Sets the frequency of the low-cut filter from 20 Hz to 1 kHz.
- **HC.** Sets the frequency of the high-cut filter from 1 kHz to 20 kHz.
- **Damp.** Adjusts the dampening of the reverberated signal from 0 to 100%.

3.1.6.12 Tremolo



- **Speed.** Adjusts the tremolo speed from 0.01 to 10 Hz.
- **Sync.** Engage this if you want to sync the tremolo speed to tempo.
- **Beats.** Selects a beat value for the synced tremolo speed.
- **Depth.** Adjusts the tremolo depth from 0 to 100%.

3.1.6.13 Tube Dreamer

This State Space-modeled overdrive pedal offers a mid-boost that has made it a staple for countless players, and can saturate an already-overdriven amp beautifully.

- **Drive.** Turn to the right to increase the Drive amount.
- **Tone.** Turn to the left to attenuate high frequencies. The tone control is post-Drive.
- **Level.** Controls the final output stage.

3.1.6.14 Tube Driver

- **Amount.** Adjusts the amount of drive from 0 to 11.

3.1.6.15 Wah-Wah

- **Type.** Selection Box Selects the type of wah-wah desired.
- **Amount.** Adjusts the amount of the wah-wah effect from 0 to 100%, equivalent to rocking a traditional wah-wah pedal forward and backward.

3.2 Ampire High Density Pack

The **Ampire High Density Pack** is an Add-on for Ampire, adding a special collection of three iconic amps and their respective, impulse response modeled cabinets—as well as six unique stomp box effects to complement the core library of amps, cabinets and effects included with Ampire. Ampire High Density Pack is the perfect Add-on not only for metal guitar players, but any musician looking for high-gain sounds and exotic effects.

This Add-on is compatible with the third-generation of Ampire (included in Fender Studio Pro 5.1.1 and later), as well as our VST/AU/AAX version of Ampire available within Fender Studio Pro+. As a bonus feature, the Impact XT Metal Kit (originally available as part of the Ampire XT Metal Pack) is available to you as a free download on Fender Exchange!

As an expansion, the basic functionality of Ampire High Density pack is identical to that of vanilla Ampire, so please read the Ampire section for information on basic use like the Toolbar, Mic Edit Controls, Pedalboard, etc. The amps and stompboxes detailed below will be added to your existing Ampire collection.

3.2.1 Amplifier Models

3.2.1.2 Metal Machine+

The backbone of contemporary metal tone, this legendary amp has defined the sound of modern metal with 120 watts of raw power across two channels. Each channel offers a 3-band EQ, plus independent resonance and presence controls. The rhythm channel offers an additional "Crunch" mode. Despite its massive power, this versatile amp is capable of a wide variety of sound, from warm hard rock tones to grim and frostbitten tremolo picking.

- **Channel Select.** Toggles between the Rhythm and Lead channels.
- **Tone Controls.** Pre determines the amount of gain going into the tone control section. High, Mid, and Low are basic tone controls for their respective frequency ranges. Post determines the output level at the end of the tone stack.
- **Crunch.** Adds significant distortion to the Rhythm channel.
- **Reverb.** Dial in this spring reverb emulation to taste. It has a long decay time, so it's easy to go overboard. And sometimes overboard is just enough.
- **Presence.** Available per-channel, Presence controls the amount of negative feedback in the circuit; increasing the value changes the high frequencies and harmonics.
- **Resonance.** Available per-channel, Resonance filters low frequencies in the negative feedback circuit. Use it to tame your low end.

3.2.1.2 Painapple #4

Terrifyingly high-gain, saturated tone with a simplified, single-channel architecture. featuring a 3-band EQ and four stages of high gain, it covers everything from Blues and vintage Rock tones to characteristic tones rooted in modern metal.

Pineapple's UI is icon-based. Reading left to right, its controls are:

- **Power.** Toggle between 7 and 15 watts.
- **Volume.** Post-EQ Master volume output.
- **Tone Controls.** For Treble, Mid, and Bass.
- **Gain.** Dial in the desired amount of distortion here.

3.2.1.3 Gazoline Emc2

Two channels of high-octane, high-gain genius; fuel for discerning metal connoisseurs as well as downtuned seven- and eight-string players. This extremely versatile amp offers a three-way mode switch that lets you move from clean to classic/vintage to heavy/modern and back.

- **Mode.** Use this three-way toggle switch to choose from three amplification modes, each adding more gain than the last. Use Mode 1 for the cleanest sounds and Mode 3 for the most high-gain. Note that the Mode control only affects Channel 1.
- **Channel.** Choose Channel 1 to command a variety of tones using the Mode switch, including cleaner tones. Use Channel 2 for consistent high-gain lead sounds.
- **Gain.** Controls the pre-amp input level, per channel.
- **Volume.** Sets the post-EQ output level, per channel.
- **Tone Controls.** Treble, Middle, and Bass are basic tone controls for their respective frequency ranges and are available individually for each channel.
- **Presence.** Presence controls the amount of negative feedback in the circuit; increasing the value changes the high frequencies and harmonics. Affects both Channels 1 and 2.
- **Deep.** Nuanced controls for the low, low end—use this to better match the Emc2 to your cabinet of choice. Affects both Channels 1 and 2.
- **Master.** Final output volume sentsend to DAW.

3.2.2 Stompbox Models

3.2.2.1 Demolition Drive

A screamer for the metal crowd and extended-range instruments. Overdrive your amp of choice while keeping the low end in check.

- **Diode.** Choose your clipping style: opamp, red LED, or silicon diode.
- **Vol.** Post-EQ output level, use this to push your amp of choice harder.
- **Bright.** A top-end boost for sparkling, or even raspy, highs. Turning righter=sounding brighter.
- **Attack.** An eight-way toggle offering many preset tonal variations.
- **Drive.** Sets the amount of gain/distortion.

3.2.2.2 Wildrive

Add grit to your cleans or push your lead tones WAY over the top. Combined with a clean amp, Wildrive will take you on a wild ride loaded with heavy, crunchy tones and harmonics. It's also a perfect match for our MCM 800 amp model.

- **Output.** Post-EQ output level, use this to push your amp of choice harder.
- **Tone.** Sets the Wildrive's overall tone. Darker to the left, brighter to the right.
- **Gain.** Sets the amount of gain/distortion.

3.2.2.3 Dual Comp

Compress your lows and highs independently for nuanced dynamics processing. Dual Comp's unusual design includes overlapping frequencies and independent controls for the amount of compression. There's no attack control, instead the Dual Comp has fixed fast attack on the high band and a slower attack on the lows. There's even a separate bypass for the highs, so only low frequencies are compressed.

- **Input.** Sets the level going into the Dual Comp.
- **Low Compression.** Sets the compression amount for the low end of your signal.
- **EQ Balance.** Dials in the ratio of Low signal to High, as processed by the compressors. Left for more Low and less High, right for more High and less Low.
- **High Compression.** Sets the compression amount for the high end of your signal.
- **Output.** Post-EQ/compressors output level.

3.2.2.4 Space Reverb

This doesn't sound like any real reverb... or space, for that matter. Space Reverb's uniquely characteristic shimmer is equally at home in guitar rigs and ambient synth setups because of its ability to produce huge reverb tails with a keyboard-like pad texture.

- **Mix.** Balances between your dry signal (left) and the reverb signal (right).
- **Decay.** Sets reverb tail length. Left is shorter, noon is long, right is looooooooong.
- **Shimmer.** Adds a bright, octave-up glassy enhancement to the reverb tone.

3.2.2.5 Pitch Shifter

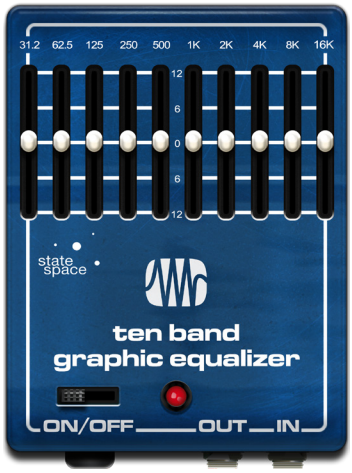
Pitch-shift your performance for radical divebombs or sky-high dog whistle tones. Covers a range of +/- two octaves. Pitch Shifter is uniquely capable of producing dramatic "dive bomb" effects, electrifying pitch-bends and stimulating harmony shifts.

- **Pitch Shifter Pedal.** (Click and drag up and down to manipulate) Changes tone pitch, based on the rules set by the
- **Harmony/Detune Knob.** Use this to choose from 16 presets, including five pitch-bends, nine harmony effects, and two detune effects.

Click the translucent panel below the Harmony/Detune knob to access the following controls:

- **Mix Harmony.** Sets the blend between the dry, unaffected signal and the harmony signal when using one of the harmony effects.
- **Mix Detune.** Sets the blend between the dry, unaffected signal and the pitch-bent signal when using one of the detune or pitch-bender effects.
- **Quality (High/Low).** Selects the harmony/detuning mode between high and low quality.

3.2.2.6 Blue EQ



Surgical tone-sculpting. Notch out troublesome frequencies or boost your favorites. With +/- 12 dB across a 10-band graphic EQ covering the full spectrum, Blue EQ is the perfect tool to fight feedback and balance your tone.

- **Band controls.** Pick any faders to precisely boost or cut the frequency ranges of your choice. Suitable for when amplifier EQs aren't cutting it (or boosting it.)

3.2.3 Cabinets

- **1x12 Metal Machine Cabinet.** The 1x12 Metal Machine cabinet is a single 12-inch, closed-back box that is the perfect match to get the signature Metal Machine+ sound.
- **2x12 Painapple Cabinet.** Pair it with the 2x12 Painapple cabinet, a dual 12-inch closed-back box. Orange you glad we didn't call it banana?
- **4x12 Gazoline Cabinet.** The 4x12 Gazoline cabinet adds even more explosive power in a quadruple 12-inch closed-back box armed with front-loaded Vintage-style drivers. Shake the rafters while you wake the neighbors.

3.3 Channel Strip Collection

The Channel Strip Collections features a State Space-modeled RC500 FET-based compressor with semi-parametric EQ and the VT1 boutique tube preamp with FET-based compressor and semi-parametric EQ.

Click on the Presets menu to recall factory presets. These presets cannot be overwritten.

Click on the Info menu to open Fender Studio Hub and this user manual.

3.3.1 VT1 Tube Modeling Channel Strip

Input

The Input section provides tools to shape your signal as it enters VT1.

- **HP Filt.** Lets you set the corner frequency of the provided high-pass filter, from 18–200 Hz. The higher you go, the more bass information is filtered out of the signal.
- **HPF Out/In.** Lets you enable (IN) or disable (OUT) the high-pass filter.
- **Trim.** Lets you adjust the amplification of your input signal, between –12 and +12 dB.
- **Impedance.** Lets you emulate the effects of changing the input impedance of a hardware preamp when a sensitive microphone is connected. The impedance mismatch yields subtle changes in tonal color. Try different settings to find what works best with the material you're processing.
- **Polarity.** Reverses the polarity of your input signal. This can be useful for minimizing the effects of phase cancellation between signals, especially when a source has been recorded with more than one microphone.
- **Gain.** This parameter directly controls the gain section of VT1's vacuum tube emulation, providing up to 20 dB of additional gain in steps of 5 dB before your signal reaches the compressor and EQ stages. This can be used to reinforce low-level signals, or to drive the compressor and EQ stages harder.

Compressor

Use the Compressor section to control or re-shape the dynamic behavior of your signal. Compression is often used to reduce the dynamic range of a signal (to control peaks and accentuate low-level information) or to accentuate signal peaks, to increase apparent "punch."

- **Threshold.** Sets the threshold of the compressor in a wide range. When the amplitude of the input signal exceeds the threshold you've set, the compressor engages.
- **Ratio.** Sets the slope of the compressor as a ratio between input and output levels, from 1:1 to 4:1. At 1:1, no gain reduction occurs at any threshold setting. At 2:1, for every 2 dB of signal level above the threshold setting, only 1 dB of signal increase occurs, and so on.
- **Attack.** Sets the speed at which the compressor acts on input signals that cross the Threshold setting, in a range between 0.5–500 ms. Slower attack times allow signal peaks to pop through uncompressed before gain reduction reaches the amount set by the Threshold and Ratio controls, accentuating the attack of transient signals. Faster attack times control peaks more closely.

- **Release.** Sets the speed at which the compressor returns to normal after input signal crosses below the threshold setting, in a range between 40–500 ms. Some signals sound their best with shorter release settings, some sound best with a longer release. Letting the signal recover faster from the compression effect can be useful to enhance the room sound of microphone signals. Slow release settings can create for smooth and less obvious compression effect. Set this control to taste.
- **Comp Bypass.** Bypasses the compressor section. When the indicator light is on, thecompressor is bypassed.
- **Gain.** Provides up to 18 dB of make-up gain for the compressed signal.
- **EQ > Comp.** Switches the order that signals enter each processing stage. When the indicator light is off, the signal goes to the compressor before heading to the EQ. When the light is on, the signal goes to the EQ before heading to the compressor.

Equalizer

Use the Equalizer to boost or cut specific areas of the frequency spectrum of your signal, to accentuate or de-emphasize those frequencies, as needed.

- **Freq.** Sets the center frequency of the current EQ band.
- **Gain.** Sets the level of boost or cut applied by the current EQ band, between –18.5 and +18.5 dB.
- **EQ Bypass.** Bypasses the equalizer section. When the indicator light is on, the EQ is bypassed.
- **HF Peak.** Switches the high EQ band between shelf (boost/cut is applied to all frequencies including and above the chosen frequency) or peak (boost/cut centers around the chosen frequency) modes. When the indicator light is on, the high band is in peak mode.
- **LF Peak.** Switches the low EQ band between shelf (boost/cut is applied to all frequencies including and below the chosen frequency) or peak (boost/cut centers around the chosen frequency) modes. When the indicator light is on, the low band is in peak mode.

Meter and Level

- **MTR-GR/MTR-IN.** Lets you choose to display either the level of gain reduction occurring in the compressor section (GR), or input signal level (IN).
- **Level.** Lets you adjust the overall output level of VT1, from –90 to +11 dB.

3.3.2 RC500 FET Modeling Channel Strip

Compressor

Use the Compressor section to control or re-shape the dynamic behavior of your signal. Compression is often used to reduce the dynamic range of a signal (to control peaks and accentuate low-level information) or to accentuate signal peaks, to increase apparent "punch."

- **Threshold** sets the threshold of the compressor, in a range between –25 and +18 dB. When the amplitude of the input signal exceeds the threshold you've set, the compressor engages.
- **Attack** sets the speed at which the compressor acts on input signals that cross the Threshold setting, in a range between 0.5–10 ms. Slower attack times allow signal peaks to pop through uncompressed before gain reduction reaches the amount set by the Threshold and Ratio controls, accentuating the attack of transient signals. Faster attack times control peaks more closely.
- **Release** sets the speed at which the compressor returns to normal after input signal crosses below the threshold setting, in a range between 40–500 ms. Some signals sound their best with shorter release settings, some sound best with a longer release. Letting the signal recover faster from the compression effect can be useful to enhance the room sound of microphone signals. Slow release settings can create for smooth and less obvious compression effect. Set this control to taste.
- **Compressor Out/In** bypasses (OUT) or enables (IN) the compressor section.

Preamp

This section provides processing tools for the signal, applied as it enters and exits RC500.

- **Gain.** Provides a generous amount of additional gain before your input signal reaches the compressor and EQ stages. Can be used to reinforce low-level signals, or to drive the compressor and EQ stages harder.
- **Polarity.** Reverses the polarity of your input signal. This can be useful for minimizing the effects of phase cancellation between signals, especially when a source has been recorded with more than one microphone.
- **Hi Pass Filter.** Lets you toggle the provided high-pass filter on or off for the input signal. When enabled, low-frequency information below 80 Hz is attenuated by a slope of 12 dB per octave.
- **Master.** Adjusts the overall output level.

Equalizer

Use the Equalizer to boost or cut specific areas of the frequency spectrum of your signal, to accentuate or de-emphasize those frequencies, as needed.

- **Freq.** Sets the center frequency of the current EQ band.
- **Gain.** Sets the level of boost or cut applied by the current EQ band, between –18.5 and +18.5 dB.
- **Peak/Shelf.** Switches the corresponding EQ band between shelf or peak modes. In peak mode, the EQ band boosts and cuts signals centered around the frequency you set with the...
- **Freq control.** In shelf mode, all frequencies at or above the chosen frequency (on the high band) or at or below the chosen frequency (in the low band) are affected.
- **Equalizer Out/In.** Bypasses (OUT) or enables (IN) the equalizer section.

Meter

- **MTR-GR/MTR-OUT.** Lets you choose to display either the level of gain reduction occurring in the compressor section (GR), or overall output level (OUT).

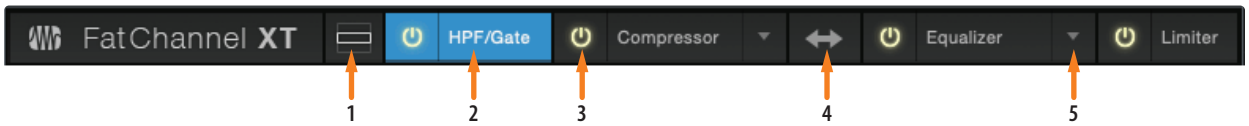
3.4 Fat Channel XT

Fat Channel XT is a complete virtual version of the channel strip found on the StudioLive Series III line of mixers. With gate/expander, compressor, parametric EQ and limiter functions, Fat Channel combines many essential processing functions into one easy-to-use tool. Along with its "clean" modern dynamics and EQ processors, Fat Channel XT gives you a selection of high-quality emulations of vintage compressors and EQs to suit your needs.

Click on the Presets menu to recall factory presets. These presets cannot be overwritten.

Click on the Info menu to open Fender Studio Hub and this user manual.

3.4.1 Fat Channel XT Global Controls



1. **Stacked Mode.** Click this option to toggle the display state of Fat Channel XT. When disabled, only the currently selected processor (such as Gate or Compressor) is displayed. When enabled, all four processors are displayed at once, in a stacked arrangement.
2. **Processor Select Buttons (Gate, Compressor, Equalizer, Limiter).** When Fat Channel XT is not in Stacked Mode, click these buttons to display the processor block of your choice.
3. **Processor Enable/Disable.** Click the round button next to the processor name of your choice to toggle the on/off state for that block of processing. Each processor also has its own enable/disable switch within the module interface.
4. **Swap Comp/EQ Order.** Click this button to swap the places of the Compressor and Equalizer processors in the signal chain.
5. **Compressor and Equalizer Model Selectors.** Click the menu next to the name of the Compressor or Equalizer processors to open a Gallery view, from which you can choose the desired Compressor or EQ model. Hover the cursor over each image to view a brief description, then click to select the one you want.

   Compressor offers the following choices:

   » **Standard.** A flexible, modern compressor, with a clean, hi-fi sound.

   » **Tube.** A model of one of the best-loved vintage tube-based opto-compressors. Excels at vocal smoothing and at making bass instruments sound larger-than-life.

   » **FET.** A model of one of the most-used vintage FET-based compressors. Great for adding an aggressive edge and accentuating room sound for drums, guitars, and other highly transient signals.

Equalizer offers the following choices:

» **Standard.** A flexible, full-featured modern EQ, with a clean, hi-fi sound.

» **Passive.** A model of the "rolls-royce" of vintage tube-based passive EQs. Deceptively simple controls and a rich, thick sound make it perfect for gentle tone shaping or adding vintage character.

» **Vintage.** A model of what some call the "final word" in vintage solid-state EQs. Combines an "everything sounds better through it" quality with musically-chosen EQ frequencies for quick, reliable tonal magic.

3.4.2 High-Pass Filter (HPF)

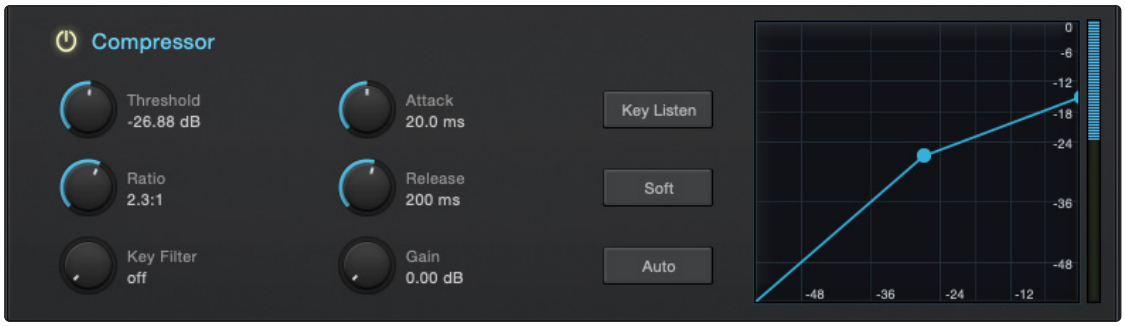
- **HPF.** Sets the frequency of the high-pass filter. Turn all the way left to disengage the filter.

3.4.3 Gate

- **Enable/Disable.** Click the power button either in the header or in the module to enable or disable the Gate.
- **Threshold.** This knob sets the level at which the gate opens. Essentially, all signals above the threshold setting are passed through unaffected, whereas signals below the threshold setting are reduced in level by the amount set by the range control. If the threshold is set all the way to the left, the gate is turned off (always open), allowing all signals to pass through unaffected. You can set the threshold from 0 to -84 dB.
- **Range.** This adjusts the amount of gain reduction the gate produces. The range can be set from 0 to -84 dB. The Range control is not available when using the expander.
- **Key Filter.** This knob adjusts the frequency at which the gate opens. Setting a specific frequency, in addition to a specific decibel level, provides more sonic shaping. The key filter can be triggered by the selected channel or bus's signal or by sidechaining a channel and using its signal as the source.
- **Attack (Att).** This adjusts the rate at which the gate opens on the selected channel or output. You can set the attack time from 0.02 to 500 ms.
- **Release (Rel).** Adjusts and displays the rate at which the gate closes on the selected channel. The release time can be set from 0.05 to 2 seconds.
- **Key Listen.** This button engages and disengages the Key Listen function, which lets you hear how the gate Key Filter is set.
- **Expander.** Switch the gate into expander mode.
- **Interactive Graph.** This graph provides a visual representation of the settings and current activity of the gate. You can also adjust the setting by moving the blue dots to adjust Threshold and Range.

3.4.4 Compressor

3.4.4.1 Standard Compressor



- **Enable/Disable.** Click the power button either in the header or in the module to enable or disable the Compressor.
- **Threshold.** This knob adjusts the compressor threshold for the selected channel. The compressor engages as soon as the signal level (amplitude) exceeds the threshold value. Moving this control to the left lowers the threshold so that compression begins at a lower amplitude value. The threshold can be set from -56 to 0 dB.
- **Ratio.** This knob adjusts the compression ratio (or slope). The ratio is a function of the output level versus the input level. For example, if you have the ratio set at 2:1, any signal levels above the threshold setting are compressed at a ratio of 2:1. This means that for every 2 dB of level increase above the threshold, the compressor's output only increases by 1 dB. The ratio can be set from 1:1 to 18:1 or "limit" which is the equivalent of infinity:1.
- **Gain.** This sets and displays the makeup gain of the compressor for the selected channel. Compressing a signal usually results in an overall reduction in level (gain reduction), and the Makeup Gain control lets you increase the volume to make up for this gain loss, if desired. You can adjust the Makeup Gain from 0 dB (no gain adjustment) to +28 dB.
- **Attack.** This adjusts the speed at which the compressor acts on the input signal. A slow attack time (moving the slider to the right) allows the beginning component of a signal (commonly referred to as the initial transient) to pass through, uncompressed, whereas a fast attack time (fully to the left) triggers compression immediately when a signal exceeds the threshold. You can set the attack from 0.2 to 150 milliseconds.
- **Release.** This determines the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after crossing below the compression threshold. Release can be set from 2.5 to 900 milliseconds.
- **Key Listen.** This button engages and disengages the Key Listen function, which lets you listen to the signal that is being fed to the compressor's detector.
- **Auto.** This enables Automatic Attack and Release mode. When Auto mode is active, the Attack and Release controls become inoperative, and a pre-programmed attack and release curve is used that sets the attack to 10 ms and the release to 150 ms. Meanwhile, all other compressor parameters can still be adjusted manually.
- **Soft.** This engages soft-knee compression. In normal operating mode, the compressor is set for hard-knee compression, meaning the gain reduction applied to the signal occurs as soon as the input signal level exceeds the threshold value. When the Soft Knee button is engaged, the ratio increases gradually as the signal reaches the threshold.
- **Interactive Graph.** This graph provides a visual representation of the settings and current activity of the compressor. You can also adjust the setting by moving the blue dot to change the Threshold and Ratio values.

3.4.4.2 Tube Compressor

- **Enable/Disable.** Click the power switch to enable or disable the Tube Compressor module.
- **Comp/Limit.** Toggles the Tube Compressor between its compressor and limiter modes. When in compressor mode, it acts with a variable ratio of 1:1-10:1. When in limiter mode, it acts with a variable ratio of 10:1-20:1, more aggressively limiting peaks.
- **Gain.** Sets input gain to the compressor. Because this type of compressor operates in a different way than a standard compressor, much of the way that it affects signals is based on the input level. Try different settings to see what suits your needs.

- **VU Meter (Gain Reduction).** This vintage-style VU meter shows a smoothed representation of gain reduction applied by the compressor over time.
- **Peak Reduction.** Sets the amount of peak reduction to apply to the signal. Higher settings result in more gain reduction and more pronounced compression effect.
- **Key Filter.** Sets the frequency of a high-pass filter that sits in the compressor sidechain. The higher the setting, the more frequencies are excluded from reaching the compressor's detector, with a variety of useful dynamic results. Ranges from "Off" to 16 kHz.
- **Key Listen.** This button engages and disengages the Key Listen function, which lets you listen to the signal that is being fed to the compressor's detector, after it has passed through the Key Filter.

3.4.4.3 FET Compressor

- **Enable/Disable.** Click the power switch to enable or disable the FET Compressor module.
- **Input.** Sets input gain to the compressor. This setting affects the action of the compressor, so feel free to try various settings to find the optimal effect for your needs.
- **Output.** Sets the amount of "makeup gain" to apply to a signal. Once a signal is compressed, its overall level is often reduced. This gain control lets you bring it back up to the proper level after compression occurs.
- **Attack.** This adjusts the speed at which the compressor acts on the input signal. A slow attack time (moving the slider to the right) allows the beginning component of a signal (commonly referred to as the initial transient) to pass through, uncompressed, whereas a fast attack time (fully to the left) triggers compression immediately when a signal exceeds the threshold. Attack ranges between 0.8 to 0.02 milliseconds.
- **Release.** This determines the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after crossing below the compression threshold. Release ranges between 1.1 second to 50milliseconds.
- **Ratio Selector Buttons.** These buttons let you choose a compression ratio: 4:1, 8:1, 12:1, 20:1, or "All."The ratio is a function of the output level versus the input level. For example, if you have the ratio set at 4:1, any signal levels above the threshold setting are compressed at a ratio of 4:1. This means that for every 4 dB of level increase above the threshold, the compressor's output only increases by 1 dB. The "All" setting recreates the "all buttons pushed in" setting that helped make this compressor type a legend, providing massive punch and crunch when driven hard.
- **VU Meter (Gain Reduction).** This vintage-style VU meter shows a smoothed representation of gain reduction applied by the compressor over time.
- **Key Listen.** This button engages and disengages the Key Listen function, which lets you listen to the signal that is being fed to the compressor's detector, after it has passed through the Key Filter.
- **Key Filter.** Sets the frequency of a high-pass filter that sits in the compressor sidechain. The higher the setting, the more frequencies are excluded from reaching the compressor's detector, with a variety of useful dynamic results. Ranges from "Off" to 16 kHz.

3.4.5 Equalizer

3.4.5.1 Standard EQ

- **Enable/Disable.** Click the power button to enable or disable the Equalizer module.

- **Freq.** This knob selects the center frequency of the corresponding band. You can adjust the center frequency in the following ranges for each band: Low Band: 36 to 465 Hz Low-Mids: 90 Hz to 1.2 kHz Hi-Mids: 380 Hz to 5 kHz Highs: 1.4 to 18 kHz
- **Gain.** This knob boosts and attenuates the selected frequency with a range of -15 to +15 dB.
- **Q.** This adjusts the Q value for the corresponding frequency band. The Q is the ratio of the center frequency to the bandwidth. When the center frequency is constant, the bandwidth is inversely proportional to the Q, so as you raise the Q, you narrow the bandwidth. Hence, the smaller the number, the wider the curve.
- **Interactive Graph.** This graph provides a visual representation of the current settings. You can change the settings by moving the blue dots to adjust the frequency and gain at the same time. The first time you touch a dot, the associated band automatically turns on. Tapping or clicking a dot turns the band on and off.
- **Low Shelf (LS) & High Shelf (HS) Buttons.** These buttons turns Shelving mode on or off for the High and Low bands. When the Shelf button is engaged, the associated High or Low frequency section is switched from parametric EQ to shelving EQ.
- **Band Enable/Disable Buttons.** These buttons select which EQ band is being controlled by the Frequency, Gain, and Q controls.

3.4.5.2 Passive EQ

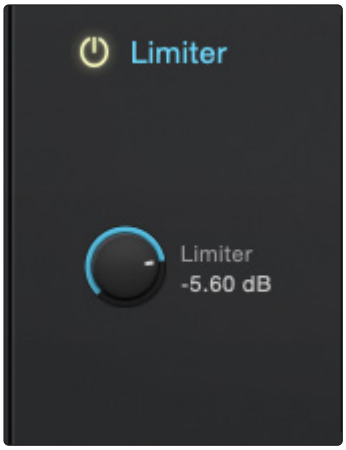
- **Enable/Disable.** Click this On/Off switch to enable or disable the Passive Equalizer module.
- **Low Boost.** Sets the level of boost applied around the chosen low frequency. This control interacts nicely with the Low Attenuation control, allowing for boosts in apparent bass energy while keeping overall bass energy within optimal limits.
- **Low Attenuation.** Sets the level of attenuation applied around the chosen low frequency. This control interacts nicely with the Low Boost control, allowing for boosts in apparent bass energy while keeping overall bass energy within optimal limits.
- **Low Frequency.** Sets the center frequency of the band covered by the Low Boost and Low Attenuation controls.
- **High Bandwidth.** Sets the bandwidth (Q) for the High Band.
- **High Boost.** Boosts the High Band.
- **High Frequency.** Sets the center frequency for the High Band.
- **High Attenuation.** Sets the attenuation for the High Band.
- **Attenuation Select.** Sets the Frequency at which the attenuation begins.

3.4.5.3 Vintage EQ

- **Enable/Disable.** Click the power button to enable or disable the Vintage Equalizer module.
- **Low Frequency.** Sets the corner frequency of the low-frequency shelving band of this EQ. Choose from 35, 60, 110, or 220 Hz.
- **Low Gain (LF).** Sets the amount of boost or cut to apply the to low-frequency band of this EQ. Range of plus or minus 16 dB.
- **Low-Mid Frequency.** Sets the center frequency of the low-mid-frequency band of this EQ. Choose from 360 Hz, 700 Hz, or 1.6 kHz.
- **Low-Mid Gain (LMF).** Sets the amount of boost or cut to apply to the low-mid-frequency band of this EQ. Range of plus or minus 16 dB.
- **High-Mid Frequency.** Sets the center frequency of the low-mid-frequency band of this EQ. Choose from 3.2, 4.8, or 7.1 kHz.

- **High-Mid Gain (HMF).** Sets the amount of boost or cut to apply to the high-mid-frequency band of this EQ. Range of plus or minus 16 dB.
- **High Gain (HF).** Sets the amount of boost or cut to apply the high-frequency shelving band of this EQ. Range of plus or minus 16 dB.

3.4.6 Limiter



- **Enable/Disable.** Click the power button either in the header or in the module to enable or disable the Limiter.
- **Threshold.** Sets and displays the threshold of the limiter on the selected channel. The limiter engages as soon as the signal level (amplitude) exceeds the threshold value. Moving this control to the left lowers the threshold so that limiting begins at a lower amplitude value. The threshold can be set from -28 to 0 dB.

3.5 Fat Channel Collection Volume 1

Fat Channel XT Plug-in Add-ons allow you to enhance your Fat Channel XT with additional processor models. These plug-ins are virtual signal processors that load in Fat Channel XT to expand your compressor and EQ processor libraries. Each plug-in comes in both StudioLive Series III format and Fat Channel XT format so you can use your new processor on your computer and in your mixer.

The Classic Studio, Modern Classics, and Vintage Channel Strips bundles are available individually from your authorized Fender dealer or from Fender.com. Or buy all three bundles—12 processors in all—in the Fat Channel Collection, Volume 1, to supercharge Fat Channel XT.

To load any Fat Channel plug-in model, add the Fat Channel XT plug-in to your session and click the model menu from either the Compressor or EQ section. You'll find any authorized and installed plug-in model available to be loaded within the Fat Channel plug-in.

3.5.1 Available Models

Available Compressor Models

The following compressor models can be added to your Fat Channel XT plug-in:

- **Brit Comp.** Capturing the unique sound of a twin VCA gain-reduction amplifier design, the Brit Comp is ideal for taming piano dynamics or adding punch to drums and percussion. Included in the Classic Studio Bundle.
- **Classic Compressor.** The smooth character of this compressor allows you to create transparent or extreme color changes to your audio, making it a workhorse for just about any application. Included in the Vintage Channel Strips Bundle.
- **Comp 160 Compressor.** With simple controls, yet capable of extreme compression traits, the Comp 160 provides VCA character with a personality all its own. Try it on drums—you'll be glad you did! Included in the Modern Classics Bundle.
- **Everest C100A Compressor.** Based on a classic design focused on gentle, natural-sounding gain reduction, the Everest C100A helps control dynamics while still letting the signal breathe. Included in the Modern Classics Bundle.
- **FC-670 Compressor.** This model of an iconic compressor/limiter of the 1950s imparts an unmistakable silky warmth on just about any signal. Included in the Classic Studio Bundle.
- **RC-500 Compressor.** FET-based compressors such as the one in the Fender RC 500 use transistors to emulate a triode tube's operation and sound. This type of compressor generally provides a faster attack time and better repeatability than the optical compressors typically found in channel strips. Modeled after Fender' RC500 signature FET compressor, this plug-in provides an ultra-fast attack time and repeatable performance.
- **The Tube CB Compressor.** In general, the response time of optical compressors tends to soften the attack and release, which can smooth out uneven volume fluctuations. Emulating an all-tube, optical design, the Tube P1B compressor delivers musicality, preserving the clarity of the signal even at the most extreme settings.
- **VT-1 Compressor.** Stunning high-end sound and incredible versatility are hallmarks of Fender' VT-1, which is modeled after a popular vacuum tube channel strip's FET compressor. The VT-1 provides fast attack and repeatable performance with a fully variable ratio.

Available Equalizer Models

The following EQ models can be added to your Fat Channel XT plug-in:

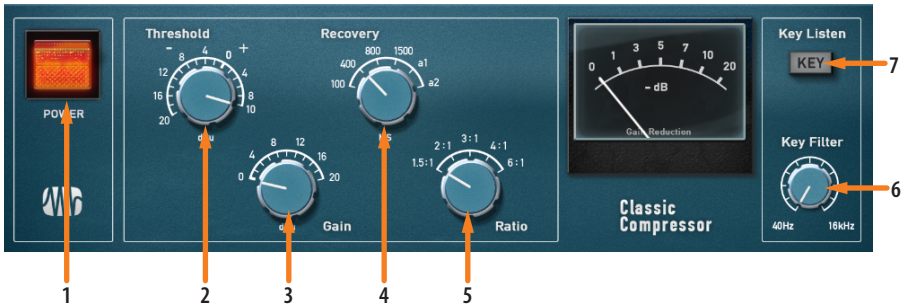
- **Alpine EQ-550.** The 1960s-vintage EQ provides consistent, repeatable equalization using three overlapping bands, divided into seven fixed frequency points, each with five steps of boost or cut. Its selectable peaking or shelving filters for the high and low band, along with an independently insertable bandpass filter, provide an easy path to creating acoustically superior equalization. Included in the Classic Studio Bundle.
- **Baxandall EQ.** This EQ offers the world's most popular EQ curve. Using gently sweeping treble and bass EQ shelves, it allows you to make subtle, yet effective, changes over wide swaths of the frequency spectrum. Included in the Modern Classics Bundle.
- **RC-500 EQ.** It's rare to find a truly outstanding solid-state channel strip that can deliver a vintage vibe reminiscent of classic high-end products, yet employs a thoroughly modern design. The Fender RC 500 was one of these rare gems. The RC 500 EQ plug-in is modeled after the channel strip's 3-band semi-parametric EQ and combines isolated filters and optimized, per-band Q to provide subtle signal-shaping without harsh artifacts.
- **Solar 69 EQ.** The sound of classic British EQ is absolutely legendary and has enhanced many a great recording. Emulating this classic British design, the Solar 69 EQ adds definition to kick drums, shapes electric guitars, and adds shimmer to acoustic guitars and vocals without sacrificing body. Included in the Classic Studio Bundle.
- **The Tube EQ.** This midrange EQ is based on a passive, all-tube design for ultra-smooth and musical equalization, making it ideal for any midrange source material. Included in the Vintage Channel Strips Bundle.
- **Vintage 3-band EQ.** With its distinct filter shaping, sheen, and bite, this three-band active EQ includes both high and low shelving filters, providing enhanced tone-shaping possibilities. Included in the Vintage Channel Strips Bundle.
- **VT-1 EQ.** The VT-1 EQ models a popular vacuum tube channel strip's 4-band semi-parametric EQ, which combines isolated filters and optimized, per-band Q. Designed with musicality in mind, this EQ is smooth and refined, which made it an instant classic.

3.5.2 Compressor Plug-in Models

3.5.2.1 Brit Comp

1. **Compressor On / Off.** Enables / disables the compressor in the signal chain.
2. **Threshold.** Sets the Threshold for the compressor.
3. **Attack.** Sets the Attack time in milliseconds (ms) for the compressor.
4. **Ratio.** Sets the compression Ratio.
5. **Release.** Sets the Release time in seconds (s) for the compressor.
6. **Makeup.** Sets the amount of make-up gain applied to the signal by the compressor.
7. **Key Filter.** Sets the Key Filter frequency.
8. **Key On/ Off.** Enables / disables the Key Filter.

3.5.2.2 Classic Compressor



1. **Compressor On / Off.** Enables / disables the compressor in the signal chain.
2. **Threshold.** Sets the Threshold for the compressor.
3. **Gain.** Sets the amount of make-up gain applied to the signal by the compressor.
4. **Recovery.** Sets the release time for the compressor.
5. **Ratio.** Sets the compression Ratio.
6. **Key Filter.** Sets the Key Filter frequency.
7. **Key Filter On/ Off.** Enables / disables the Key Filter.

3.5.2.3 Comp 160 Compressor

1. **Threshold.** Sets the Threshold for the compressor.
2. **Compressor On / Off.** Enables / disables the compressor in the signal chain.
3. **Compression.** Sets the Compression ratio.
4. **Output Gain.** Adjusts the gain of the of the output amplifier stage. The Output Gain control does not interact with the compressor threshold.
5. **Key Listen.** Enables / disables the Key Filter.
6. **Key Filter.** Sets the Key Filter frequency.

3.5.2.4 Everest C100A Compressor

1. **Compressor On / Off.** Enables / disables the compressor in the signal chain. This can be controlled both by clicking or tapping on the LED or by clicking or tapping on the switch (Call-out #5).
2. **Attack.** Sets the Attack time for the compressor between Fast, Medium, and Slow.
3. **Release.** Sets the Release time for the compressor between Fast, Medium, and Slow.

4. **Gain.** Sets the output gain of the compressor. This level is monitored by the meter.
5. **Compressor On / Off.** Enables / disables the compressor in the signal chain. This can be controlled both by clicking or tapping on the switch or by clicking or tapping on the LED (Call-out #1).
6. **Gain Reduction.** Sets the amount of gain reduction applied to the signal.
7. **Key Filter.** Sets the Key Filter frequency.
8. **Key Listen On/ Off.** Enables / disables the Key Filter.

3.5.2.5 FC-670 Compressor

1. **Compressor On / Off.** Enables / disables the compressor in the signal chain.
2. **Input Gain.** Adjusts the gain of the input signal as it goes into the compressor.
3. **Threshold.** Sets the Threshold for the compressor.
4. **Time Constant.** Adjusts the fixed Attack and Release settings.
5. **Key Filter.** Sets the Key Filter frequency.
6. **Key Listen On/ Off.** Enables / disables the Key Filter.

3.5.2.6 RC-500 Compressor

1. **Compressor On / Off.** Enables / disables the compressor in the signal chain.
2. **Threshold.** Sets the Threshold for the compressor.
3. **Attack.** Sets the Attack time in milliseconds (ms) for the compressor from 0.5 ms (Fast) to 10 ms (Slow).
4. **Makeup.** Sets the amount of make-up gain applied to the signal by the compressor.
5. **Release.** Sets the Release time in seconds (s) for the compressor from 40 ms (Fast) to 500 ms (Slow).
6. **Key Filter.** Sets the Key Filter frequency.
7. **Key Filter On/ Off.** Enables / disables the Key Filter.

Power User Tip: *The RC-500 Compressor has a fixed ratio of 3:1. This ratio will work well for a wide variety of instruments. If you would like more compression, lower the threshold while raising the input level. Lighter compression can be easily achieved by raising the threshold.*

3.5.2.7 The Tube CB Compressor

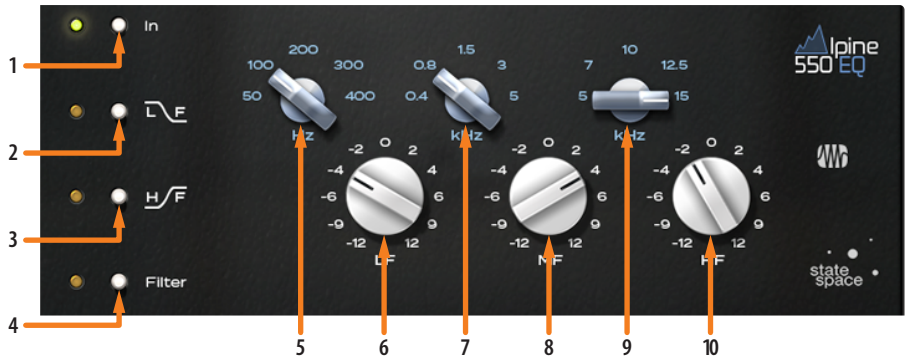
1. **Power On/Off.** Enables / disables the compressor in the signal chain.
2. **Gain.** Sets the amount of make-up gain applied to the signal by the compressor.
3. **Attack.** Sets the Attack time for the compressor.
4. **Threshold.** Sets the Threshold for the compressor.
5. **Release.** Sets the Release time for the compressor.
6. **Ratio.** Sets the compression Ratio.
7. **Attack / Release Mode.** When set to "Fixed", the Attack time is set to 1 ms and the Release time is set to 50 ms. When set to "Manual", these times are user definable by the Attack (#2) and Release (#4) controls.
8. **Key On/ Off.** Enables / disables the Key Filter.
9. **Key Filter.** Sets the Key Filter frequency.

3.5.2.8 VT-1 Compressor

1. **Compressor On / Off.** Enables / disables the compressor in the signal chain.
2. **Threshold.** Sets the Threshold for the compressor.
3. **Attack.** Sets the Attack time for the compressor from 0.5 ms (Fast) to 10 ms (Slow).
4. **Release.** Sets the Release time in seconds (s) for the compressor from 40 ms (Fast) to 500 ms (Slow).
5. **Ratio.** Sets the compression Ratio.
6. **Gain.** Sets the amount of make-up gain applied to the signal by the compressor.
7. **Key Filter.** Sets the Key Filter frequency.
8. **Key Filter On/ Off.** Enables / disables the Key Filter.

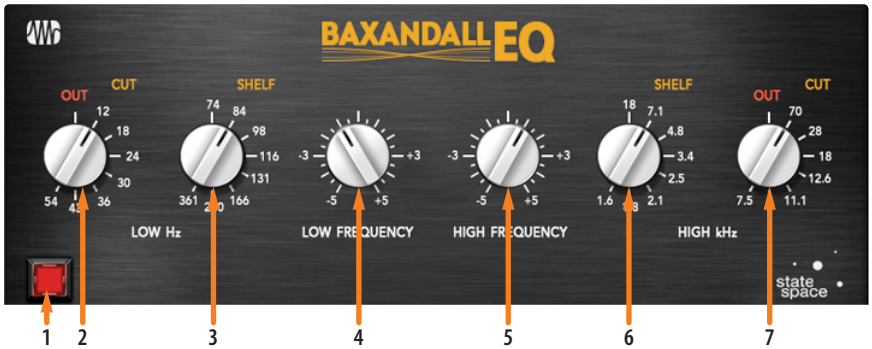
3.5.3 Equalizer Plug-in Models

3.5.3.1 Alpine EQ-550



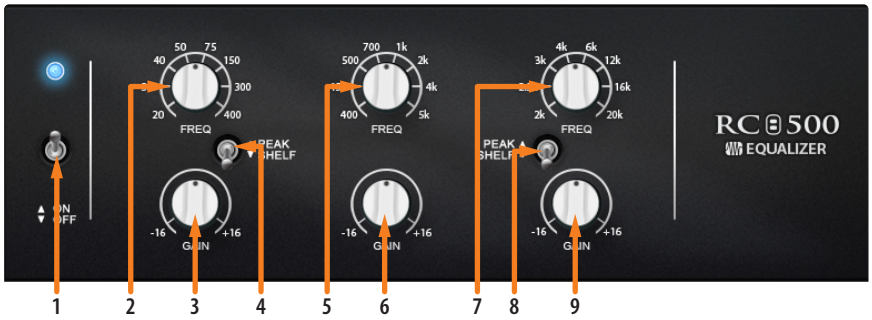
1. **EQ On/Off.** Enables / disables the EQ in the signal chain.
2. **L/F.** Toggles between a Peak and Shelf EQ for the low band.
3. **H/F.** Toggles between a Peak and Shelf EQ for the high band.
4. **Filter.** Toggles a Band Pass filter from 50 to 15 kHz on or off.
5. **Low Band Frequency.** Sets the frequency for the low band.
6. **Low Band Gain.** Sets the amount of gain for the low band.
7. **Mid Band Frequency.** Sets the frequency for the mid band.
8. **Mid Band Gain.** Sets the amount of gain for the mid band.
9. **High Band Frequency.** Sets the frequency for the high band.
10. **High Band Gain.** Sets the amount of gain for the high band.

3.5.3.2 Baxandall EQ



1. **EQ On/Off.** Enables / disables EQ in the signal chain.
2. **Low Cut.** This controls the cut-off frequency of the 12 dB/octave Low Cut filter from 54 to 12 Hz.
3. **Low Shelf.** Sets the threshold of the low shelf from 74 to 361 Hz.
4. **Low Frequency.** Sets the amount of gain for the low band.
5. **High Frequency.** Sets the amount of gain for the high band.
6. **High Shelf.** Sets the threshold for the high shelf from 1.6 to 18 kHz.
7. **High Cut.** This controls the cut-off frequency of the 12 dB/octave High Cut filter from 54 to 12 Hz.

3.5.3.3 RC-500 EQ



1. **EQ In/Out.** Bypasses the EQ circuit in the signal chain.
2. **Low Band Frequency.** Sets the center frequency of the EQ's low-frequency band.
3. **Low Band Gain.** Sets the attenuation or boost of the EQ's low-frequency band.
4. **Low Band Shelf.** By default, the low band of the EQ is a standard peak filter with a fixed Q of 0.6. When the Shelf switch is engaged, the low band functions as a shelving filter.
5. **Mid Band Frequency.** Sets the center frequency of the EQ's mid-frequency band.
6. **Mid Band Gain.** Sets the attenuation or boost of the EQ's mid-frequency band.
7. **High Band Frequency.** Sets the center frequency of the EQ's high-frequency band.
8. **High Band Shelf.** By default, the high band of the EQ is a standard peak filter with a fixed Q of 0.6. When the Shelf switch is engaged, the high band functions as a shelving filter.
9. **High Band Gain.** Sets the attenuation or boost of the EQ's high-frequency band.

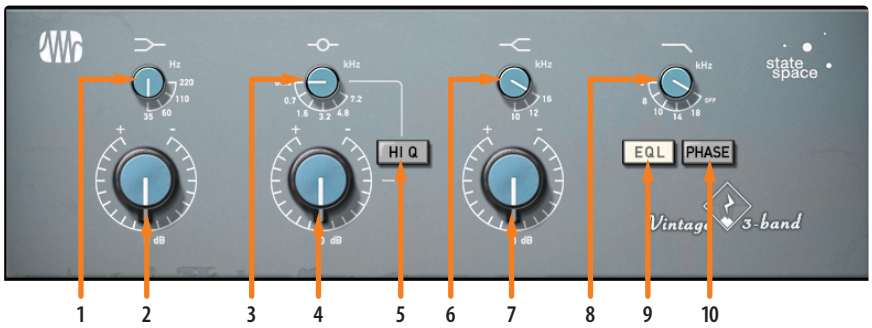
3.5.3.4 Solar 69 EQ

1. **Bass.** Sets the center frequency of the EQ's low-frequency band. The Bass parameter is both a low cut filter fixed at 50Hz with adjustable gain reduction (rotating the knob down from "flat") as well as the frequency parameter for the gain parameter (rotating the knob up from "flat").
2. **Low Band Gain.** Sets the boost of the EQ's low-frequency band.
3. **Mid.** Sets the center frequency of the EQ's mid-frequency band.
4. **Mid Band Gain.** Sets the attenuation or boost of the EQ's mid-frequency band.
5. **Peak / Trough.** When in Peak mode, the Mid Gain control will boost the midrange. When switched to Trough mode, the Mid Gain will cut the midrange and narrow the Q to function as a notch filter.
6. **10kHz.** Use the 10kHz parameter to cut or boost the fixed 10kHz frequency from -10 dB to 10 dB.
7. **Level Adjust.** Sets the output level.
8. **Polarity.** Flips the polarity 180 degrees, inverting the audio signal.
9. **EQ On / Off.** Enables / disables the EQ in the signal chain.

3.5.3.5 The Tube EQ

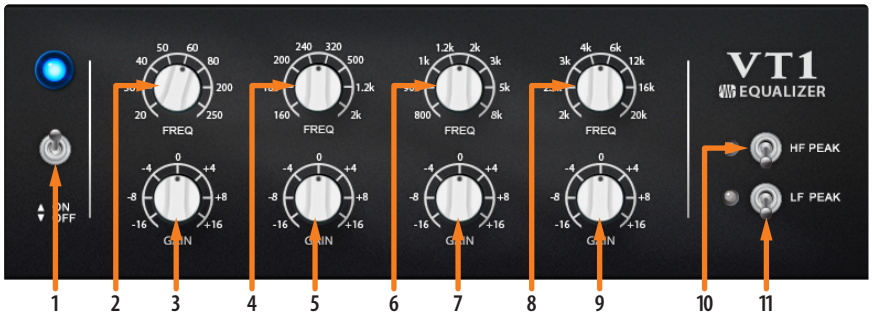
1. **Peak.** Sets the center frequency of the EQ's low-frequency band.
2. **Low Frequency kHz.** Sets the boost of the EQ's low-frequency band.
3. **Dip.** Sets the center frequency of the EQ's mid-frequency band.
4. **Mid Frequency kHz.** Sets the boost of the EQ's mid-frequency band.
5. **Peak.** Sets the center frequency of the EQ's high-frequency band.
6. **High Frequency kHz.** Sets the boost of the EQ's high-frequency band.
7. **EQ On / Off.** Enables / disables the EQ in the signal chain.

3.5.3.6 Vintage 3-band EQ



1. **Low Band Frequency.** Sets the center frequency of the EQ's low-frequency band.
2. **Low Band Gain.** Sets the attenuation or boost of the EQ's low-frequency band.
3. **Mid Band Frequency.** Sets the center frequency of the EQ's mid-frequency band.
4. **Mid Band Gain.** Sets the attenuation or boost of the EQ's mid-frequency band.
5. **Hi Q.** Switches between a wide and narrow Q.
6. **High Band Frequency.** Sets the center frequency of the EQ's high-frequency band.
7. **High Band Gain.** Sets the attenuation or boost of the EQ's high-frequency band.
8. **High Pass Filter Frequency.** Sets the frequency threshold for the High Pass Filter.
9. **EQ L.** This enables or disables the EQ in the signal chain.
10. **Phase.** Reverses the polarity by 180° at the output.

3.5.3.7 VT-1 EQ



1. **EQ On / Off.** Enables / disables the EQ in the signal chain.
2. **Low Band Frequency.** Sets the center frequency of the EQ's low-frequency band.
3. **Low Band Gain.** Sets the attenuation or boost of the EQ's low band.
4. **Low Mid Band Frequency.** Sets the center frequency of the EQ's low-mid frequency band. The low-mid band has a fixed Q of 0.6.
5. **Low Mid Band Gain.** Sets the attenuation or boost of the EQ's low-mid band.
6. **High Mid Band Frequency.** Sets the center frequency of the EQ's high-mid band. The high-mid band has a fixed Q of 0.6.
7. **High Mid Band Gain.** Sets the attenuation or boost of the EQ's high-mid band.
8. **High Band Frequency.** Sets the center frequency of the EQ's high-frequency band.
9. **High Band Gain.** Sets the attenuation or boost of the EQ's high band.
10. **HF Peak.** When the Peak switch is engaged, the high band of the EQ becomes a standard peak filter with a fixed Q of 0.6. When it is disengaged, the high band is a shelving filter.
11. **LF Peak.** When the Peak switch is engaged, the low band of the EQ becomes a standard peak filter with a fixed Q of 0.6. When it is disengaged, the low band is a shelving filter.

4 Virtual Instruments

4.1 Cinematic Lights

Cinematic Lights is a sample-based hybrid virtual instrument that enables you to layer three different sound components (based on analog modular and digital synthesis, as well as field recordings) to create unique sounds, perfect for melodies, solos, ensembles and any other cinematic sound you can imagine. Each part provides control over volume, panning, and effects, including reverb, delay, distortion and note-based arpeggiator and repeater FX. On the Part level, you can shape your sounds with the provided filter, LFOs, envelopes, Modulation Matrix, and effects.

Cinematic Lights has two pages: the Global and Part Edit page. The Global page has a minimal, sleek design, emphasizing the main elements of Cinematic Lights to simplify your experience playing the instrument. The Part Edit page offers many more sound design options, giving you more control of each part.

Before starting to experiment with Cinematic Lights, make sure you've downloaded the Sound Set that came bundled with Cinematic Lights. Cinematic Lights requires the Sound Set download for sounds and presets to be present within the instrument.

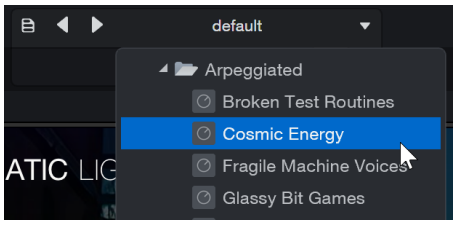
4.1.1 Global Page

The Global Page grants users easy access to the virtual instrument's global controls. From the Global Control Panel, you can quickly modify the volume, cutoff, sample shift, Attack and Release of all three layered Parts. Additionally, you have access to the Part Preset selection, the Global FX panel and virtual keyboard.

4.1.1.1 Instrument Presets

To begin experimenting with Cinematic Lights, you might consider loading one of the Instrument Presets. Once the preset is loaded, check out the sections below about adjusting the part parameters.

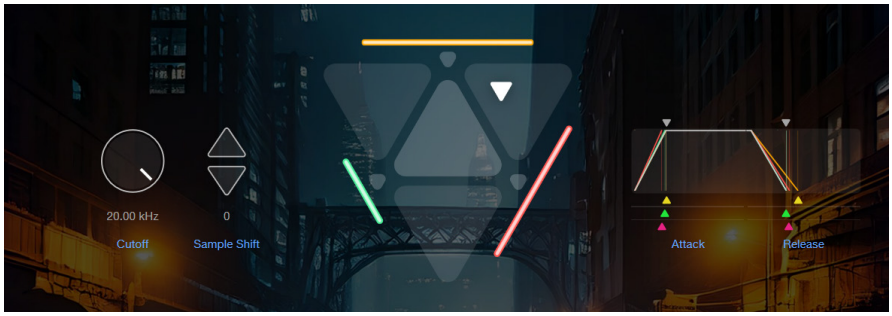
You can locate and load Instrument Presets by clicking the arrow of the instrument preset drop-down menu located at the top left corner of Cinematic Lights. Find your preset of choice and click on the preset name to load it:



Once loaded, the instrument preset is immediately playable with your MIDI controller or by clicking the virtual keyboard at the bottom of the plug-in window.

Note: *You cannot record notes using the virtual instrument keyboard. For Fender Studio Pro users who want to record notes without an external keyboard, use the QWERTY Keyboard device instead.*

4.1.1.2 Main Control Panel



The Main Control Panel on the Global page includes the Cutoff, Sample Shift, Triangle Vector Pad, and the Attack and Release parameters.

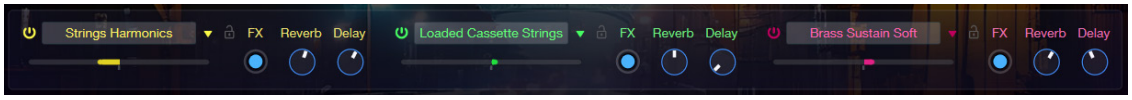
- **Cutoff:** This lets you set the corner frequency of the filter (the point in the slope of the filter at which the filter cuts incoming audio by 12 or 24 dB). In the case of a Band-Pass filter, this sets the center frequency of the passed frequency band.
- **Sample Shift:** Lets you manipulate playback speed with no change in pitch. While pitch is not affected, the range of Sample Shift is -12 or +12 semitones, which references the amount of transposition that would normally be needed to shift a sample's speed the specified amount. Shift between the semitones to explore textural and harmonic changes to the parts.
- **Triangle Vector Pad:** Provides combined volume control for all three parts. Move the small white triangle within the Vector Pad to seamlessly control the volume blend between the three parts. Each part has its own color, seen here on the color volume bars surrounding the pad.

  If you would like to assign all the volume to one part and mute the others, click onto the small gray tri angles next to the volume bar associated with the part you'd like to solo:

  When you hover your mouse over the corners of the Triangle Vector Pad, gray triangles appear. If you click onto these triangles, the two parts adjacent to the point will share the volume balance while the part on the opposite side of the triangle will be muted:

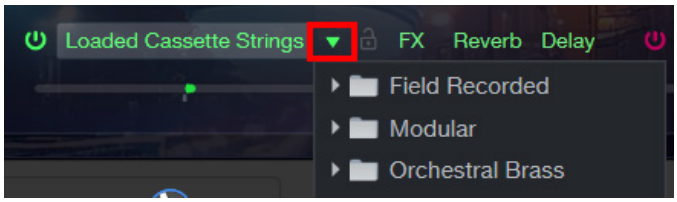
- **Attack Parameters:** Adjusts the amount of time it takes a note's level to reach its maximum once a key has been triggered. Variable from 0 to 20s.
- **Release Parameters:** Adjusts the amount of time it takes a note's level to go back to zero after a key has been released. Variable from 0 to 30s.

4.1.1.3 Part Selection



Under the main control panel, there are three drop-down menu boxes where you can choose your Part Preset sounds, enable/disable load sample only lock (see below), enable/disable the arpeggiator or repeater for each specific part, modify each individual part with Reverb and Delay sends to the global FX and set the panning for each part (set the bar underneath each part left or right to determine where it's panned).

4.1.1.4 Part Presets



From the part preset drop-down menu boxes, you can choose from a list of preset sounds. With each instrument preset, there are three predetermined part presets, but you can swap out part presets and experiment with preset layering as desired.

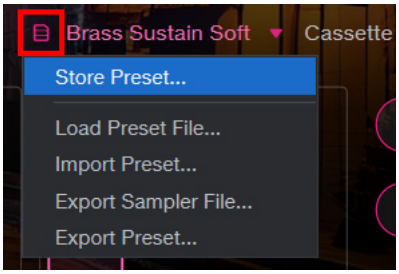
4.1.1.5 Load Samples Only Lock

The "Load Samples Only" lock is located to the right of each part preset drop down box. When the lock is engaged, any changes you make to the parameters of a part will remain if you decide to change the part's preset sound.

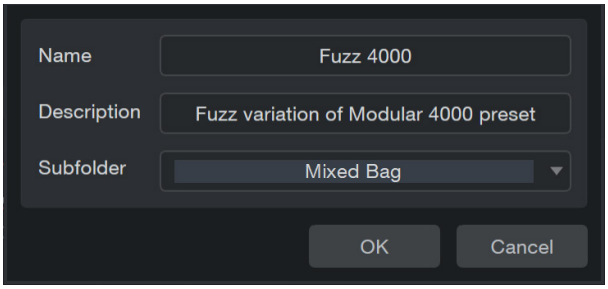
4.1.1.6 Saving Part Presets

You can save your own part presets after experimenting with the parameters. To do so, follow the directions below:

1. Press the colored menu located at the top right of the part edit page.



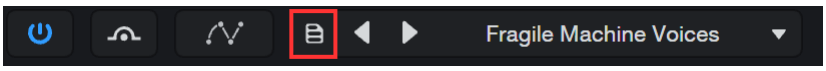
2. From the drop-down menu, select the "Store Preset" option.
3. Specify the preset information you'd like and assign it to an existing or new folder for simple organization. Once you are ready, press OK and the preset will be stored for future use.



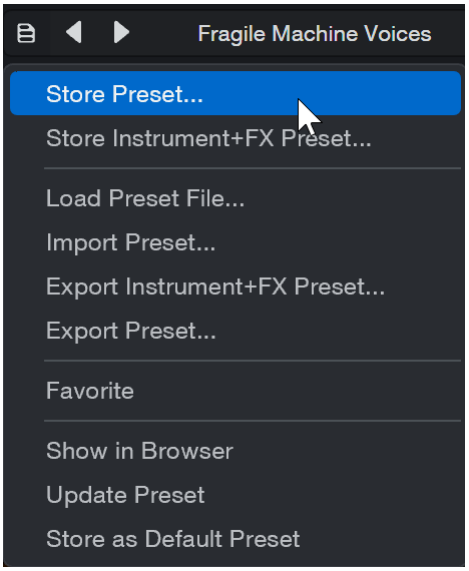
4.1.1.7 Saving Instrument Presets

Alternatively, you may wish to save a new instrument preset. When you've found a new layering of parts you'd like to store for later, you can save it as a new instrument preset by following a similar process:

1. Press onto the Preset Actions button at the top of the Cinematic Lights window.



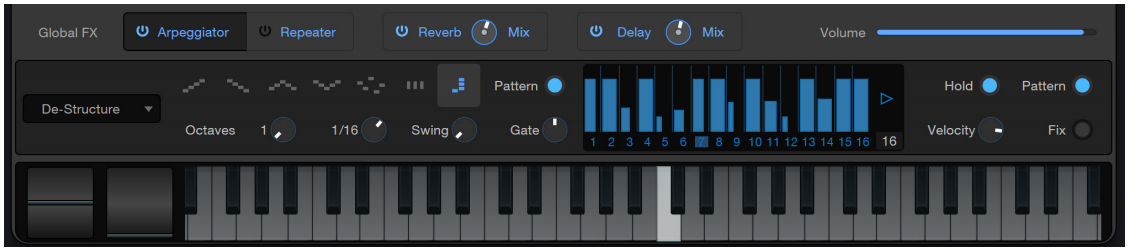
2. Select the "Store Preset" option from the drop-down menu.



3. Specify the preset information you'd like and assign it to an existing or new folder for simple organization. Once you are ready, press OK and the preset will be stored for future use.

4.1.1.8 Global FX & the Virtual Keyboard

The Global page parameters located at the bottom of the window include Global FX options, including the Arpeggiator, Repeater, two FX mix knobs, and the instrument's main volume control.



4.1.1.9 The Arpeggiator

The Arpeggiator turns chords (as well as single notes) into arpeggios—rhythmic cycles of single notes, derived from the notes currently held. You can adjust the parameters and patterns of the arpeggiator in the FX editor box that appears after enabling the Arpeggiator.

Going beyond the basics, you can use the Pattern function to create rhythmic patterns of velocity and note length that the arpeggio follows as it plays, opening many creative options for repeating musical articulations.

The following parameters and functions are available in Arpeggiator:

- **Arpeggiator Presets:** Click onto the drop down Arpeggiator Preset menu to choose from several arpeggiator templates. These presets provide you a great starting point and showcase the range of sonic possibilities that the Arpeggiator affords.
- **Arpeggio Direction:** Choose the note direction that creates the pattern you want from the following options:
  - » **Up:** The arpeggio starts at the lowest held note and travels upward through the held notes, then returns to the lowest note as the arpeggio begins again.
  - » **Down:** The arpeggio starts at the highest held note and travels downward through the held notes, then returns to the highest note as the arpeggio begins again.

- » **Up/Down:** The arpeggio starts at the lowest held note, travels upward to the highest, then travels back down to the lowest note, and the arpeggio begins again.
- » **Down/Up:** The arpeggio starts at the highest held note, travels downward to the lowest, then travels back up to the highest note, and the arpeggio begins again.
- » **Random:** The arpeggio plays the currently held notes in a random pattern.
- » **From Input:** The arpeggio pattern is derived from the order in which notes are played and held.
- » **Chord Mode:** The currently held chord (or single note) is repeated as long as it is held.

- **Pattern:** This is a 32-step pattern sequencer you can use to create repeating patterns of note velocity and gate (length) that are applied to the control output of the Arpeggiator. The Pattern area contains the following controls:
  - » **Activate Pattern:** Toggle this on or off to enable or disable the Pattern sequencer.
  - » **Pattern Sequencer:** This series of sliders is where you'll create your velocity/gate pattern. You can click and drag each step in the sequence vertically (to set velocity) and horizontally (to set gate length). 16 steps are shown at any one time. To reach the second set of 16 steps, click the right-arrow to the right of the pattern. To return to steps 1-16, click the left-arrow to the right of the pattern.
  - » **Pattern Length:** Lets you choose the length of the pattern.
- **Octave Range:** Lets you extend the range of the arpeggio by mirroring the currently held notes an octave above (at a setting of 2), the two octaves above (at 3), or the three octaves above (at 4). At 1, no mirroring occurs and the arpeggio plays according to the held notes. Setting Octave Range above 1 when in Chord Mode causes an upward-rising cycle of repeated chords, an octave apart, its length determined by the Octave Range setting.
- **Rate:** Lets you select a rhythmic value, determining the length of each arpeggio step. Range is from a bar to a 64th-note, in all triplet and dotted varieties.
- **Swing:** Lets you add swing to the timing of the arpeggio, creating a range of rhythmic feels.
- **Gate:** Acts as an overall note-length control. Lower settings create shorter notes, and higher settings create longer notes. When a Pattern is active, Gate works relative to the gate settings in the Pattern.
- **Hold:** Enable this to hold all currently played notes. Each new chord or single note that you play replaces the previous memory and holds until the next note input is received.
- **Velocity:** Acts as an overall note-velocity control. When a Pattern is active, Velocity works relative to the velo city settings in the Pattern.
- **Pattern/Fix:** Switches Choose Pattern to allow Velocity data from the Pattern to control note velocity (in tandem with the Velocity control). Choose Fix to set a fixed velocity for all notes, with the Velocity control.

4.1.1.10 The Repeater

The Repeater creates rhythmic repetitions of the notes you play. These repetitions can be simple copies of incoming notes, or they can change in velocity, gate length, and pitch as the pattern plays. You can adjust the parameters of the repeater in the FX editor box that appears after enabling the Repeater.

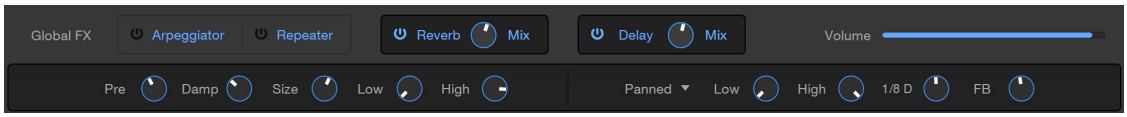
The following parameters and features are available:

- **Repeater Presets:** Click onto the drop down Repeater Preset menu to choose from several repeater templates. These presets provide you a great starting point and showcase the range of sonic possibilities that the repeater affords.
- **Rate:** Lets you set the rate of repetition. When Sync is enabled, you can choose a tempo-synced rhythmic value between one bar and one 64th-note, in all triplet and dotted varieties. When Sync is disabled, you can choose a repetition frequency between 2 and 25 Hz (repetitions per second).
- **Steps:** Select your desired number of repetitions (and sequencer steps), from just two steps to 32.
- **Sync:** Enable Sync to snap the Rate control to rhythmic values in sync with Song tempo. Disable to set repetition speed in Hz.
- **Pitch Scale:** Turn this to introduce a sloping change in pitch in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, you get a range of downward pitch shift slopes. Turn to the right for upward slopes.
- **Velocity Level:** Acts as an overall velocity level control, scaling every step's velocity, relative to its custom setting.
- **(Velocity Relative to) Input:** Enable this to derive overall repetition velocity from note input, attenuated or boosted by the setting of the Velocity Level control.
- **Velocity Scale:** Turn this to introduce a sloping change in velocity in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, a downward slope is introduced. Turn to the right for an upward slope.
- **Gate:** Acts as an overall gate length control, scaling every step's gate length, relative to its custom setting.

- **Gate Scale:** Turn this to introduce a sloping change in gate length in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, each step in the sequence is shortened more than the last. Turn to the right, and each step is lengthened more than the last.
- **Sequencer:** This lets you specify velocity, gate, and pitch values for each step in the sequence of repetitions. The following controls are available for each step:
  - » **Level/Gate Slider:** With Individual Velocity and Gate enabled, you can click and drag the upper edge of this slider up or down to set note velocity for the current step. You can also click and drag the right edge of the slider left or right to set note gate length for the current step. With Individual Velocity and Gate disabled, moving velocity or gate sliders manipulates that setting for all steps (equivalent to turning the Velocity Level and Gate knobs).
  - » **Pitch:** With Individual Pitch enabled, you can click and drag this slider up or down to apply a positive or negative pitch transposition to the current step. With Individual Pitch disabled, moving a pitch slider changes pitch transposition for all steps simultaneously. At 0, no transposition occurs.
- **Individual Velocity:** Engage to enable individual setting of note velocity and gate length for each step.
- **Individual Pitch:** Engage to enable individual setting of note pitch for each step.

4.1.1.11 Reverb and Delay

Beside the Repeater and Arpeggiator options on the Global FX bar, Reverb and Delay Knobs provide you control over the global reverb and delay return mix. Click onto either reverb or delay boxes and their respective parameters will appear below where you can finetune the effect to your liking.



Reverb

- **Pre-delay:** Adjust the pre-delay time. Variable from 0 ms to 500 ms. Pre-delay is the amount of time before the first reverberated signals are heard.
- **Damp:** Adjusts the relative amount of damping (attenuation of the upper frequencies) of the reverberated signal. Variable from 0 to 100%.
- **Reverb Size:** This control lets you set the length of reverberation from the moment a sound starts, in a range between 100 ms and 10 seconds.
- **Low and High:** These controls let you set the cutoff frequencies of the provided high-pass and low-pass filters, which affect only the reverb signal.

Delay

- **Ping-Pong Mode:** Use this drop down menu to choose from several ping-pong delay modes.
- **Low and High:** These controls let you set the cutoff frequencies of the provided high-pass and low-pass filters, which affect only the delayed signal.
- **Delay Time:** This control lets you specify the length of the delay effect, in rhythmic values (such as 1/8th-note or 16th-note) relative to the song's tempo.
- **Feedback:** This control lets you set the amount of affected signal that is fed back into the Delay effect. At zero, there is just one repeat. As you increase the value, the trail of repeats grows.

4.1.1.12 Volume

The Volume Bar Adjust the output volume of the virtual instrument.

4.1.1.13 Virtual Keyboard

The virtual keyboard lets you easily click to play notes or manipulate the Pitch and Mod wheels, while auditioning or editing patches when you're away from a MIDI keyboard. The keyboard display also shows you which notes are currently being played.

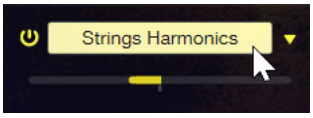
The Bend wheel parameters can be edited by clicking onto the arrow beside the pitch wheel (see screenshot below). The Bend parameters let you set the pitch bend range of the Pitch wheel in semitones. The upper value sets up-bend range, while the lower value sets down-bend range. The Pitch range parameter sets the range for the per note pitch.

Please note that you can't record notes using this virtual keyboard. For Fender Studio Pro users who want to record notes without an external keyboard, use the QWERTY Keyboard device instead.

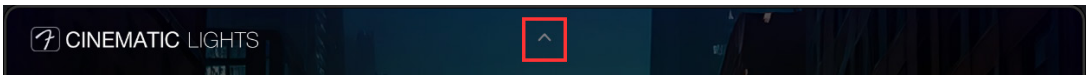
4.1.2 Part Edit Page

Shape each part to your liking by experimenting within the Part Edit page. The Part Edit page gives you access to additional LFO, Envelope, Modulation, EQ, and other sound parameters. There are two ways to access the Part Edit page:

1. Click onto the title box of the part you would like to edit.



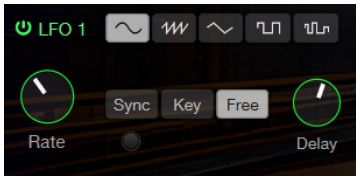
2. Click onto the arrow at the top center of the Global page. The Part Edit page of the most recently edited part will appear.



Note: *To quickly identify which part is selected, the Part Edit page will be illuminated in the same color as the selected part sound. Simply click onto the desired part title box and the Part Edit page will change color to match that of the part you chose.*

4.1.2.1 LFO

LFO stands for Low Frequency Oscillator, and each part has two of them. LFOs create slow-moving regular cycles of control signals that are useful for modulating other parameters over time. One common example is the way many keyboard patches respond when you move the Mod Wheel up from zero; the pitch of the oscillators wavers up and down in an expressive manner, much like the sound of vocal vibrato. This is simply an LFO modulating oscillator pitch to a degree set by the position of the mod wheel.



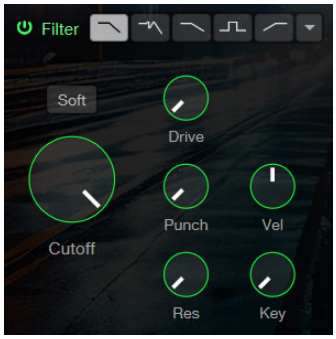
LFO 1 and 2 have identical controls, so the following explanations apply to both:

- **Bypass:** Click the [LFO 1] or [LFO 2] button to turn the selected LFO on or off.
- **LFO Type:** Choose between Sine, Triangle, Sawtooth, Square, and Random shapes for the oscillation of the LFO.
- **Rate:** Sets the rate at which the LFO oscillates, from inaudibly low (0.01 Hz) for long, sweeping changes, all the way to higher ranges (up to 8 kHz) useful for FM and AM techniques. When the LFO's [Sync] button is engaged, Rate can be set in terms of rhythmic values relative to Song tempo (such as 1/8th-note or 1/4-note)

- **Sync:** Engage this option to enable setting LFO Rate to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate by Hz.
- **Key:** Engage this option to bind LFO speed to incoming note pitch. Higher notes result in higher LFO speeds, while lower notes result in lower LFO speeds.
- **Free:** Engage this option to let the LFO run continuously, resulting in a differing LFO start point for each note played. Disengage to restart the LFO waveform at the start of each note.
- **Delay:** This control lets you specify an amount of time (in milliseconds) for the LFO to wait before becoming active after a note is played. This lets you do things like adding a bit of expression to held notes, or creating layers of modulation that start at different points in each note by setting distinct Delay values for each LFO.

4.1.2.2 Filter

Cinematic Lights offers a versatile multi-mode Filter, which lets you shape and enhance your sounds. The filter is often one of the most important defining elements to the sound of an analog synthesizer.



The Filter offers the following controls:

- **Bypass:** Click the [Filter] button to turn the filter on or off.
- **Filter Mode:** Choose from the following filter modes, each with its own sound-shaping characteristics.
  - » **LP 24 dB Ladder:** This mode emulates a classic 24-dB-per-octave low-pass filter based on a transistor-ladder configuration, as found in many classic synthesizers. This type of filter allows frequencies below the chosen Cutoff frequency to pass through, cutting frequencies above Cutoff at a rate of 24 decibels per octave—a fairly aggressive slope.
  - » **LP 24 dB Zero:** This is a 24-dB-per-octave low-pass filter, based on a zero-delay-feedback architecture that closely models the tone and modulation behavior of analog filters.
  - » **LP 12 dB Ladder:** This is a low-pass filter with a 12-dB-per-octave curve, which cuts frequencies less aggressively than the 24 dB filters.
  - » **BP 12 dB Ladder:** This is a high-pass and low-pass filter in series, known collectively as a band-pass filter. It allows a selected band of frequencies to pass through, then cuts frequencies above and below that band at a rate of 12 decibels per octave.
  - » **HP 12 dB Ladder:** This is a high-pass filter with a 12-dB-per-octave slope. This lets frequencies above the chosen Cutoff frequency pass through, while cutting frequencies below Cutoff at a rate of 12 decibels per octave.
  - » **LP 12 State, BP 12 State, HP 12 State, Eco Filter:** These are a set of simple, clean digital filter models, in low-pass, band-pass, high-pass, and eco (low-CPU low-pass) modes. You can access these filter types in the drop-down menu at the end of the row of Filter Mode switches.
- **Soft:** This control lets you switch between two differing analog-modeled processing circuits within the filter. Engage Soft for a mellower, darker tone. Disengage it for a brighter, more aggressive sound.
- **Cutoff:** This lets you set the corner frequency of the filter—the point in the slope of the filter at which the filter cuts incoming audio by 3 dB. In the case of the Band-Pass filter, this sets the center frequency of the passed frequency band.
- **Drive:** This lets you specify an amount of filter overdrive, to add fullness and saturation artifacts to your sound.
- **Punch:** This control lets you add a range of percussive attack to the start of each note. At the lowest setting, dynamics are unchanged. At higher settings, the sound becomes more aggressive and more readily pops through the mix.
- **Velocity (Vel):** This control sets the relationship between incoming note Velocity and filter Cutoff. When set at the center, velocity does not affect cutoff. When moved to the right, the cutoff rises as note velocity increases. When moved to the left, cutoff lowers as note velocity increases.
- **Resonance (Res):** This lets you set the amount of resonance in the filter, which is an emphasis centered on the chosen cutoff frequency.
  - » At lower settings, the filter cuts frequencies smoothly. As you increase Res, the emphasis at the cutoff frequency becomes more pronounced, able to mimic resonances such as those in voices or acoustic instruments, as well as many classic synthesis effects.
  - » At the highest settings, the filter can self-oscillate, emitting a pitched tone at the current cutoff frequency. This filter oscillation can be treated somewhat like an extra oscillator, especially in conjunction with the Key parameter.

- **Key:** This control sets the relationship between incoming note pitch and filter cutoff. In physical instruments, higher notes tend to produce higher harmonics, brightening slightly as you go up the scale.
  - » On a synthesized instrument, if the filter stays static, setting the proper tone in the lower note ranges may cause inappropriate dullness in the higher notes. Therefore, with the Key parameter, we can compensate for this and create a more natural-sounding range of timbres up and down the keyboard.
  - » When Key is set all the way to the left, the filter is unaffected by note pitch. In the middle, cutoff follows note pitch subtly, allowing high notes to shine. When set all the way right, filter cutoff follows note pitch closely in a relative fashion, moving upward and downward in semitone values as notes are received. This lets you use the filter as an additional pitched oscillator or resonator when filter Res is set high.

4.1.2.3 Envelopes

Envelope generators are a vital part of sound synthesis, giving us the ability to shape the amplitude and timbre of our sounds within the time-scale of each note. Cinematic Lights has two envelope generators per part, labeled "Amp Envelope" (so named because it is hard-wired to amplitude) and "Filter Envelope".

Both envelope modules are triggered when a note is played. Each envelope then outputs a control signal that follows the shape set by the following controls:

- **Delay (Δ - Filter Envelope only):** This control lets you specify a length of time (in ms) for the Env to pause before starting its attack phase after a note is played. This can assist in creating evolving sounds, where cycles of modulation occur at differing times over the length of a note.
- **Attack (A):** This control lets you set the time required for the envelope to go from zero (silence) to full amplitude, in a range from 0 ms to 20 seconds.
- **Decay (D):** This control lets you set the time required to drop from full amplitude to the sustain level, in a range from 0 ms to 20 seconds.
- **Sustain (S):** This control lets you set the signal level that is maintained from the end of the decay period, until the key is released, in a range from -∞ dB (silence) to 0.0 dB (full amplitude).
- **Release (R):** This control lets you set the time required to fall back to silence after the key is released, in a range from 0 ms to 30 seconds.

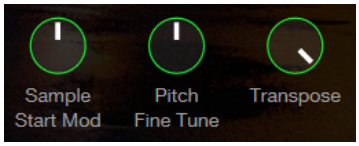
4.1.2.4 Envelope Graphical Display

Each envelope has a corresponding graphical display that represents the shape created by the settings of its parameters. There are handles on the corners and slopes of each envelope that you can click and drag, letting you shape the ADSR envelope and the curve between its points visually.

If you wish to lengthen any phase of the envelope beyond the time limits of the current display, simply drag the point toward the right of the graph, and the time scale adjusts to properly display the new setting.

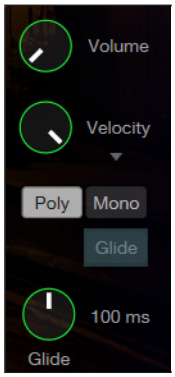
4.1.2.5 Pitch Controls

The pitch control knobs let you manipulate the way that Cinematic Lights plays the parts in the currently loaded preset.



- **Sample Start Mod:** Lets you specify an amount of negative or positive velocity-controlled offset, applied to the point in the part which playback begins. At settings above and below the default of 0, lower-velocity notes trigger a smaller amount of sample offset, and higher-velocity notes trigger a larger amount of offset.
- **Pitch Fine Tune:** Adjusts the tuning, in cents, for the selected pad. Variable from -100 to 100 cents.
- **Transpose:** Adjusts the transposition in semitones for the selected pad. Variable from -48 to +48.

4.1.2.6 Part Global Settings



The following Global parameters let you configure Cinematic Lights's overall behavior and capabilities specific to the part you're editing:

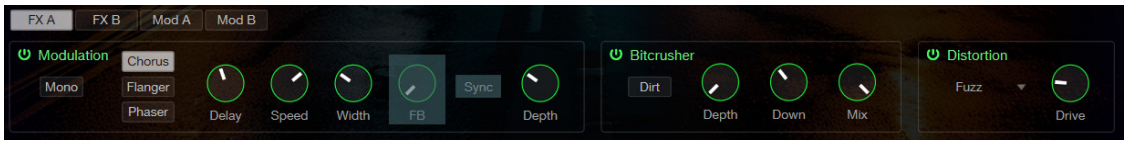
- **Volume:** This control lets you set the total output volume, in a range from -∞ dB (silence) to +10.0 dB (ten decibels above unity gain).
- **Velocity:** This control lets you set the degree to which Cinematic Lights's volume is affected by note velocity, from zero (no velocity sensitivity) to 1.0 (full velocity sensitivity).
- **Poly, Mono, and Glide:** Enable Poly mode to allow polyphonic playing (more than one note at a time). Enable Mono mode to play just one note at a time. When in Mono mode, you can enable Glide to cause the pitch to sweep smoothly from that of the currently held note to that of the next note, when played legato (one note played while the previous note is held). The Glide knob lets you set the rate of pitch change over time, from 1 ms to 1 second.

4.1.2.7 FX and Mods

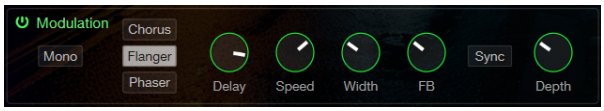
Cinematic Lights offers built-in effects processors to add dimension to your sounds. They are arranged in four banks: FX A, FX B, Mod A, and Mod B. You can enable or disable each effect by clicking its name.

FX A

The first bank, FX A, includes Modulation (Chorus, Flanger, and Phaser), Bitcrusher, and Distortion effects.



Modulation



This processor creates time-based modulation effects. Choose from the following modes by clicking the [Chorus], [Flanger], or [Phaser] button.

Chorus: This processor creates effects similar to that of multiple identical instruments playing the same part simultaneously. The synth signal is fed through a short, modulated delay, which is then mixed with the dry signal. Chorus offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Delay:** This control lets you set the length of the modulated delay. Higher settings create full-bodied chorusing effects, while lower settings create more pronounced harmonics, akin to the effects of a Flanger.
- **Speed:** This control lets you set the speed at which the delay line is modulated. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the delay line is modulated. Lower settings produce subtler chorusing effects, while higher settings produce more pronounced changes in timbre over time.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the chorused signal (all the way right).

Flanger: This processor creates resonant, hollow-sounding sweeping effects. The synth signal is fed through a short, modulated delay, which is mixed with the dry signal. While similar to the workings of a Chorus effect, Flangers get their signature sound by employing smaller delay times than those used in chorusing, combined with a feedback system that can add extra resonance to the sweep. Flanger offers the following controls:

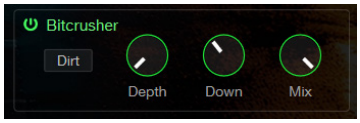
- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Delay:** This control lets you set the length of the modulated delay (in ms), which changes the pitch of the resultant resonance. Higher settings create lower-pitched resonance, while lower settings create resonances at a higher pitch.
- **Speed:** This control lets you set the speed at which the delay line is modulated. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the delay line is modulated. Lower settings produce subtler flanging effects, while higher settings produce more pronounced changes in timbre over time.
- **Feedback (FB):** This control lets you set the amount of output signal to feed back into the Flanger. Higher amounts of Feedback add to the resonance of the sweeping effect.
- **Sync:** Engage this option to enable setting Flanger modulation speed to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate on a continuous scale.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the flanged signal (all the way right).

Phaser: This processor creates dreamy, otherworldly sweeping effects. The synth signal is fed through a series of all pass filters that alter its phase. When mixed with the dry signal, this creates a series of peaks and valleys in the frequency response that changes depending on the degree of phase shift applied. Phaser offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Shift:** This control lets you specify the amount of phase shift to apply. Lower settings focus the phasing effect in the lower frequencies, while higher settings focus the effect in higher frequencies.
- **Speed:** This control lets you set the speed of modulation applied to the phase shift amount. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the phase shift amount is modulated. Lower settings produce subtler effects, while higher settings produce more pronounced changes in timbre over time.
- **Feedback (FB):** This control lets you set the amount of output signal to feed back into the Phaser. Higher amounts of Feedback add to the resonance of the sweeping effect.
- **Sync:** Engage this option to enable setting Phaser modulation speed to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate on a continuous scale.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the phase-shifted signal (all the way right).

Bitcrusher

Perfect for audio abuse, Bitcrusher combines overdrive, bit-depth reduction, downsampling and clipping into a single plug-in. Bit depth reduction and downsampling are both digital resolution-reduction techniques, but each has its own sonic effect. When used in combination, they create a wide variety of tonal options.

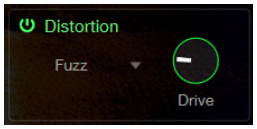


The following parameters are available in Bitcrusher:

- **Dirt:** Enable this to introduce a high-frequency instability in the Bit Depth reduction effect. Good for creating aggressive sounds.
- **Depth:** Lets you specify the level of Bit Depth reduction to apply, from 24-bit to 1-bit.
- **Down:** Lets you specify the level of downsampling to apply.
- **Mix:** Lets you blend between the dry (0%) and effected (100%) signals.

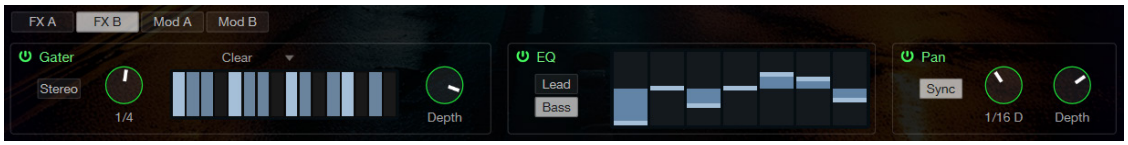
Distortion

This processor adds grit and character to your sounds. The distortion FX parameters within Cinematic Lights are quite simple: Use the dropdown menu to choose from a variety of distortion types, from fizzy transistor fuzzes to thick, warm tube overdrives. Then, set the amount of distortion with the Drive knob.



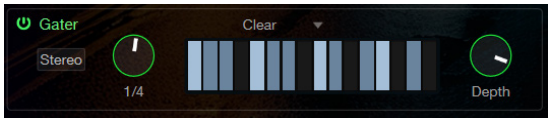
FX B

The second bank, FX B, includes Gater, Lead/Bass EQ, and Panning FX options.



Gater

The Gater will add a rhythmic gating effect, able to create a series of syncopated breaks in the synth signal. A variety of presets are provided, each with a different rhythmic gating pattern. However, the fun really begins when you create your own.



The Gater offers the following controls:

- **Stereo:** Engaging this option creates a separate beat grid for each side of the stereo field. When engaged, you'll see two rows of beat steps, the top row specifying gate steps for the left channel, and the bottom row gating the right channel.
- **Beats:** This control lets you set the length of the gating cycle, in rhythmic values (such as 1 bar or 1/2-note) relative to Song tempo. For example, at a setting of 1 bar, the 16 steps in the cycle repeat every bar, effectively representing 16th-notes. At a 1/2-note setting, the 16 steps repeat each half-bar, representing 32nd-note values.
- **Gate Presets:** There are nine provided gate presets to choose from.

- **Gate Steps:** This grid lets you specify which steps in the cycle let signals pass through, and which gate the signal to silence. Click on a step to enable or disable gating for that step.
- **Depth:** This control lets you blend between the gated and dry signals, allowing for rhythmic gating effects while retaining the continuity of the synth sound.

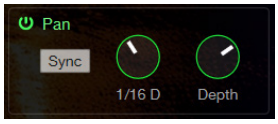
EQ

FX B includes a graphic equalizer effect, perfect for quick tonal shaping. Set the EQ bands to emphasize or attenuate bands of frequencies to suit your needs. When a band is in the center of its range, it neither adds or subtracts. When moved above the center, it emphasizes the chosen frequency. Moved below the center, it attenuates that frequency.

Choose between Lead mode, with frequencies chosen to suit aggressive, up-front sounds, or Bass mode, with wider ranging frequencies that work better for basses and mellower chordal parts.

Pan

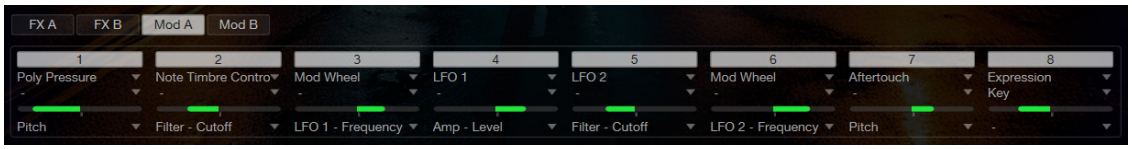
This is an auto-pan effect, which pans the synth signal left and right over time.



Pan offers the following controls:

- **Sync:** Enable this option to set pan speed to a rhythmic value (such as 1/4-note or 16th-note) relative to Song tempo. Disable this option to set pan speed along a continuous range.
- **Speed:** This control lets you set the speed at which the signal is panned left and right.
- **Depth:** This control lets you set the degree to which the signal is panned. Lower settings give a subtly panned effect, while higher settings pan the signal more radically, all the way to fully left and right in each cycle.

4.1.2.8 Modulation Matrix



If you would like to map additional modulation parameters, Cinematic Lights provides 16 configurable modulation routings, split between two banks of eight (Mod A and Mod B). Modulation signals can be routed from a selection of incoming MIDI controller signals, including MPE data (such as Pitch Bend, Mod Wheel, Aftertouch, Poly Pressure, Note Timbre, and Pitch Control), modulation generators (such as the LFOs and envelopes), or the pitch or velocity of played notes.

These modulation signals can be used to vary most of the parameters throughout Cinematic Lights, including modulation sources themselves (such as LFO 2 modulating the rate of LFO 1, or the Decay of Env 2). The following list describes the functions of each option within the Modulation Matrix, from top to bottom:

- **Bypass:** Each modulation slot has a bypass button at the top, which lets you enable or disable the flow of the modulation signal.
- **Main Source Selector:** Use this selector to specify the modulation signal you'd like to route to the Modifier Selector or Target.
- **Modifier Selector ("Via"):** Similar to the Main Source Selector, you can use this selector to specify the modulation signal you'd like to route to the Modifier Selector or Target. Additionally, you can use this selector in conjunction with the Main Source to further experiment with routing.
  - » For example, if you assign a modulation source to the input selector only, that signal is routed directly to the chosen destination. In some cases, you'll want to govern the flow of one mod source before it reaches its destination, using the signal from another mod source. On the other hand, you may want to control the output level of LFO 1 (routed to a parameter such as oscillator pitch) with the Mod Wheel. In this case, you'd choose Mod Wheel with the input selector, and LFO 1 with the modifier selector below.
- **Slider:** The slider underneath the selectors controls the amplitude and polarity of the modulation signal. Set at its center, no modulation occurs. Move the handle right of center to send an increasing amount of the modulation signal, at its normal (positive) polarity, to the chosen destination. Move it left of center to send the signal to its destination with a negative value.
  - » If the parameter you wish to modulate is set to a high value, you may want to send a negative modulation signal to it, driving the setting downward and causing more audible effects. Positive-going modulation signals are more efficient when modulating parameters set to low values.
- **Target Selector:** A selector at the bottom of each modulation slot lets you choose the destination of the chosen modulation signals. Signals can be mapped to any of the available destinations within Cinematic Lights's Part Edit page.

4.2 Deep Flight One

Deep Flight One is a sample-based hybrid virtual instrument that enables you to layer three different sound components (based on analog modular and digital synthesis, as well as field recordings) to create unique sounds, perfect for melodies, solos, ensembles and any other cinematic sound you can imagine. Each part provides control over volume, panning, and effects, including reverb, delay, distortion and note-based arpeggiator and repeater FX. On the Part level, you can shape your sounds with the provided filter, LFOs, envelopes, Modulation Matrix, and effects.

Deep Flight One has two pages: the Global and Part Edit page. The Global page has a minimal, sleek design, emphasizing the main elements of Deep Flight One to simplify your experience playing the instrument. The Part Edit page offers many more sound design options, giving you more control of each part.

Before starting to experiment with Deep Flight One, make sure you've downloaded the Sound Set that came bundled with Deep Flight One. Deep Flight One requires the Sound Set download for sounds and presets to be present within the instrument.

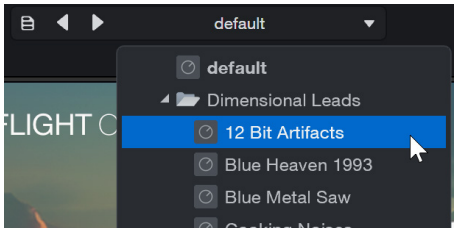
4.2.1 Global Page

The Global Page grants users easy access to the virtual instrument's global controls. From the Global Control Panel, you can quickly modify the volume, cutoff, sample shift, Attack and Release of all three layered Parts. Additionally, you have access to the Part Preset selection, the Global FX panel and virtual keyboard.

4.2.1.1 Instrument Presets

To begin experimenting with Deep Flight One, you might consider loading one of the Instrument Presets. Once the preset is loaded, check out the sections below about adjusting the part parameters.

You can locate and load Instrument Presets by clicking the arrow of the instrument preset drop-down menu located at the top left corner of Deep Flight One. Find your preset of choice and click on the preset name to load it:



Once loaded, the instrument preset is immediately playable with your MIDI controller or by clicking the virtual keyboard at the bottom of the plug-in window.

Note: *You cannot record notes using the virtual instrument keyboard. For Fender Studio Pro users who want to record notes without an external keyboard, use the QWERTY Keyboard device instead.*

4.2.1.2 Main Control Panel

The Main Control Panel on the Global page includes the Cutoff, Sample Shift, Triangle Vector Pad, and the Attack and Release parameters.

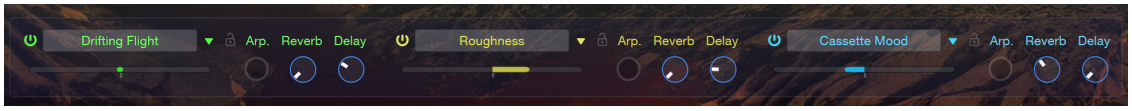
- **Cutoff:** This lets you set the corner frequency of the filter (the point in the slope of the filter at which the filter cuts incoming audio by 12 or 24 dB). In the case of a Band-Pass filter, this sets the center frequency of the passed frequency band.
- **Sample Shift:** Lets you manipulate playback speed with no change in pitch. While pitch is not affected, the range of Sample Shift is -12 or +12 semitones, which references the amount of transposition that would normally be needed to shift a sample's speed the specified amount. Shift between the semitones to explore textural and harmonic changes to the parts.
- **Triangle Vector Pad:** Provides combined volume control for all three parts. Move the small white triangle within the Vector Pad to seamlessly control the volume blend between the three parts. Each part has its own color, seen here on the color volume bars surrounding the pad.

  If you would like to assign all the volume to one part and mute the others, click onto the small gray tri angles next to the volume bar associated with the part you'd like to solo:

  When you hover your mouse over the corners of the Triangle Vector Pad, gray triangles appear. If you click onto these triangles, the two parts adjacent to the point will share the volume balance while the part on the opposite side of the triangle will be muted:

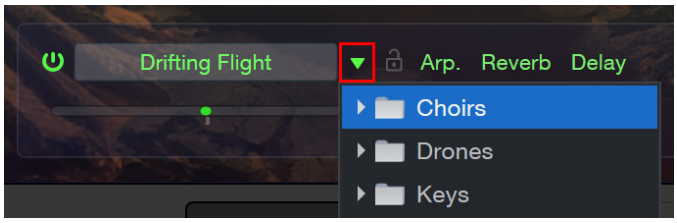
- **Attack Parameters:** Adjusts the amount of time it takes a note's level to reach its maximum once a key has been triggered. Variable from 0 to 20s.
- **Release Parameters:** Adjusts the amount of time it takes a note's level to go back to zero after a key has been released. Variable from 0 to 30s.

4.2.1.3 Part Selection



Under the main control panel, there are three drop-down menu boxes where you can choose your Part Preset sounds, enable/disable load sample only lock (see below), enable/disable the arpeggiator or repeater for each specific part, modify each individual part with Reverb and Delay sends to the global FX and set the panning for each part (set the bar underneath each part left or right to determine where it's panned).

4.2.1.4 Part Presets



From the part preset drop-down menu boxes, you can choose from a list of preset sounds. With each instrument preset, there are three predetermined part presets, but you can swap out part presets and experiment with preset layering as desired.

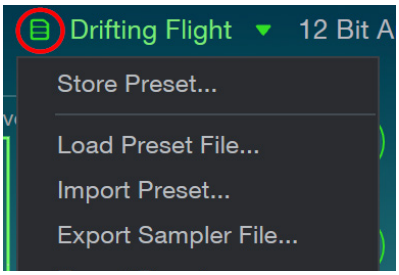
4.2.1.5 Load Samples Only Lock

The "Load Samples Only" lock is located to the right of each part preset drop down box. When the lock is engaged, any changes you make to the parameters of a part will remain if you decide to change the part's preset sound.

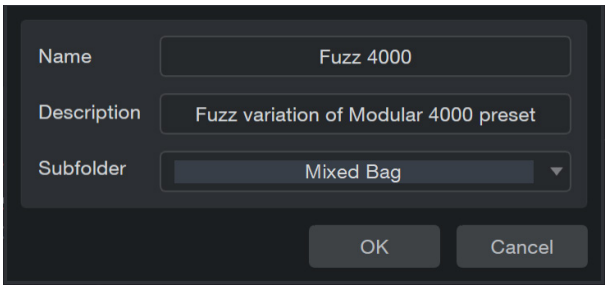
4.2.1.6 Saving Part Presets

You can save your own part presets after experimenting with the parameters. To do so, follow the directions below:

1. Press the colored menu located at the top right of the part edit page.



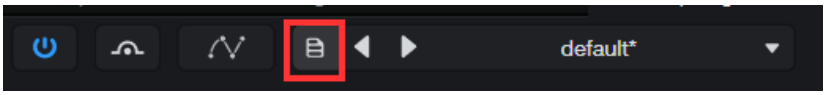
2. From the drop-down menu, select the "Store Preset" option.
3. Specify the preset information you'd like and assign it to an existing or new folder for simple organization. Once you are ready, press OK and the preset will be stored for future use.



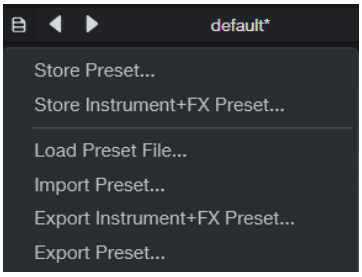
4.2.1.7 Saving Instrument Presets

Alternatively, you may wish to save a new instrument preset. When you've found a new layering of parts you'd like to store for later, you can save it as a new instrument preset by following a similar process:

1. Press onto the Preset Actions button at the top of the Deep Flight One window.



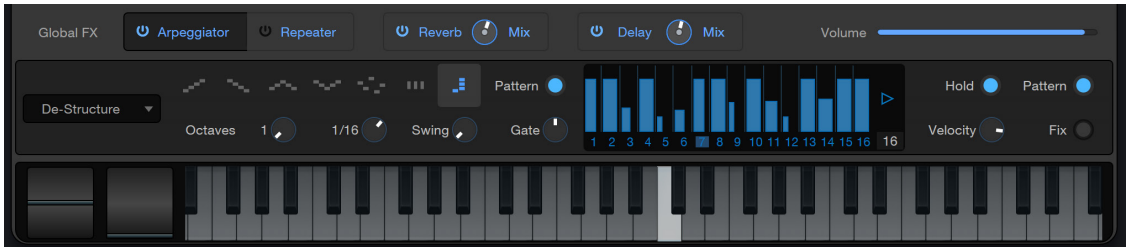
2. Select the "Store Preset" option from the drop-down menu.



3. Specify the preset information you'd like and assign it to an existing or new folder for simple organization. Once you are ready, press OK and the preset will be stored for future use.

4.2.1.8 Global FX & the Virtual Keyboard

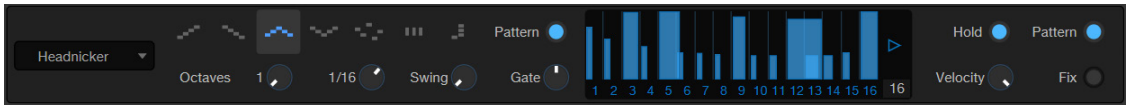
The Global page parameters located at the bottom of the window include Global FX options, including the Arpeggiator, Repeater, two FX mix knobs, and the instrument's main volume control.



4.2.1.9 The Arpeggiator

The Arpeggiator turns chords (as well as single notes) into arpeggios—rhythmic cycles of single notes, derived from the notes currently held. You can adjust the parameters and patterns of the arpeggiator in the FX editor box that appears after enabling the Arpeggiator.

Going beyond the basics, you can use the Pattern function to create rhythmic patterns of velocity and note length that the arpeggio follows as it plays, opening many creative options for repeating musical articulations.



The following parameters and functions are available in Arpeggiator:

- **Arpeggiator Presets:** Click onto the drop down Arpeggiator Preset menu to choose from several arpeggiator templates. These presets provide you a great starting point and showcase the range of sonic possibilities that the Arpeggiator affords.
- **Arpeggio Direction:** Choose the note direction that creates the pattern you want from the following options:
  - » **Up:** The arpeggio starts at the lowest held note and travels upward through the held notes, then returns to the lowest note as the arpeggio begins again.
  - » **Down:** The arpeggio starts at the highest held note and travels downward through the held notes, then returns to the highest note as the arpeggio begins again.
  - » **Up/Down:** The arpeggio starts at the lowest held note, travels upward to the highest, then travels back down to the lowest note, and the arpeggio begins again.
  - » **Down/Up:** The arpeggio starts at the highest held note, travels downward to the lowest, then travels back up to the highest note, and the arpeggio begins again.
  - » **Random:** The arpeggio plays the currently held notes in a random pattern.
  - » **From Input:** The arpeggio pattern is derived from the order in which notes are played and held.
  - » **Chord Mode:** The currently held chord (or single note) is repeated as long as it is held.
- **Pattern:** This is a 32-step pattern sequencer you can use to create repeating patterns of note velocity and gate (length) that are applied to the control output of the Arpeggiator. The Pattern area contains the following controls:
  - » **Activate Pattern:** Toggle this on or off to enable or disable the Pattern sequencer.
  - » **Pattern Sequencer:** This series of sliders is where you'll create your velocity/gate pattern. You can click and drag each step in the sequence vertically (to set velocity) and horizontally (to set gate length). 16 steps are shown at any one time. To reach the second set of 16 steps, click the right-arrow to the right of the pattern. To return to steps 1-16, click the left-arrow to the right of the pattern.
  - » **Pattern Length:** Lets you choose the length of the pattern.

- **Octave Range:** Lets you extend the range of the arpeggio by mirroring the currently held notes an octave above (at a setting of 2), the two octaves above (at 3), or the three octaves above (at 4). At 1, no mirroring occurs and the arpeggio plays according to the held notes. Setting Octave Range above 1 when in Chord Mode causes an upward-rising cycle of repeated chords, an octave apart, its length determined by the Octave Range setting.
- **Rate:** Lets you select a rhythmic value, determining the length of each arpeggio step. Range is from a bar to a 64th-note, in all triplet and dotted varieties.
- **Swing:** Lets you add swing to the timing of the arpeggio, creating a range of rhythmic feels.
- **Gate:** Acts as an overall note-length control. Lower settings create shorter notes, and higher settings create longer notes. When a Pattern is active, Gate works relative to the gate settings in the Pattern.
- **Hold:** Enable this to hold all currently played notes. Each new chord or single note that you play replaces the previous memory and holds until the next note input is received.
- **Velocity:** Acts as an overall note-velocity control. When a Pattern is active, Velocity works relative to the velo city settings in the Pattern.
- **Pattern/Fix:** Switches Choose Pattern to allow Velocity data from the Pattern to control note velocity (in tandem with the Velocity control). Choose Fix to set a fixed velocity for all notes, with the Velocity control.

4.2.1.10 The Repeater

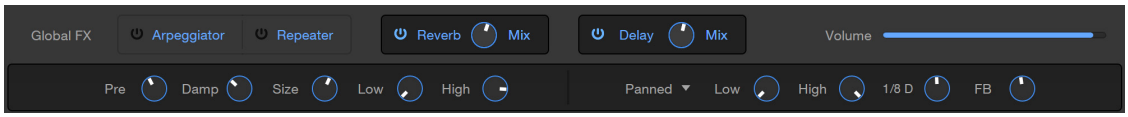
The Repeater creates rhythmic repetitions of the notes you play. These repetitions can be simple copies of incoming notes, or they can change in velocity, gate length, and pitch as the pattern plays. You can adjust the parameters of the repeater in the FX editor box that appears after enabling the Repeater.

The following parameters and features are available:

- **Repeater Presets:** Click onto the drop down Repeater Preset menu to choose from several repeater templates. These presets provide you a great starting point and showcase the range of sonic possibilities that the repeater affords.
- **Rate:** Lets you set the rate of repetition. When Sync is enabled, you can choose a tempo-synced rhythmic value between one bar and one 64th-note, in all triplet and dotted varieties. When Sync is disabled, you can choose a repetition frequency between 2 and 25 Hz (repetitions per second).
- **Steps:** Select your desired number of repetitions (and sequencer steps), from just two steps to 32.
- **Sync:** Enable Sync to snap the Rate control to rhythmic values in sync with Song tempo. Disable to set repetition speed in Hz.
- **Pitch Scale:** Turn this to introduce a sloping change in pitch in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, you get a range of downward pitch shift slopes. Turn to the right for upward slopes.
- **Velocity Level:** Acts as an overall velocity level control, scaling every step's velocity, relative to its custom setting.
- **(Velocity Relative to) Input:** Enable this to derive overall repetition velocity from note input, attenuated or boosted by the setting of the Velocity Level control.
- **Velocity Scale:** Turn this to introduce a sloping change in velocity in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, a downward slope is introduced. Turn to the right for an upward slope.
- **Gate:** Acts as an overall gate length control, scaling every step's gate length, relative to its custom setting.
- **Gate Scale:** Turn this to introduce a sloping change in gate length in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, each step in the sequence is shortened more than the last. Turn to the right, and each step is lengthened more than the last.
- **Sequencer:** This lets you specify velocity, gate, and pitch values for each step in the sequence of repetitions. The following controls are available for each step:
  - » **Level/Gate Slider:** With Individual Velocity and Gate enabled, you can click and drag the upper edge of this slider up or down to set note velocity for the current step. You can also click and drag the right edge of the slider left or right to set note gate length for the current step. With Individual Velocity and Gate disabled, moving velocity or gate sliders manipulates that setting for all steps (equivalent to turning the Velocity Level and Gate knobs).
  - » **Pitch:** With Individual Pitch enabled, you can click and drag this slider up or down to apply a positive or negative pitch transposition to the current step. With Individual Pitch disabled, moving a pitch slider changes pitch transposition for all steps simultaneously. At 0, no transposition occurs.
- **Individual Velocity:** Engage to enable individual setting of note velocity and gate length for each step.
- **Individual Pitch:** Engage to enable individual setting of note pitch for each step.

4.2.1.11 Reverb and Delay

Beside the Repeater and Arpeggiator options on the Global FX bar, Reverb and Delay Knobs provide you control over the global reverb and delay return mix. Click onto either reverb or delay boxes and their respective parameters will appear below where you can finetune the effect to your liking.



Reverb

- **Pre-delay:** Adjust the pre-delay time. Variable from 0 ms to 500 ms. Pre-delay is the amount of time before the first reverberated signals are heard.
- **Damp:** Adjusts the relative amount of damping (attenuation of the upper frequencies) of the reverberated signal. Variable from 0 to 100%.
- **Reverb Size:** This control lets you set the length of reverberation from the moment a sound starts, in a range between 100 ms and 10 seconds.
- **Low and High:** These controls let you set the cutoff frequencies of the provided high-pass and low-pass filters, which affect only the reverb signal.

Delay

- **Ping-Pong Mode:** Use this drop down menu to choose from several ping-pong delay modes.
- **Low and High:** These controls let you set the cutoff frequencies of the provided high-pass and low-pass filters, which affect only the delayed signal.
- **Delay Time:** This control lets you specify the length of the delay effect, in rhythmic values (such as 1/8th-note or 16th-note) relative to the song's tempo.
- **Feedback:** This control lets you set the amount of affected signal that is fed back into the Delay effect. At zero, there is just one repeat. As you increase the value, the trail of repeats grows.

4.2.1.12 Volume

The Volume Bar Adjust the output volume of the virtual instrument.

4.2.1.13 Virtual Keyboard

The virtual keyboard lets you easily click to play notes or manipulate the Pitch and Mod wheels, while auditioning or editing patches when you're away from a MIDI keyboard. The keyboard display also shows you which notes are currently being played.

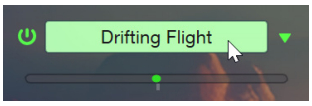
The Bend wheel parameters can be edited by clicking onto the arrow beside the pitch wheel (see screenshot below). The Bend parameters let you set the pitch bend range of the Pitch wheel in semitones. The upper value sets up-bend range, while the lower value sets down-bend range. The Pitch range parameter sets the range for the per note pitch.

Please note that you can't record notes using this virtual keyboard. For Fender Studio Pro users who want to record notes without an external keyboard, use the QWERTY Keyboard device instead.

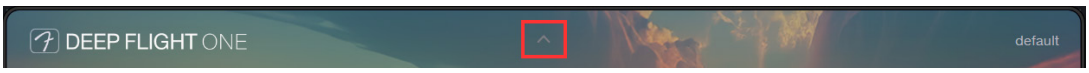
4.2.2 Part Edit Page

Shape each part to your liking by experimenting within the Part Edit page. The Part Edit page gives you access to additional LFO, Envelope, Modulation, EQ, and other sound parameters. There are two ways to access the Part Edit page:

1. Click onto the title box of the part you would like to edit.



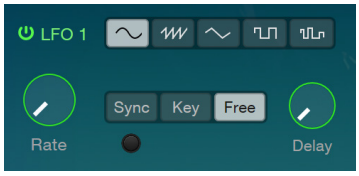
2. Click onto the arrow at the top center of the Global page. The Part Edit page of the most recently edited part will appear.



Note: *To quickly identify which part is selected, the Part Edit page will be illuminated in the same color as the selected part sound. Simply click onto the desired part title box and the Part Edit page will change color to match that of the part you chose.*

4.2.2.1 LFO

LFO stands for Low Frequency Oscillator, and each part has two of them. LFOs create slow-moving regular cycles of control signals that are useful for modulating other parameters over time. One common example is the way many keyboard patches respond when you move the Mod Wheel up from zero; the pitch of the oscillators wavers up and down in an expressive manner, much like the sound of vocal vibrato. This is simply an LFO modulating oscillator pitch to a degree set by the position of the mod wheel.

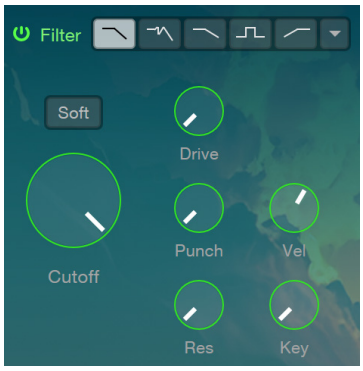


LFO 1 and 2 have identical controls, so the following explanations apply to both:

- **Bypass:** Click the [LFO 1] or [LFO 2] button to turn the selected LFO on or off.
- **LFO Type:** Choose between Sine, Triangle, Sawtooth, Square, and Random shapes for the oscillation of the LFO.
- **Rate:** Sets the rate at which the LFO oscillates, from inaudibly low (0.01 Hz) for long, sweeping changes, all the way to higher ranges (up to 8 kHz) useful for FM and AM techniques. When the LFO's [Sync] button is engaged, Rate can be set in terms of rhythmic values relative to Song tempo (such as 1/8th-note or 1/4-note).
- **Sync:** Engage this option to enable setting LFO Rate to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate by Hz.
- **Key:** Engage this option to bind LFO speed to incoming note pitch. Higher notes result in higher LFO speeds, while lower notes result in lower LFO speeds.
- **Free:** Engage this option to let the LFO run continuously, resulting in a differing LFO start point for each note played. Disengage to restart the LFO waveform at the start of each note.
- **Delay:** This control lets you specify an amount of time (in milliseconds) for the LFO to wait before becoming active after a note is played. This lets you do things like adding a bit of expression to held notes, or creating layers of modulation that start at different points in each note by setting distinct Delay values for each LFO.

4.2.2.2 Filter

Deep Flight One offers a versatile multi-mode Filter, which lets you shape and enhance your sounds. The filter is often one of the most important defining elements to the sound of an analog synthesizer.



The Filter offers the following controls:

- **Bypass:** Click the [Filter] button to turn the filter on or off.
- **Filter Mode:** Choose from the following filter modes, each with its own sound-shaping characteristics.
  - » **LP 24 dB Ladder:** This mode emulates a classic 24-dB-per-octave low-pass filter based on a transistor-ladder configuration, as found in many classic synthesizers. This type of filter allows frequencies below the chosen Cutoff frequency to pass through, cutting frequencies above Cutoff at a rate of 24 decibels per octave—a fairly aggressive slope.
  - » **LP 24 dB Zero:** This is a 24-dB-per-octave low-pass filter, based on a zero-delay-feedback architecture that closely models the tone and modulation behavior of analog filters.
  - » **LP 12 dB Ladder:** This is a low-pass filter with a 12-dB-per-octave curve, which cuts frequencies less aggressively than the 24 dB filters.
  - » **BP 12 dB Ladder:** This is a high-pass and low-pass filter in series, known collectively as a band-pass filter. It allows a selected band of frequencies to pass through, then cuts frequencies above and below that band at a rate of 12 decibels per octave.
  - » **HP 12 dB Ladder:** This is a high-pass filter with a 12-dB-per-octave slope. This lets frequencies above the chosen Cutoff frequency pass through, while cutting frequencies below Cutoff at a rate of 12 decibels per octave.
  - » **LP 12 State, BP 12 State, HP 12 State, Eco Filter:** These are a set of simple, clean digital filter models, in low-pass, band-pass, high-pass, and eco (low-CPU low-pass) modes. You can access these filter types in the drop-down menu at the end of the row of Filter Mode switches.
- **Soft:** This control lets you switch between two differing analog-modeled processing circuits within the filter. Engage Soft for a mellower, darker tone. Disengage it for a brighter, more aggressive sound.
- **Cutoff:** This lets you set the corner frequency of the filter—the point in the slope of the filter at which the filter cuts incoming audio by 3 dB. In the case of the Band-Pass filter, this sets the center frequency of the passed frequency band.
- **Drive:** This lets you specify an amount of filter overdrive, to add fullness and saturation artifacts to your sound.
- **Punch:** This control lets you add a range of percussive attack to the start of each note. At the lowest setting, dynamics are unchanged. At higher settings, the sound becomes more aggressive and more readily pops through the mix.
- **Velocity (Vel):** This control sets the relationship between incoming note Velocity and filter Cutoff. When set at the center, velocity does not affect cutoff. When moved to the right, the cutoff rises as note velocity increases. When moved to the left, cutoff lowers as note velocity increases.
- **Resonance (Res):** This lets you set the amount of resonance in the filter, which is an emphasis centered on the chosen cutoff frequency.
  - » At lower settings, the filter cuts frequencies smoothly. As you increase Res, the emphasis at the cutoff frequency becomes more pronounced, able to mimic resonances such as those in voices or acoustic instruments, as well as many classic synthesis effects.
  - » At the highest settings, the filter can self-oscillate, emitting a pitched tone at the current cutoff frequency. This filter oscillation can be treated somewhat like an extra oscillator, especially in conjunction with the Key parameter.
- **Key:** This control sets the relationship between incoming note pitch and filter cutoff. In physical instruments, higher notes tend to produce higher harmonics, brightening slightly as you go up the scale.
  - » On a synthesized instrument, if the filter stays static, setting the proper tone in the lower note ranges may cause inappropriate dullness in the higher notes. Therefore, with the Key parameter, we can compensate for this and create a more natural-sounding range of timbres up and down the keyboard.
  - » When Key is set all the way to the left, the filter is unaffected by note pitch. In the middle, cutoff follows note pitch subtly, allowing high notes to shine. When set all the way right, filter cutoff follows note pitch closely in a relative fashion, moving upward and downward in semitone values as notes are received. This lets you use the filter as an additional pitched oscillator or resonator when filter Res is set high.

4.2.2.3 Envelopes

Envelope generators are a vital part of sound synthesis, giving us the ability to shape the amplitude and timbre of our sounds within the time-scale of each note. Deep Flight One has two envelope generators per part, labeled "Amp Envelope" (so named because it is hard-wired to amplitude) and "Filter Envelope".

Both envelope modules are triggered when a note is played. Each envelope then outputs a control signal that follows the shape set by the following controls:

- **Delay (Δ - Filter Envelope only):** This control lets you specify a length of time (in ms) for the Env to pause before starting its attack phase after a note is played. This can assist in creating evolving sounds, where cycles of modulation occur at differing times over the length of a note.
- **Attack (A):** This control lets you set the time required for the envelope to go from zero (silence) to full amplitude, in a range from 0 ms to 20 seconds.
- **Decay (D):** This control lets you set the time required to drop from full amplitude to the sustain level, in a range from 0 ms to 20 seconds.
- **Sustain (S):** This control lets you set the signal level that is maintained from the end of the decay period, until the key is released, in a range from -∞ dB (silence) to 0.0 dB (full amplitude).
- **Release (R):** This control lets you set the time required to fall back to silence after the key is released, in a range from 0 ms to 30 seconds.

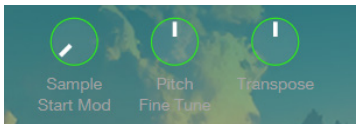
4.2.2.4 Envelope Graphical Display

Each envelope has a corresponding graphical display that represents the shape created by the settings of its parameters. There are handles on the corners and slopes of each envelope that you can click and drag, letting you shape the ADSR envelope and the curve between its points visually.

If you wish to lengthen any phase of the envelope beyond the time limits of the current display, simply drag the point toward the right of the graph, and the time scale adjusts to properly display the new setting.

4.2.2.5 Pitch Controls

The pitch control knobs let you manipulate the way that Deep Flight One plays the parts in the currently loaded preset.



- **Sample Start Mod:** Lets you specify an amount of negative or positive velocity-controlled offset, applied to the point in the part which playback begins. At settings above and below the default of 0, lower-velocity notes trigger a smaller amount of sample offset, and higher-velocity notes trigger a larger amount of offset.
- **Pitch Fine Tune:** Adjusts the tuning, in cents, for the selected pad. Variable from -100 to 100 cents.
- **Transpose:** Adjusts the transposition in semitones for the selected pad. Variable from -48 to +48.

4.2.2.6 Part Global Settings

The following Global parameters let you configure Deep Flight One's overall behavior and capabilities specific to the part you're editing:

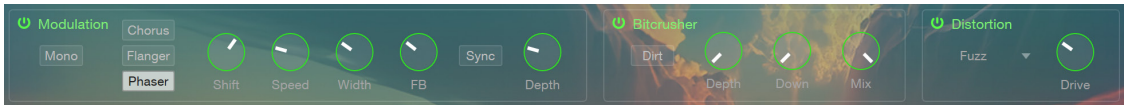
- **Volume:** This control lets you set the total output volume, in a range from -∞ dB (silence) to +10.0 dB (ten decibels above unity gain).
- **Velocity:** This control lets you set the degree to which Deep Flight One's volume is affected by note velocity, from zero (no velocity sensitivity) to 1.0 (full velocity sensitivity).
- **Poly, Mono, and Glide:** Enable Poly mode to allow polyphonic playing (more than one note at a time). Enable Mono mode to play just one note at a time. When in Mono mode, you can enable Glide to cause the pitch to sweep smoothly from that of the currently held note to that of the next note, when played legato (one note played while the previous note is held). The Glide knob lets you set the rate of pitch change over time, from 1 ms to 1 second.

4.2.2.7 FX and Mods

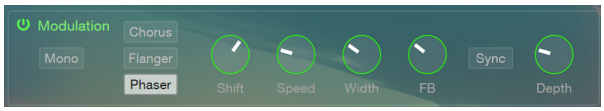
Deep Flight One offers built-in effects processors to add dimension to your sounds. They are arranged in four banks: FX A, FX B, Mod A, and Mod B. You can enable or disable each effect by clicking its name.

FX A

The first bank, FX A, includes Modulation (Chorus, Flanger, and Phaser), Bitcrusher, and Distortion effects.



Modulation



This processor creates time-based modulation effects. Choose from the following modes by clicking the [Chorus], [Flanger], or [Phaser] button.

Chorus: This processor creates effects similar to that of multiple identical instruments playing the same part simultaneously. The synth signal is fed through a short, modulated delay, which is then mixed with the dry signal. Chorus offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Delay:** This control lets you set the length of the modulated delay. Higher settings create full-bodied chorusing effects, while lower settings create more pronounced harmonics, akin to the effects of a Flanger.
- **Speed:** This control lets you set the speed at which the delay line is modulated. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the delay line is modulated. Lower settings produce subtler chorusing effects, while higher settings produce more pronounced changes in timbre over time.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the chorused signal (all the way right).

Flanger: This processor creates resonant, hollow-sounding sweeping effects. The synth signal is fed through a short, modulated delay, which is mixed with the dry signal. While similar to the workings of a Chorus effect, Flangers get their signature sound by employing smaller delay times than those used in chorusing, combined with a feedback system that can add extra resonance to the sweep. Flanger offers the following controls:

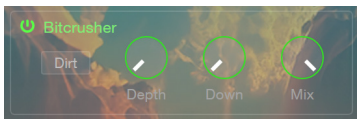
- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Delay:** This control lets you set the length of the modulated delay (in ms), which changes the pitch of the resultant resonance. Higher settings create lower-pitched resonance, while lower settings create resonances at a higher pitch.
- **Speed:** This control lets you set the speed at which the delay line is modulated. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the delay line is modulated. Lower settings produce subtler flanging effects, while higher settings produce more pronounced changes in timbre over time.
- **Feedback (FB):** This control lets you set the amount of output signal to feed back into the Flanger. Higher amounts of Feedback add to the resonance of the sweeping effect.
- **Sync:** Engage this option to enable setting Flanger modulation speed to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate on a continuous scale.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the flanged signal (all the way right).

Phaser: This processor creates dreamy, otherworldly sweeping effects. The synth signal is fed through a series of all pass filters that alter its phase. When mixed with the dry signal, this creates a series of peaks and valleys in the frequency response that changes depending on the degree of phase shift applied. Phaser offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Shift:** This control lets you specify the amount of phase shift to apply. Lower settings focus the phasing effect in the lower frequencies, while higher settings focus the effect in higher frequencies.
- **Speed:** This control lets you set the speed of modulation applied to the phase shift amount. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the phase shift amount is modulated. Lower settings produce subtler effects, while higher settings produce more pronounced changes in timbre over time.
- **Feedback (FB):** This control lets you set the amount of output signal to feed back into the Phaser. Higher amounts of Feedback add to the resonance of the sweeping effect.
- **Sync:** Engage this option to enable setting Phaser modulation speed to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate on a continuous scale.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the phase-shifted signal (all the way right).

Bitcrusher

Perfect for audio abuse, Bitcrusher combines overdrive, bit-depth reduction, downsampling and clipping into a single plug-in. Bit depth reduction and downsampling are both digital resolution-reduction techniques, but each has its own sonic effect. When used in combination, they create a wide variety of tonal options.

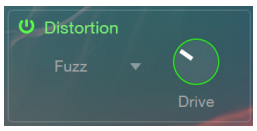


The following parameters are available in Bitcrusher:

- **Dirt:** Enable this to introduce a high-frequency instability in the Bit Depth reduction effect. Good for creating aggressive sounds.
- **Depth:** Lets you specify the level of Bit Depth reduction to apply, from 24-bit to 1-bit.
- **Down:** Lets you specify the level of downsampling to apply.
- **Mix:** Lets you blend between the dry (0%) and effected (100%) signals.

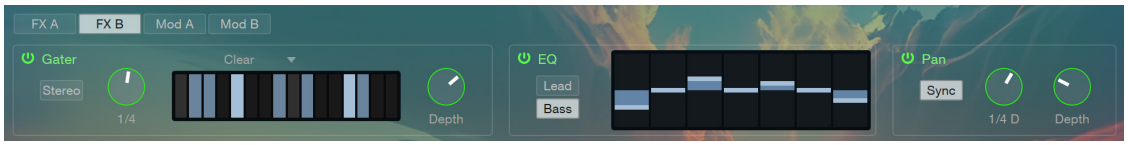
Distortion

This processor adds grit and character to your sounds. The distortion FX parameters within Deep Flight One are quite simple: Use the dropdown menu to choose from a variety of distortion types, from fizzy transistor fuzzes to thick, warm tube overdrives. Then, set the amount of distortion with the Drive knob.



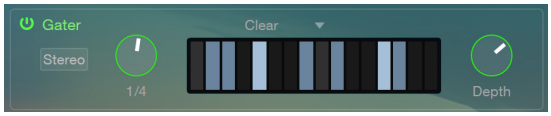
FX B

The second bank, FX B, includes Gater, Lead/Bass EQ, and Panning FX options.



Gater

The Gater will add a rhythmic gating effect, able to create a series of syncopated breaks in the synth signal. A variety of presets are provided, each with a different rhythmic gating pattern. However, the fun really begins when you create your own.



The Gater offers the following controls:

- **Stereo:** Engaging this option creates a separate beat grid for each side of the stereo field. When engaged, you'll see two rows of beat steps, the top row specifying gate steps for the left channel, and the bottom row gating the right channel.
- **Beats:** This control lets you set the length of the gating cycle, in rhythmic values (such as 1 bar or 1/2-note) relative to Song tempo. For example, at a setting of 1 bar, the 16 steps in the cycle repeat every bar, effectively representing 16th-notes. At a 1/2-note setting, the 16 steps repeat each half-bar, representing 32nd-note values.
- **Gate Presets:** There are nine provided gate presets to choose from.
- **Gate Steps:** This grid lets you specify which steps in the cycle let signals pass through, and which gate the signal to silence. Click on a step to enable or disable gating for that step.
- **Depth:** This control lets you blend between the gated and dry signals, allowing for rhythmic gating effects while retaining the continuity of the synth sound.

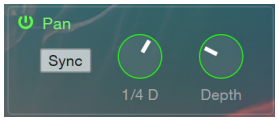
EQ

FX B includes a graphic equalizer effect, perfect for quick tonal shaping. Set the EQ bands to emphasize or attenuate bands of frequencies to suit your needs. When a band is in the center of its range, it neither adds or subtracts. When moved above the center, it emphasizes the chosen frequency. Moved below the center, it attenuates that frequency.

Choose between Lead mode, with frequencies chosen to suit aggressive, up-front sounds, or Bass mode, with wider ranging frequencies that work better for basses and mellower chordal parts.

Pan

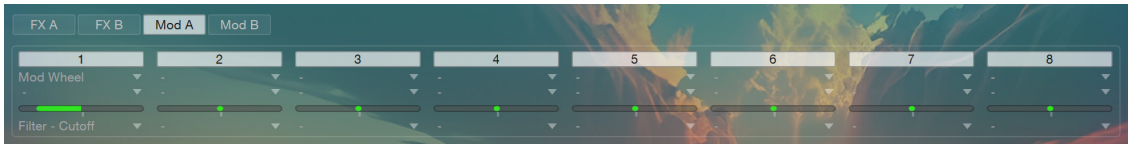
This is an auto-pan effect, which pans the synth signal left and right over time.



Pan offers the following controls:

- **Sync:** Enable this option to set pan speed to a rhythmic value (such as 1/4-note or 16th-note) relative to Song tempo. Disable this option to set pan speed along a continuous range.
- **Speed:** This control lets you set the speed at which the signal is panned left and right.
- **Depth:** This control lets you set the degree to which the signal is panned. Lower settings give a subtly panned effect, while higher settings pan the signal more radically, all the way to fully left and right in each cycle.

4.2.2.8 Modulation Matrix



If you would like to map additional modulation parameters, Deep Flight One provides 16 configurable modulation routings, split between two banks of eight (Mod A and Mod B). Modulation signals can be routed from a selection of incoming MIDI controller signals, including MPE data (such as Pitch Bend, Mod Wheel, Aftertouch, Poly Pressure, Note Timbre, and Pitch Control), modulation generators (such as the LFOs and envelopes), or the pitch or velocity of played notes.

These modulation signals can be used to vary most of the parameters throughout Deep Flight One, including modulation sources themselves (such as LFO 2 modulating the rate of LFO 1, or the Decay of Env 2). The following list describes the functions of each option within the Modulation Matrix, from top to bottom:

- **Bypass:** Each modulation slot has a bypass button at the top, which lets you enable or disable the flow of the modulation signal.
- **Main Source Selector:** Use this selector to specify the modulation signal you'd like to route to the Modifier Selector or Target.
- **Modifier Selector ("Via"):** Similar to the Main Source Selector, you can use this selector to specify the modulation signal you'd like to route to the Modifier Selector or Target. Additionally, you can use this selector in conjunction with the Main Source to further experiment with routing.
  - » For example, if you assign a modulation source to the input selector only, that signal is routed directly to the chosen destination. In some cases, you'll want to govern the flow of one mod source before it reaches its destination, using the signal from another mod source. On the other hand, you may want to control the output level of LFO 1 (routed to a parameter such as oscillator pitch) with the Mod Wheel. In this case, you'd choose Mod Wheel with the input selector, and LFO 1 with the modifier selector below.
- **Slider:** The slider underneath the selectors controls the amplitude and polarity of the modulation signal. Set at its center, no modulation occurs. Move the handle right of center to send an increasing amount of the modulation signal, at its normal (positive) polarity, to the chosen destination. Move it left of center to send the signal to its destination with a negative value.
  - » If the parameter you wish to modulate is set to a high value, you may want to send a negative modulation signal to it, driving the setting downward and causing more audible effects. Positive-going modulation signals are more efficient when modulating parameters set to low values.
- **Target Selector:** A selector at the bottom of each modulation slot lets you choose the destination of the chosen modulation signals. Signals can be mapped to any of the available destinations within Deep Flight One's Part Edit page.

4.3 Lead Architect

Lead Achitect is a sample-based hybrid virtual instrument that enables you to layer three different sound components (based on analog modular and digital synthesis, as well as field recordings) to create unique sounds, perfect for melodies, solos, ensembles and any other cinematic sound you can imagine. Each part provides control over volume, panning, and effects, including reverb, delay, distortion and note-based arpeggiator and repeater FX. On the Part level, you can shape your sounds with the provided filter, LFOs, envelopes, Modulation Matrix, and effects.

Lead Achitect has two pages: the Global and Part Edit page. The Global page has a minimal, sleek design, emphasizing the main elements of Lead Achitect to simplify your experience playing the instrument. The Part Edit page offers many more sound design options, giving you more control of each part.

Before starting to experiment with Lead Achitect, make sure you've downloaded the Sound Set that came bundled with Lead Achitect. Lead Achitect requires the Sound Set download for sounds and presets to be present within the instrument.

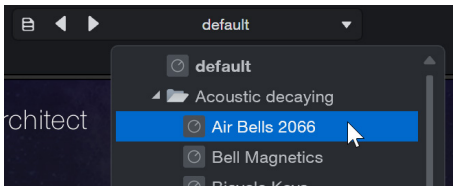
4.3.1 Global Page

The Global Page grants users easy access to the virtual instrument's global controls. From the Global Control Panel, you can quickly modify the volume, cutoff, sample shift, Attack and Release of all three layered Parts. Additionally, you have access to the Part Preset selection, the Global FX panel and virtual keyboard.

4.3.1.1 Instrument Presets

To begin experimenting with Lead Achitect, you might consider loading one of the Instrument Presets. Once the preset is loaded, check out the sections below about adjusting the part parameters.

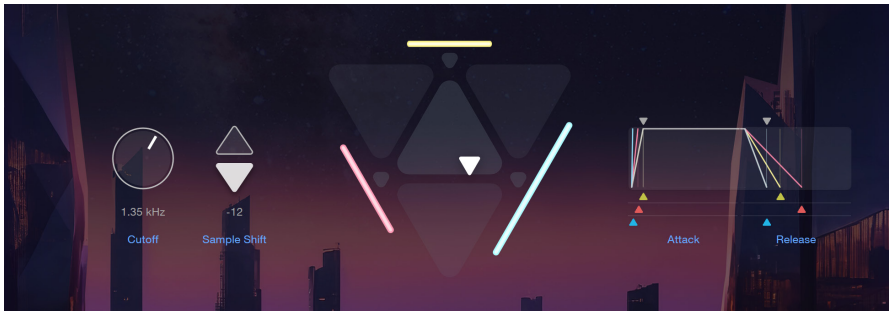
You can locate and load Instrument Presets by clicking the arrow of the instrument preset drop-down menu located at the top left corner of Lead Achitect. Find your preset of choice and click on the preset name to load it:



Once loaded, the instrument preset is immediately playable with your MIDI controller or by clicking the virtual keyboard at the bottom of the plug-in window.

Note: *You cannot record notes using the virtual instrument keyboard. For Fender Studio Pro users who want to record notes without an external keyboard, use the QWERTY Keyboard device instead.*

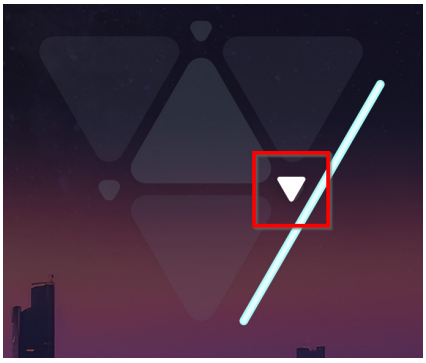
4.3.1.2 Main Control Panel



The Main Control Panel on the Global page includes the Cutoff, Sample Shift, Triangle Vector Pad, and the Attack and Release parameters.

- **Cutoff:** This lets you set the corner frequency of the filter (the point in the slope of the filter at which the filter cuts incoming audio by 12 or 24 dB). In the case of a Band-Pass filter, this sets the center frequency of the passed frequency band.
- **Sample Shift:** Lets you manipulate playback speed with no change in pitch. While pitch is not affected, the range of Sample Shift is -12 or +12 semitones, which references the amount of transposition that would normally be needed to shift a sample's speed the specified amount. Shift between the semitones to explore textural and harmonic changes to the parts.
- **Triangle Vector Pad:** Provides combined volume control for all three parts. Move the small white triangle within the Vector Pad to seamlessly control the volume blend between the three parts. Each part has its own color, seen here on the color volume bars surrounding the pad.

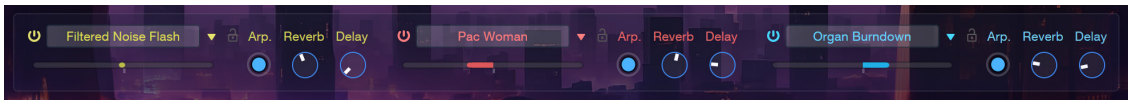
  If you would like to assign all the volume to one part and mute the others, click onto the small gray tri angles next to the volume bar associated with the part you'd like to solo:

  When you hover your mouse over the corners of the Triangle Vector Pad, gray triangles appear. If you click onto these triangles, the two parts adjacent to the point will share the volume balance while the part on the opposite side of the triangle will be muted:

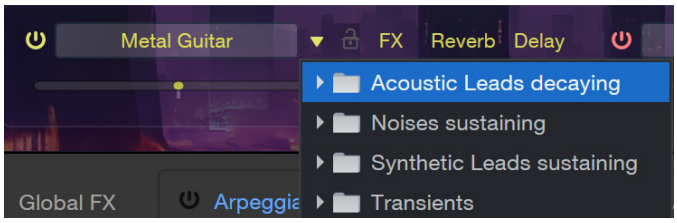
- **Attack Parameters:** Adjusts the amount of time it takes a note's level to reach its maximum once a key has been triggered. Variable from 0 to 20s.
- **Release Parameters:** Adjusts the amount of time it takes a note's level to go back to zero after a key has been released. Variable from 0 to 30s.

4.3.1.3 Part Selection



Under the main control panel, there are three drop-down menu boxes where you can choose your Part Preset sounds, enable/disable load sample only lock (see below), enable/disable the arpeggiator or repeater for each specific part, modify each individual part with Reverb and Delay sends to the global FX and set the panning for each part (set the bar underneath each part left or right to determine where it's panned).

4.3.1.4 Part Presets



From the part preset drop-down menu boxes, you can choose from a list of preset sounds. With each instrument preset, there are three predetermined part presets, but you can swap out part presets and experiment with preset layering as desired.

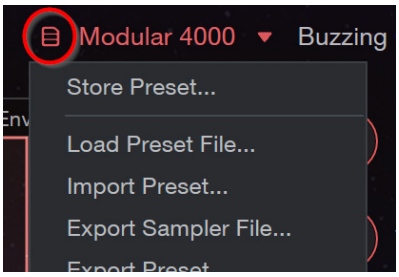
4.3.1.5 Load Samples Only Lock

The "Load Samples Only" lock is located to the right of each part preset drop down box. When the lock is engaged, any changes you make to the parameters of a part will remain if you decide to change the part's preset sound.

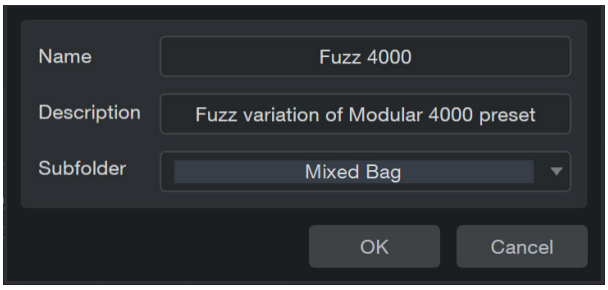
4.3.1.6 Saving Part Presets

You can save your own part presets after experimenting with the parameters. To do so, follow the directions below:

1. Press the colored menu located at the top right of the part edit page.



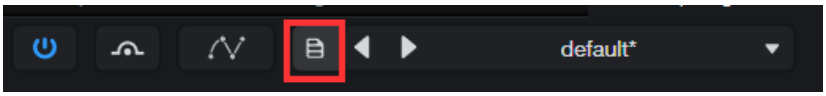
2. From the drop-down menu, select the "Store Preset" option.
3. Specify the preset information you'd like and assign it to an existing or new folder for simple organization. Once you are ready, press OK and the preset will be stored for future use.



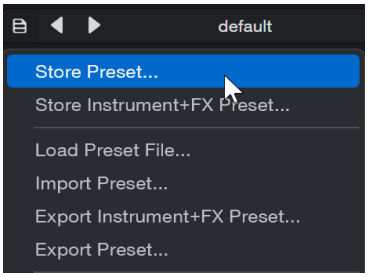
4.3.1.7 Saving Instrument Presets

Alternatively, you may wish to save a new instrument preset. When you've found a new layering of parts you'd like to store for later, you can save it as a new instrument preset by following a similar process:

1. Press onto the Preset Actions button at the top of the Lead Achitect window.



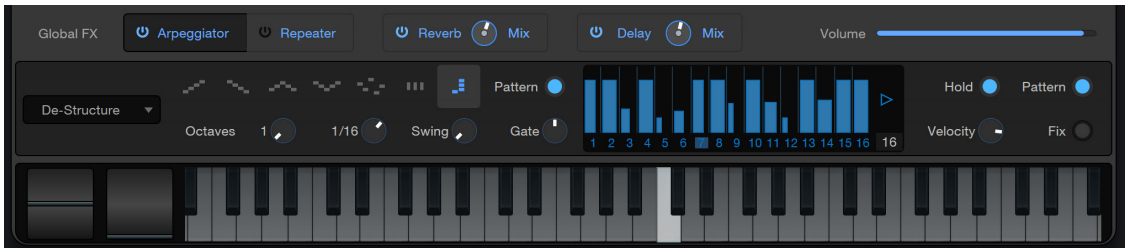
2. Select the "Store Preset" option from the drop-down menu.



3. Specify the preset information you'd like and assign it to an existing or new folder for simple organization. Once you are ready, press OK and the preset will be stored for future use.

4.3.1.8 Global FX & the Virtual Keyboard

The Global page parameters located at the bottom of the window include Global FX options, including the Arpeggiator, Repeater, two FX mix knobs, and the instrument's main volume control.



4.3.1.9 The Arpeggiator

The Arpeggiator turns chords (as well as single notes) into arpeggios—rhythmic cycles of single notes, derived from the notes currently held. You can adjust the parameters and patterns of the arpeggiator in the FX editor box that appears after enabling the Arpeggiator.

Going beyond the basics, you can use the Pattern function to create rhythmic patterns of velocity and note length that the arpeggio follows as it plays, opening many creative options for repeating musical articulations.

The following parameters and functions are available in Arpeggiator:

- **Arpeggiator Presets:** Click onto the drop down Arpeggiator Preset menu to choose from several arpeggiator templates. These presets provide you a great starting point and showcase the range of sonic possibilities that the Arpeggiator affords.
- **Arpeggio Direction:** Choose the note direction that creates the pattern you want from the following options:
  - » **Up:** The arpeggio starts at the lowest held note and travels upward through the held notes, then returns to the lowest note as the arpeggio begins again.
  - » **Down:** The arpeggio starts at the highest held note and travels downward through the held notes, then returns to the highest note as the arpeggio begins again.
  - » **Up/Down:** The arpeggio starts at the lowest held note, travels upward to the highest, then travels back down to the lowest note, and the arpeggio begins again.
  - » **Down/Up:** The arpeggio starts at the highest held note, travels downward to the lowest, then travels back up to the highest note, and the arpeggio begins again.
  - » **Random:** The arpeggio plays the currently held notes in a random pattern.
  - » **From Input:** The arpeggio pattern is derived from the order in which notes are played and held.
  - » **Chord Mode:** The currently held chord (or single note) is repeated as long as it is held.
- **Pattern:** This is a 32-step pattern sequencer you can use to create repeating patterns of note velocity and gate (length) that are applied to the control output of the Arpeggiator. The Pattern area contains the following controls:
  - » **Activate Pattern:** Toggle this on or off to enable or disable the Pattern sequencer.
  - » **Pattern Sequencer:** This series of sliders is where you'll create your velocity/gate pattern. You can click and drag each step in the sequence vertically (to set velocity) and horizontally (to set gate length). 16 steps are shown at any one time. To reach the second set of 16 steps, click the right-arrow to the right of the pattern. To return to steps 1-16, click the left-arrow to the right of the pattern.
  - » **Pattern Length:** Lets you choose the length of the pattern.

- **Octave Range:** Lets you extend the range of the arpeggio by mirroring the currently held notes an octave above (at a setting of 2), the two octaves above (at 3), or the three octaves above (at 4). At 1, no mirroring occurs and the arpeggio plays according to the held notes. Setting Octave Range above 1 when in Chord Mode causes an upward-rising cycle of repeated chords, an octave apart, its length determined by the Octave Range setting.
- **Rate:** Lets you select a rhythmic value, determining the length of each arpeggio step. Range is from a bar to a 64th-note, in all triplet and dotted varieties.
- **Swing:** Lets you add swing to the timing of the arpeggio, creating a range of rhythmic feels.
- **Gate:** Acts as an overall note-length control. Lower settings create shorter notes, and higher settings create longer notes. When a Pattern is active, Gate works relative to the gate settings in the Pattern.
- **Hold:** Enable this to hold all currently played notes. Each new chord or single note that you play replaces the previous memory and holds until the next note input is received.
- **Velocity:** Acts as an overall note-velocity control. When a Pattern is active, Velocity works relative to the velo city settings in the Pattern.
- **Pattern/Fix:** Switches Choose Pattern to allow Velocity data from the Pattern to control note velocity (in tandem with the Velocity control). Choose Fix to set a fixed velocity for all notes, with the Velocity control.

4.3.1.10 The Repeater

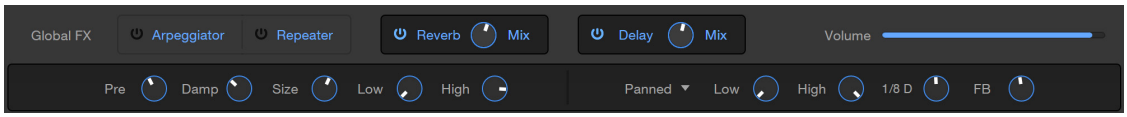
The Repeater creates rhythmic repetitions of the notes you play. These repetitions can be simple copies of incoming notes, or they can change in velocity, gate length, and pitch as the pattern plays. You can adjust the parameters of the repeater in the FX editor box that appears after enabling the Repeater.

The following parameters and features are available:

- **Repeater Presets:** Click onto the drop down Repeater Preset menu to choose from several repeater templates. These presets provide you a great starting point and showcase the range of sonic possibilities that the repeater affords.
- **Rate:** Lets you set the rate of repetition. When Sync is enabled, you can choose a tempo-synced rhythmic value between one bar and one 64th-note, in all triplet and dotted varieties. When Sync is disabled, you can choose a repetition frequency between 2 and 25 Hz (repetitions per second).
- **Steps:** Select your desired number of repetitions (and sequencer steps), from just two steps to 32.
- **Sync:** Enable Sync to snap the Rate control to rhythmic values in sync with Song tempo. Disable to set repetition speed in Hz.
- **Pitch Scale:** Turn this to introduce a sloping change in pitch in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, you get a range of downward pitch shift slopes. Turn to the right for upward slopes.
- **Velocity Level:** Acts as an overall velocity level control, scaling every step's velocity, relative to its custom setting.
- **(Velocity Relative to) Input:** Enable this to derive overall repetition velocity from note input, attenuated or boosted by the setting of the Velocity Level control.
- **Velocity Scale:** Turn this to introduce a sloping change in velocity in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, a downward slope is introduced. Turn to the right for an upward slope.
- **Gate:** Acts as an overall gate length control, scaling every step's gate length, relative to its custom setting.
- **Gate Scale:** Turn this to introduce a sloping change in gate length in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, each step in the sequence is shortened more than the last. Turn to the right, and each step is lengthened more than the last.
- **Sequencer:** This lets you specify velocity, gate, and pitch values for each step in the sequence of repetitions. The following controls are available for each step:
  - » **Level/Gate Slider:** With Individual Velocity and Gate enabled, you can click and drag the upper edge of this slider up or down to set note velocity for the current step. You can also click and drag the right edge of the slider left or right to set note gate length for the current step. With Individual Velocity and Gate disabled, moving velocity or gate sliders manipulates that setting for all steps (equivalent to turning the Velocity Level and Gate knobs).
  - » **Pitch:** With Individual Pitch enabled, you can click and drag this slider up or down to apply a positive or negative pitch transposition to the current step. With Individual Pitch disabled, moving a pitch slider changes pitch transposition for all steps simultaneously. At 0, no transposition occurs.
- **Individual Velocity:** Engage to enable individual setting of note velocity and gate length for each step.
- **Individual Pitch:** Engage to enable individual setting of note pitch for each step.

4.3.1.11 Reverb and Delay

Beside the Repeater and Arpeggiator options on the Global FX bar, Reverb and Delay Knobs provide you control over the global reverb and delay return mix. Click onto either reverb or delay boxes and their respective parameters will appear below where you can finetune the effect to your liking.



Reverb

- **Pre-delay:** Adjust the pre-delay time. Variable from 0 ms to 500 ms. Pre-delay is the amount of time before the first reverberated signals are heard.
- **Damp:** Adjusts the relative amount of damping (attenuation of the upper frequencies) of the reverberated signal. Variable from 0 to 100%.
- **Reverb Size:** This control lets you set the length of reverberation from the moment a sound starts, in a range between 100 ms and 10 seconds.
- **Low and High:** These controls let you set the cutoff frequencies of the provided high-pass and low-pass filters, which affect only the reverb signal.

Delay

- **Ping-Pong Mode:** Use this drop down menu to choose from several ping-pong delay modes.
- **Low and High:** These controls let you set the cutoff frequencies of the provided high-pass and low-pass filters, which affect only the delayed signal.
- **Delay Time:** This control lets you specify the length of the delay effect, in rhythmic values (such as 1/8th-note or 16th-note) relative to the song's tempo.
- **Feedback:** This control lets you set the amount of affected signal that is fed back into the Delay effect. At zero, there is just one repeat. As you increase the value, the trail of repeats grows.

4.3.1.12 Volume

The Volume Bar Adjust the output volume of the virtual instrument.

4.3.1.13 Virtual Keyboard

The virtual keyboard lets you easily click to play notes or manipulate the Pitch and Mod wheels, while auditioning or editing patches when you're away from a MIDI keyboard. The keyboard display also shows you which notes are currently being played.

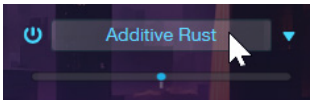
The Bend wheel parameters can be edited by clicking onto the arrow beside the pitch wheel (see screenshot below). The Bend parameters let you set the pitch bend range of the Pitch wheel in semitones. The upper value sets up-bend range, while the lower value sets down-bend range. The Pitch range parameter sets the range for the per note pitch.

Please note that you can't record notes using this virtual keyboard. For Fender Studio Pro users who want to record notes without an external keyboard, use the QWERTY Keyboard device instead.

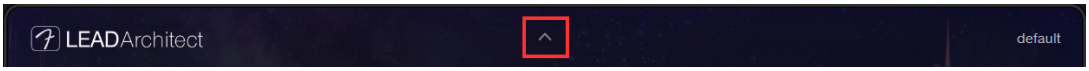
4.3.2 Part Edit Page

Shape each part to your liking by experimenting within the Part Edit page. The Part Edit page gives you access to additional LFO, Envelope, Modulation, EQ, and other sound parameters. There are two ways to access the Part Edit page:

1. Click onto the title box of the part you would like to edit.



2. Click onto the arrow at the top center of the Global page. The Part Edit page of the most recently edited part will appear.



Note: *To quickly identify which part is selected, the Part Edit page will be illuminated in the same color as the selected part sound. Simply click onto the desired part title box and the Part Edit page will change color to match that of the part you chose.*

4.3.2.1 LFO

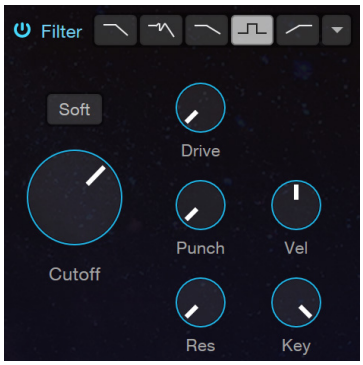
LFO stands for Low Frequency Oscillator, and each part has two of them. LFOs create slow-moving regular cycles of control signals that are useful for modulating other parameters over time. One common example is the way many keyboard patches respond when you move the Mod Wheel up from zero; the pitch of the oscillators wavers up and down in an expressive manner, much like the sound of vocal vibrato. This is simply an LFO modulating oscillator pitch to a degree set by the position of the mod wheel.

LFO 1 and 2 have identical controls, so the following explanations apply to both:

- **Bypass:** Click the [LFO 1] or [LFO 2] button to turn the selected LFO on or off.
- **LFO Type:** Choose between Sine, Triangle, Sawtooth, Square, and Random shapes for the oscillation of the LFO.
- **Rate:** Sets the rate at which the LFO oscillates, from inaudibly low (0.01 Hz) for long, sweeping changes, all the way to higher ranges (up to 8 kHz) useful for FM and AM techniques. When the LFO's [Sync] button is engaged, Rate can be set in terms of rhythmic values relative to Song tempo (such as 1/8th-note or 1/4-note).
- **Sync:** Engage this option to enable setting LFO Rate to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate by Hz.
- **Key:** Engage this option to bind LFO speed to incoming note pitch. Higher notes result in higher LFO speeds, while lower notes result in lower LFO speeds.
- **Free:** Engage this option to let the LFO run continuously, resulting in a differing LFO start point for each note played. Disengage to restart the LFO waveform at the start of each note.
- **Delay:** This control lets you specify an amount of time (in milliseconds) for the LFO to wait before becoming active after a note is played. This lets you do things like adding a bit of expression to held notes, or creating layers of modulation that start at different points in each note by setting distinct Delay values for each LFO.

4.3.2.2 Filter

Lead Achitect offers a versatile multi-mode Filter, which lets you shape and enhance your sounds. The filter is often one of the most important defining elements to the sound of an analog synthesizer.



The Filter offers the following controls:

- **Bypass:** Click the [Filter] button to turn the filter on or off.
- **Filter Mode:** Choose from the following filter modes, each with its own sound-shaping characteristics.
  - » **LP 24 dB Ladder:** This mode emulates a classic 24-dB-per-octave low-pass filter based on a transistor-ladder configuration, as found in many classic synthesizers. This type of filter allows frequencies below the chosen Cutoff frequency to pass through, cutting frequencies above Cutoff at a rate of 24 decibels per octave—a fairly aggressive slope.
  - » **LP 24 dB Zero:** This is a 24-dB-per-octave low-pass filter, based on a zero-delay-feedback architecture that closely models the tone and modulation behavior of analog filters.
  - » **LP 12 dB Ladder:** This is a low-pass filter with a 12-dB-per-octave curve, which cuts frequencies less aggressively than the 24 dB filters.
  - » **BP 12 dB Ladder:** This is a high-pass and low-pass filter in series, known collectively as a band-pass filter. It allows a selected band of frequencies to pass through, then cuts frequencies above and below that band at a rate of 12 decibels per octave.
  - » **HP 12 dB Ladder:** This is a high-pass filter with a 12-dB-per-octave slope. This lets frequencies above the chosen Cutoff frequency pass through, while cutting frequencies below Cutoff at a rate of 12 decibels per octave.
  - » **LP 12 State, BP 12 State, HP 12 State, Eco Filter:** These are a set of simple, clean digital filter models, in low-pass, band-pass, high-pass, and eco (low-CPU low-pass) modes. You can access these filter types in the drop-down menu at the end of the row of Filter Mode switches.
- **Soft:** This control lets you switch between two differing analog-modeled processing circuits within the filter. Engage Soft for a mellower, darker tone. Disengage it for a brighter, more aggressive sound.
- **Cutoff:** This lets you set the corner frequency of the filter—the point in the slope of the filter at which the filter cuts incoming audio by 3 dB. In the case of the Band-Pass filter, this sets the center frequency of the passed frequency band.
- **Drive:** This lets you specify an amount of filter overdrive, to add fullness and saturation artifacts to your sound.
- **Punch:** This control lets you add a range of percussive attack to the start of each note. At the lowest setting, dynamics are unchanged. At higher settings, the sound becomes more aggressive and more readily pops through the mix.
- **Velocity (Vel):** This control sets the relationship between incoming note Velocity and filter Cutoff. When set at the center, velocity does not affect cutoff. When moved to the right, the cutoff rises as note velocity increases. When moved to the left, cutoff lowers as note velocity increases.
- **Resonance (Res):** This lets you set the amount of resonance in the filter, which is an emphasis centered on the chosen cutoff frequency.
  - » At lower settings, the filter cuts frequencies smoothly. As you increase Res, the emphasis at the cutoff frequency becomes more pronounced, able to mimic resonances such as those in voices or acoustic instruments, as well as many classic synthesis effects.
  - » At the highest settings, the filter can self-oscillate, emitting a pitched tone at the current cutoff frequency. This filter oscillation can be treated somewhat like an extra oscillator, especially in conjunction with the Key parameter.
- **Key:** This control sets the relationship between incoming note pitch and filter cutoff. In physical instruments, higher notes tend to produce higher harmonics, brightening slightly as you go up the scale.
  - » On a synthesized instrument, if the filter stays static, setting the proper tone in the lower note ranges may cause inappropriate dullness in the higher notes. Therefore, with the Key parameter, we can compensate for this and create a more natural-sounding range of timbres up and down the keyboard.
  - » When Key is set all the way to the left, the filter is unaffected by note pitch. In the middle, cutoff follows note pitch subtly, allowing high notes to shine. When set all the way right, filter cutoff follows note pitch closely in a relative fashion, moving upward and downward in semitone values as notes are received. This lets you use the filter as an additional pitched oscillator or resonator when filter Res is set high.

4.3.2.3 Envelopes

Envelope generators are a vital part of sound synthesis, giving us the ability to shape the amplitude and timbre of our sounds within the time-scale of each note. Lead Achitect has two envelope generators per part, labeled "Amp Envelope" (so named because it is hard-wired to amplitude) and "Filter Envelope".

Both envelope modules are triggered when a note is played. Each envelope then outputs a control signal that follows the shape set by the following controls:

- **Delay (Δ - Filter Envelope only):** This control lets you specify a length of time (in ms) for the Env to pause before starting its attack phase after a note is played. This can assist in creating evolving sounds, where cycles of modulation occur at differing times over the length of a note.
- **Attack (A):** This control lets you set the time required for the envelope to go from zero (silence) to full amplitude, in a range from 0 ms to 20 seconds.
- **Decay (D):** This control lets you set the time required to drop from full amplitude to the sustain level, in a range from 0 ms to 20 seconds.
- **Sustain (S):** This control lets you set the signal level that is maintained from the end of the decay period, until the key is released, in a range from -∞ dB (silence) to 0.0 dB (full amplitude).
- **Release (R):** This control lets you set the time required to fall back to silence after the key is released, in a range from 0 ms to 30 seconds.

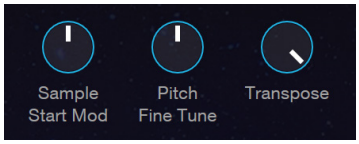
4.3.2.4 Envelope Graphical Display

Each envelope has a corresponding graphical display that represents the shape created by the settings of its parameters. There are handles on the corners and slopes of each envelope that you can click and drag, letting you shape the ADSR envelope and the curve between its points visually.

If you wish to lengthen any phase of the envelope beyond the time limits of the current display, simply drag the point toward the right of the graph, and the time scale adjusts to properly display the new setting.

4.3.2.5 Pitch Controls

The pitch control knobs let you manipulate the way that Lead Achitect plays the parts in the currently loaded preset.



- **Sample Start Mod:** Lets you specify an amount of negative or positive velocity-controlled offset, applied to the point in the part which playback begins. At settings above and below the default of 0, lower-velocity notes trigger a smaller amount of sample offset, and higher-velocity notes trigger a larger amount of offset.
- **Pitch Fine Tune:** Adjusts the tuning, in cents, for the selected pad. Variable from -100 to 100 cents.
- **Transpose:** Adjusts the transposition in semitones for the selected pad. Variable from -48 to +48.

4.3.2.6 Part Global Settings

The following Global parameters let you configure Lead Achitect's overall behavior and capabilities specific to the part you're editing:

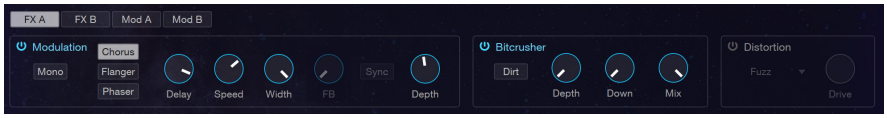
- **Volume:** This control lets you set the total output volume, in a range from -∞ dB (silence) to +10.0 dB (ten decibels above unity gain).
- **Velocity:** This control lets you set the degree to which Lead Achitect's volume is affected by note velocity, from zero (no velocity sensitivity) to 1.0 (full velocity sensitivity).
- **Poly, Mono, and Glide:** Enable Poly mode to allow polyphonic playing (more than one note at a time). Enable Mono mode to play just one note at a time. When in Mono mode, you can enable Glide to cause the pitch to sweep smoothly from that of the currently held note to that of the next note, when played legato (one note played while the previous note is held). The Glide knob lets you set the rate of pitch change over time, from 1 ms to 1 second.

4.3.2.7 FX and Mods

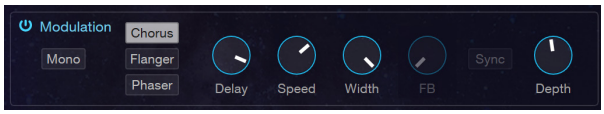
Lead Achitect offers built-in effects processors to add dimension to your sounds. They are arranged in four banks: FX A, FX B, Mod A, and Mod B. You can enable or disable each effect by clicking its name.

FX A

The first bank, FX A, includes Modulation (Chorus, Flanger, and Phaser), Bitcrusher, and Distortion effects.



Modulation



This processor creates time-based modulation effects. Choose from the following modes by clicking the [Chorus], [Flanger], or [Phaser] button.

Chorus: This processor creates effects similar to that of multiple identical instruments playing the same part simultaneously. The synth signal is fed through a short, modulated delay, which is then mixed with the dry signal. Chorus offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Delay:** This control lets you set the length of the modulated delay. Higher settings create full-bodied chorusing effects, while lower settings create more pronounced harmonics, akin to the effects of a Flanger.
- **Speed:** This control lets you set the speed at which the delay line is modulated. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the delay line is modulated. Lower settings produce subtler chorusing effects, while higher settings produce more pronounced changes in timbre over time.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the chorused signal (all the way right).

Flanger: This processor creates resonant, hollow-sounding sweeping effects. The synth signal is fed through a short, modulated delay, which is mixed with the dry signal. While similar to the workings of a Chorus effect, Flangers get their signature sound by employing smaller delay times than those used in chorusing, combined with a feedback system that can add extra resonance to the sweep. Flanger offers the following controls:

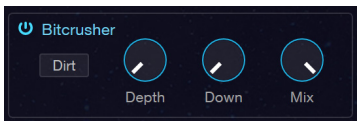
- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Delay:** This control lets you set the length of the modulated delay (in ms), which changes the pitch of the resultant resonance. Higher settings create lower-pitched resonance, while lower settings create resonances at a higher pitch.
- **Speed:** This control lets you set the speed at which the delay line is modulated. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the delay line is modulated. Lower settings produce subtler flanging effects, while higher settings produce more pronounced changes in timbre over time.
- **Feedback (FB):** This control lets you set the amount of output signal to feed back into the Flanger. Higher amounts of Feedback add to the resonance of the sweeping effect.
- **Sync:** Engage this option to enable setting Flanger modulation speed to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate on a continuous scale.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the flanged signal (all the way right).

Phaser: This processor creates dreamy, otherworldly sweeping effects. The synth signal is fed through a series of all pass filters that alter its phase. When mixed with the dry signal, this creates a series of peaks and valleys in the frequency response that changes depending on the degree of phase shift applied. Phaser offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Shift:** This control lets you specify the amount of phase shift to apply. Lower settings focus the phasing effect in the lower frequencies, while higher settings focus the effect in higher frequencies.
- **Speed:** This control lets you set the speed of modulation applied to the phase shift amount. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the phase shift amount is modulated. Lower settings produce subtler effects, while higher settings produce more pronounced changes in timbre over time.
- **Feedback (FB):** This control lets you set the amount of output signal to feed back into the Phaser. Higher amounts of Feedback add to the resonance of the sweeping effect.
- **Sync:** Engage this option to enable setting Phaser modulation speed to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate on a continuous scale.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the phase-shifted signal (all the way right).

Bitcrusher

Perfect for audio abuse, Bitcrusher combines overdrive, bit-depth reduction, downsampling and clipping into a single plug-in. Bit depth reduction and downsampling are both digital resolution-reduction techniques, but each has its own sonic effect. When used in combination, they create a wide variety of tonal options.

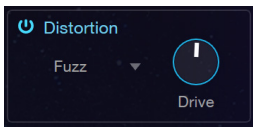


The following parameters are available in Bitcrusher:

- **Dirt:** Enable this to introduce a high-frequency instability in the Bit Depth reduction effect. Good for creating aggressive sounds.
- **Depth:** Lets you specify the level of Bit Depth reduction to apply, from 24-bit to 1-bit.
- **Down:** Lets you specify the level of downsampling to apply.
- **Mix:** Lets you blend between the dry (0%) and effected (100%) signals.

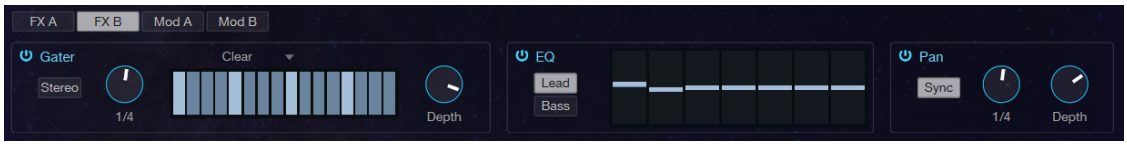
Distortion

This processor adds grit and character to your sounds. The distortion FX parameters within Lead Achitect are quite simple: Use the dropdown menu to choose from a variety of distortion types, from fizzy transistor fuzzes to thick, warm tube overdrives. Then, set the amount of distortion with the Drive knob.



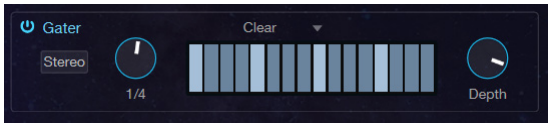
FX B

The second bank, FX B, includes Gater, Lead/Bass EQ, and Panning FX options.



Gater

The Gater will add a rhythmic gating effect, able to create a series of syncopated breaks in the synth signal. A variety of presets are provided, each with a different rhythmic gating pattern. However, the fun really begins when you create your own.



The Gater offers the following controls:

- **Stereo:** Engaging this option creates a separate beat grid for each side of the stereo field. When engaged, you'll see two rows of beat steps, the top row specifying gate steps for the left channel, and the bottom row gating the right channel.
- **Beats:** This control lets you set the length of the gating cycle, in rhythmic values (such as 1 bar or 1/2-note) relative to Song tempo. For example, at a setting of 1 bar, the 16 steps in the cycle repeat every bar, effectively representing 16th-notes. At a 1/2-note setting, the 16 steps repeat each half-bar, representing 32nd-note values.
- **Gate Presets:** There are nine provided gate presets to choose from.
- **Gate Steps:** This grid lets you specify which steps in the cycle let signals pass through, and which gate the signal to silence. Click on a step to enable or disable gating for that step.
- **Depth:** This control lets you blend between the gated and dry signals, allowing for rhythmic gating effects while retaining the continuity of the synth sound.

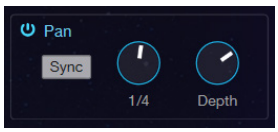
EQ

FX B includes a graphic equalizer effect, perfect for quick tonal shaping. Set the EQ bands to emphasize or attenuate bands of frequencies to suit your needs. When a band is in the center of its range, it neither adds or subtracts. When moved above the center, it emphasizes the chosen frequency. Moved below the center, it attenuates that frequency.

Choose between Lead mode, with frequencies chosen to suit aggressive, up-front sounds, or Bass mode, with wider ranging frequencies that work better for basses and mellower chordal parts.

Pan

This is an auto-pan effect, which pans the synth signal left and right over time.



Pan offers the following controls:

- **Sync:** Enable this option to set pan speed to a rhythmic value (such as 1/4-note or 16th-note) relative to Song tempo. Disable this option to set pan speed along a continuous range.
- **Speed:** This control lets you set the speed at which the signal is panned left and right.
- **Depth:** This control lets you set the degree to which the signal is panned. Lower settings give a subtly panned effect, while higher settings pan the signal more radically, all the way to fully left and right in each cycle.

4.3.2.8 Modulation Matrix

If you would like to map additional modulation parameters, Lead Achitect provides 16 configurable modulation routings, split between two banks of eight (Mod A and Mod B). Modulation signals can be routed from a selection of incoming MIDI controller signals, including MPE data (such as Pitch Bend, Mod Wheel, Aftertouch, Poly Pressure, Note Timbre, and Pitch Control), modulation generators (such as the LFOs and envelopes), or the pitch or velocity of played notes.

These modulation signals can be used to vary most of the parameters throughout Lead Achitect, including modulation sources themselves (such as LFO 2 modulating the rate of LFO 1, or the Decay of Env 2). The following list describes the functions of each option within the Modulation Matrix, from top to bottom:

- **Bypass:** Each modulation slot has a bypass button at the top, which lets you enable or disable the flow of the modulation signal.
- **Main Source Selector:** Use this selector to specify the modulation signal you'd like to route to the Modifier Selector or Target.
- **Modifier Selector ("Via"):** Similar to the Main Source Selector, you can use this selector to specify the modulation signal you'd like to route to the Modifier Selector or Target. Additionally, you can use this selector in conjunction with the Main Source to further experiment with routing.
  - » For example, if you assign a modulation source to the input selector only, that signal is routed directly to the chosen destination. In some cases, you'll want to govern the flow of one mod source before it reaches its destination, using the signal from another mod source. On the other hand, you may want to control the output level of LFO 1 (routed to a parameter such as oscillator pitch) with the Mod Wheel. In this case, you'd choose Mod Wheel with the input selector, and LFO 1 with the modifier selector below.
- **Slider:** The slider underneath the selectors controls the amplitude and polarity of the modulation signal. Set at its center, no modulation occurs. Move the handle right of center to send an increasing amount of the modulation signal, at its normal (positive) polarity, to the chosen destination. Move it left of center to send the signal to its destination with a negative value.
  - » If the parameter you wish to modulate is set to a high value, you may want to send a negative modulation signal to it, driving the setting downward and causing more audible effects. Positive-going modulation signals are more efficient when modulating parameters set to low values.
- **Target Selector:** A selector at the bottom of each modulation slot lets you choose the destination of the chosen modulation signals. Signals can be mapped to any of the available destinations within Lead Achitect's Part Edit page.

4.4 Sub Zero Bass

Sub Zero Bass is a sample-based hybrid virtual instrument that enables you to layer three different sound components (based on analog modular and digital synthesis, as well as field recordings) to create unique sounds, perfect for melodies, solos, ensembles and any other cinematic sound you can imagine. Each part provides control over volume, panning, and effects, including reverb, delay, distortion and note-based arpeggiator and repeater FX. On the Part level, you can shape your sounds with the provided filter, LFOs, envelopes, Modulation Matrix, and effects.

Sub Zero Bass has two pages: the Global and Part Edit page. The Global page has a minimal, sleek design, emphasizing the main elements of Sub Zero Bass to simplify your experience playing the instrument. The Part Edit page offers many more sound design options, giving you more control of each part.

Before starting to experiment with Sub Zero Bass, make sure you've downloaded the Sound Set that came bundled with Sub Zero Bass. Sub Zero Bass requires the Sound Set download for sounds and presets to be present within the instrument.

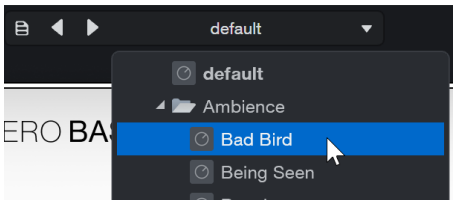
4.4.1 Global Page

The Global Page grants users easy access to the virtual instrument's global controls. From the Global Control Panel, you can quickly modify the volume, cutoff, sample shift, Attack and Release of all three layered Parts. Additionally, you have access to the Part Preset selection, the Global FX panel and virtual keyboard.

4.4.1.1 Instrument Presets

To begin experimenting with Sub Zero Bass, you might consider loading one of the Instrument Presets. Once the preset is loaded, check out the sections below about adjusting the part parameters.

You can locate and load Instrument Presets by clicking the arrow of the instrument preset drop-down menu located at the top left corner of Sub Zero Bass. Find your preset of choice and click on the preset name to load it:



Once loaded, the instrument preset is immediately playable with your MIDI controller or by clicking the virtual keyboard at the bottom of the plug-in window.

Note: *You cannot record notes using the virtual instrument keyboard. For Fender Studio Pro users who want to record notes without an external keyboard, use the QWERTY Keyboard device instead.*

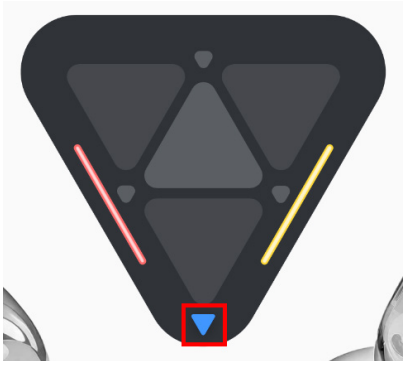
4.4.1.2 Main Control Panel

The Main Control Panel on the Global page includes the Cutoff, Drive, Bit Depth, Triangle Vector Pad, and the Attack and Release parameters.

- **Cutoff:** This lets you set the corner frequency of the filter (the point in the slope of the filter at which the filter cuts incoming audio by 12 or 24 dB). In the case of a Band-Pass filter, this sets the center frequency of the passed frequency band.
- **Drive:** Lets you specify the percentage of Drive (distortion) to apply to all three layered Parts at once.
- **Bit Depth:** Lets you specify the level of Bit Depth reduction to apply to all three layered Parts at once.
- **Triangle Vector Pad:** Provides combined volume control for all three parts. Move the small white triangle within the Vector Pad to seamlessly control the volume blend between the three parts. Each part has its own color, seen here on the color volume bars surrounding the pad.

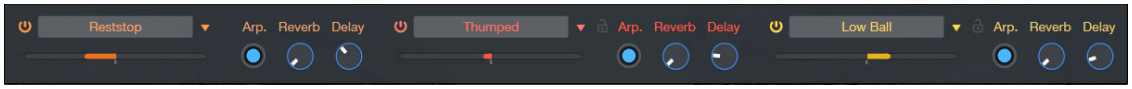
  If you would like to assign all the volume to one part and mute the others, click onto the small gray tri angles next to the volume bar associated with the part you'd like to solo:

  When you hover your mouse over the corners of the Triangle Vector Pad, gray triangles appear. If you click onto these triangles, the two parts adjacent to the point will share the volume balance while the part on the opposite side of the triangle will be muted:

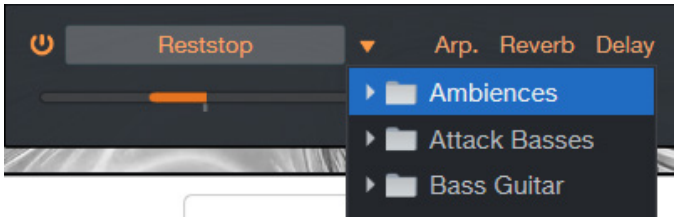
- **Attack Parameters:** Adjusts the amount of time it takes a note's level to reach its maximum once a key has been triggered. Variable from 0 to 20s.
- **Release Parameters:** Adjusts the amount of time it takes a note's level to go back to zero after a key has been released. Variable from 0 to 30s.

4.4.1.3 Part Selection



Under the main control panel, there are three drop-down menu boxes where you can choose your Part Preset sounds, enable/disable load sample only lock (see below), enable/disable the arpeggiator or repeater for each specific part, modify each individual part with Reverb and Delay sends to the global FX and set the panning for each part (set the bar underneath each part left or right to determine where it's panned).

4.4.1.4 Part Presets



From the part preset drop-down menu boxes, you can choose from a list of preset sounds. With each instrument preset, there are three predetermined part presets, but you can swap out part presets and experiment with preset layering as desired.

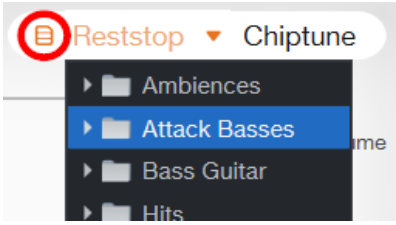
4.4.1.5 Load Samples Only Lock

The "Load Samples Only" lock is located to the right of each part preset drop down box. When the lock is engaged, any changes you make to the parameters of a part will remain if you decide to change the part's preset sound.

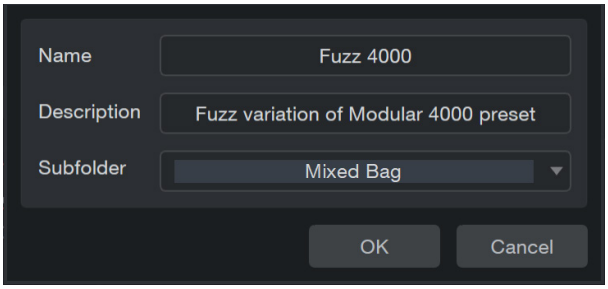
4.4.1.6 Saving Part Presets

You can save your own part presets after experimenting with the parameters. To do so, follow the directions below:

1. Press the colored menu located at the top right of the part edit page.



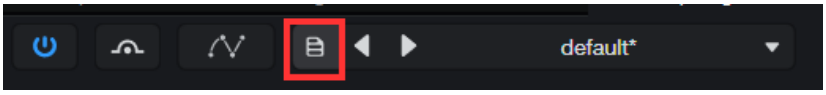
2. From the drop-down menu, select the "Store Preset" option.
3. Specify the preset information you'd like and assign it to an existing or new folder for simple organization. Once you are ready, press OK and the preset will be stored for future use.



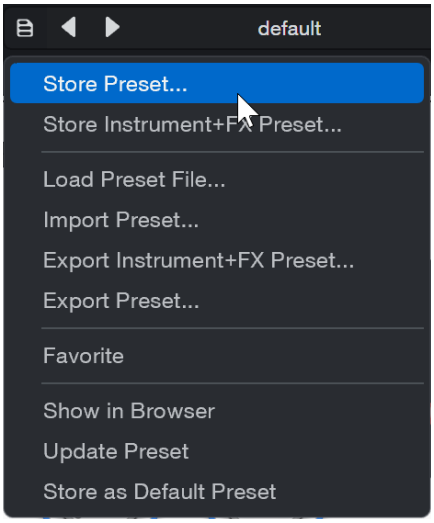
4.1.1.7 Saving Instrument Presets

Alternatively, you may wish to save a new instrument preset. When you've found a new layering of parts you'd like to store for later, you can save it as a new instrument preset by following a similar process:

1. Press onto the Preset Actions button at the top of the Sub Zero Bass window.



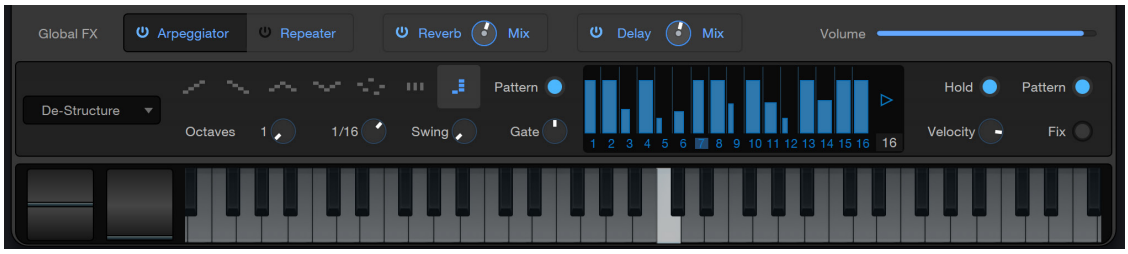
2. Select the "Store Preset" option from the drop-down menu.



3. Specify the preset information you'd like and assign it to an existing or new folder for simple organization. Once you are ready, press OK and the preset will be stored for future use.

4.4.1.8 Global FX & the Virtual Keyboard

The Global page parameters located at the bottom of the window include Global FX options, including the Arpeggiator, Repeater, two FX mix knobs, and the instrument's main volume control.



4.4.1.9 The Arpeggiator

The Arpeggiator turns chords (as well as single notes) into arpeggios—rhythmic cycles of single notes, derived from the notes currently held. You can adjust the parameters and patterns of the arpeggiator in the FX editor box that appears after enabling the Arpeggiator.

Going beyond the basics, you can use the Pattern function to create rhythmic patterns of velocity and note length that the arpeggio follows as it plays, opening many creative options for repeating musical articulations.

The following parameters and functions are available in Arpeggiator:

- **Arpeggiator Presets:** Click onto the drop down Arpeggiator Preset menu to choose from several arpeggiator templates. These presets provide you a great starting point and showcase the range of sonic possibilities that the Arpeggiator affords.
- **Arpeggio Direction:** Choose the note direction that creates the pattern you want from the following options:
  - » **Up:** The arpeggio starts at the lowest held note and travels upward through the held notes, then returns to the lowest note as the arpeggio begins again.
  - » **Down:** The arpeggio starts at the highest held note and travels downward through the held notes, then returns to the highest note as the arpeggio begins again.
  - » **Up/Down:** The arpeggio starts at the lowest held note, travels upward to the highest, then travels back down to the lowest note, and the arpeggio begins again.
  - » **Down/Up:** The arpeggio starts at the highest held note, travels downward to the lowest, then travels back up to the highest note, and the arpeggio begins again.
  - » **Random:** The arpeggio plays the currently held notes in a random pattern.
  - » **From Input:** The arpeggio pattern is derived from the order in which notes are played and held.
  - » **Chord Mode:** The currently held chord (or single note) is repeated as long as it is held.

- **Pattern:** This is a 32-step pattern sequencer you can use to create repeating patterns of note velocity and gate (length) that are applied to the control output of the Arpeggiator. The Pattern area contains the following controls:
  - » **Activate Pattern:** Toggle this on or off to enable or disable the Pattern sequencer.
  - » **Pattern Sequencer:** This series of sliders is where you'll create your velocity/gate pattern. You can click and drag each step in the sequence vertically (to set velocity) and horizontally (to set gate length). 16 steps are shown at any one time. To reach the second set of 16 steps, click the right-arrow to the right of the pattern. To return to steps 1-16, click the left-arrow to the right of the pattern.
  - » **Pattern Length:** Lets you choose the length of the pattern.
- **Octave Range:** Lets you extend the range of the arpeggio by mirroring the currently held notes an octave above (at a setting of 2), the two octaves above (at 3), or the three octaves above (at 4). At 1, no mirroring occurs and the arpeggio plays according to the held notes. Setting Octave Range above 1 when in Chord Mode causes an upward-rising cycle of repeated chords, an octave apart, its length determined by the Octave Range setting.
- **Rate:** Lets you select a rhythmic value, determining the length of each arpeggio step. Range is from a bar to a 64th-note, in all triplet and dotted varieties.
- **Swing:** Lets you add swing to the timing of the arpeggio, creating a range of rhythmic feels.
- **Gate:** Acts as an overall note-length control. Lower settings create shorter notes, and higher settings create longer notes. When a Pattern is active, Gate works relative to the gate settings in the Pattern.
- **Hold:** Enable this to hold all currently played notes. Each new chord or single note that you play replaces the previous memory and holds until the next note input is received.
- **Velocity:** Acts as an overall note-velocity control. When a Pattern is active, Velocity works relative to the velo city settings in the Pattern.
- **Pattern/Fix:** Switches Choose Pattern to allow Velocity data from the Pattern to control note velocity (in tandem with the Velocity control). Choose Fix to set a fixed velocity for all notes, with the Velocity control.

4.4.1.10 The Repeater

The Repeater creates rhythmic repetitions of the notes you play. These repetitions can be simple copies of incoming notes, or they can change in velocity, gate length, and pitch as the pattern plays. You can adjust the parameters of the repeater in the FX editor box that appears after enabling the Repeater.

The following parameters and features are available:

- **Repeater Presets:** Click onto the drop down Repeater Preset menu to choose from several repeater templates. These presets provide you a great starting point and showcase the range of sonic possibilities that the repeater affords.
- **Rate:** Lets you set the rate of repetition. When Sync is enabled, you can choose a tempo-synced rhythmic value between one bar and one 64th-note, in all triplet and dotted varieties. When Sync is disabled, you can choose a repetition frequency between 2 and 25 Hz (repetitions per second).
- **Steps:** Select your desired number of repetitions (and sequencer steps), from just two steps to 32.
- **Sync:** Enable Sync to snap the Rate control to rhythmic values in sync with Song tempo. Disable to set repetition speed in Hz.
- **Pitch Scale:** Turn this to introduce a sloping change in pitch in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, you get a range of downward pitch shift slopes. Turn to the right for upward slopes.
- **Velocity Level:** Acts as an overall velocity level control, scaling every step's velocity, relative to its custom setting.
- **(Velocity Relative to) Input:** Enable this to derive overall repetition velocity from note input, attenuated or boosted by the setting of the Velocity Level control.
- **Velocity Scale:** Turn this to introduce a sloping change in velocity in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, a downward slope is introduced. Turn to the right for an upward slope.
- **Gate:** Acts as an overall gate length control, scaling every step's gate length, relative to its custom setting.
- **Gate Scale:** Turn this to introduce a sloping change in gate length in the sequence over time, relative to each step's setting. At center, no slope is added. Turned left of center, each step in the sequence is shortened more than the last. Turn to the right, and each step is lengthened more than the last.
- **Sequencer:** This lets you specify velocity, gate, and pitch values for each step in the sequence of repetitions. The following controls are available for each step:

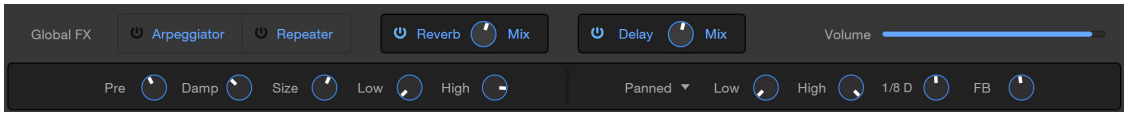
» **Level/Gate Slider:** With Individual Velocity and Gate enabled, you can click and drag the upper edge of this slider up or down to set note velocity for the current step. You can also click and drag the right edge of the slider left or right to set note gate length for the current step. With Individual Velocity and Gate disabled, moving velocity or gate sliders manipulates that setting for all steps (equivalent to turning the Velocity Level and Gate knobs).

» **Pitch:** With Individual Pitch enabled, you can click and drag this slider up or down to apply a positive or negative pitch transposition to the current step. With Individual Pitch disabled, moving a pitch slider changes pitch transposition for all steps simultaneously. At 0, no transposition occurs.

- **Individual Velocity:** Engage to enable individual setting of note velocity and gate length for each step.
- **Individual Pitch:** Engage to enable individual setting of note pitch for each step.

4.4.1.11 Reverb and Delay

Beside the Repeater and Arpeggiator options on the Global FX bar, Reverb and Delay Knobs provide you control over the global reverb and delay return mix. Click onto either reverb or delay boxes and their respective parameters will appear below where you can finetune the effect to your liking.



Reverb

- **Pre-delay:** Adjust the pre-delay time. Variable from 0 ms to 500 ms. Pre-delay is the amount of time before the first reverberated signals are heard.
- **Damp:** Adjusts the relative amount of damping (attenuation of the upper frequencies) of the reverberated signal. Variable from 0 to 100%.
- **Reverb Size:** This control lets you set the length of reverberation from the moment a sound starts, in a range between 100 ms and 10 seconds.
- **Low and High:** These controls let you set the cutoff frequencies of the provided high-pass and low-pass filters, which affect only the reverb signal.

Delay

- **Ping-Pong Mode:** Use this drop down menu to choose from several ping-pong delay modes.
- **Low and High:** These controls let you set the cutoff frequencies of the provided high-pass and low-pass filters, which affect only the delayed signal.
- **Delay Time:** This control lets you specify the length of the delay effect, in rhythmic values (such as 1/8th-note or 16th-note) relative to the song's tempo.
- **Feedback:** This control lets you set the amount of affected signal that is fed back into the Delay effect. At zero, there is just one repeat. As you increase the value, the trail of repeats grows.

4.4.1.12 Volume

The Volume Bar Adjust the output volume of the virtual instrument.

4.4.1.13 Virtual Keyboard

The virtual keyboard lets you easily click to play notes or manipulate the Pitch and Mod wheels, while auditioning or editing patches when you're away from a MIDI keyboard. The keyboard display also shows you which notes are currently being played.

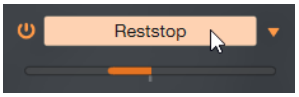
The Bend wheel parameters can be edited by clicking onto the arrow beside the pitch wheel (see screenshot below). The Bend parameters let you set the pitch bend range of the Pitch wheel in semitones. The upper value sets up-bend range, while the lower value sets down-bend range. The Pitch range parameter sets the range for the per note pitch.

Please note that you can't record notes using this virtual keyboard. For Fender Studio Pro users who want to record notes without an external keyboard, use the QWERTY Keyboard device instead.

4.4.2 Part Edit Page

Shape each part to your liking by experimenting within the Part Edit page. The Part Edit page gives you access to additional LFO, Envelope, Modulation, EQ, and other sound parameters. There are two ways to access the Part Edit page:

1. Click onto the title box of the part you would like to edit.



2. Click onto the arrow at the top center of the Global page. The Part Edit page of the most recently edited part will appear.

Note: *To quickly identify which part is selected, the Part Edit page will be illuminated in the same color as the selected part sound. Simply click onto the desired part title box and the Part Edit page will change color to match that of the part you chose.*

4.4.2.1 LFO

LFO stands for Low Frequency Oscillator, and each part has two of them. LFOs create slow-moving regular cycles of control signals that are useful for modulating other parameters over time. One common example is the way many keyboard patches respond when you move the Mod Wheel up from zero; the pitch of the oscillators wavers up and down in an expressive manner, much like the sound of vocal vibrato. This is simply an LFO modulating oscillator pitch to a degree set by the position of the mod wheel.

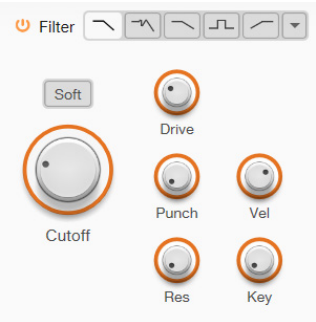
LFO 1 and 2 have identical controls, so the following explanations apply to both:

- **Bypass:** Click the [LFO 1] or [LFO 2] button to turn the selected LFO on or off.
- **LFO Type:** Choose between Sine, Triangle, Sawtooth, Square, and Random shapes for the oscillation of the LFO.
- **Rate:** Sets the rate at which the LFO oscillates, from inaudibly low (0.01 Hz) for long, sweeping changes, all the way to higher ranges (up to 8 kHz) useful for FM and AM techniques. When the LFO's [Sync] button is engaged, Rate can be set in terms of rhythmic values relative to Song tempo (such as 1/8th-note or 1/4-note).
- **Sync:** Engage this option to enable setting LFO Rate to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate by Hz.

- **Key:** Engage this option to bind LFO speed to incoming note pitch. Higher notes result in higher LFO speeds, while lower notes result in lower LFO speeds.
- **Free:** Engage this option to let the LFO run continuously, resulting in a differing LFO start point for each note played. Disengage to restart the LFO waveform at the start of each note.
- **Delay:** This control lets you specify an amount of time (in milliseconds) for the LFO to wait before becoming active after a note is played. This lets you do things like adding a bit of expression to held notes, or creating layers of modulation that start at different points in each note by setting distinct Delay values for each LFO.

4.4.2.2 Filter

Sub Zero Bass offers a versatile multi-mode Filter, which lets you shape and enhance your sounds. The filter is often one of the most important defining elements to the sound of an analog synthesizer.



The Filter offers the following controls:

- **Bypass:** Click the [Filter] button to turn the filter on or off.
- **Filter Mode:** Choose from the following filter modes, each with its own sound-shaping characteristics.
  - » **LP 24 dB Ladder:** This mode emulates a classic 24-dB-per-octave low-pass filter based on a transistor-ladder configuration, as found in many classic synthesizers. This type of filter allows frequencies below the chosen Cutoff frequency to pass through, cutting frequencies above Cutoff at a rate of 24 decibels per octave—a fairly aggressive slope.
  - » **LP 24 dB Zero:** This is a 24-dB-per-octave low-pass filter, based on a zero-delay-feedback architecture that closely models the tone and modulation behavior of analog filters.
  - » **LP 12 dB Ladder:** This is a low-pass filter with a 12-dB-per-octave curve, which cuts frequencies less aggressively than the 24 dB filters.
  - » **BP 12 dB Ladder:** This is a high-pass and low-pass filter in series, known collectively as a band-pass filter. It allows a selected band of frequencies to pass through, then cuts frequencies above and below that band at a rate of 12 decibels per octave.
  - » **HP 12 dB Ladder:** This is a high-pass filter with a 12-dB-per-octave slope. This lets frequencies above the chosen Cutoff frequency pass through, while cutting frequencies below Cutoff at a rate of 12 decibels per octave.
  - » **LP 12 State, BP 12 State, HP 12 State, Eco Filter:** These are a set of simple, clean digital filter models, in low-pass, band-pass, high-pass, and eco (low-CPU low-pass) modes. You can access these filter types in the drop-down menu at the end of the row of Filter Mode switches.
- **Soft:** This control lets you switch between two differing analog-modeled processing circuits within the filter. Engage Soft for a mellower, darker tone. Disengage it for a brighter, more aggressive sound.
- **Cutoff:** This lets you set the corner frequency of the filter—the point in the slope of the filter at which the filter cuts incoming audio by 3 dB. In the case of the Band-Pass filter, this sets the center frequency of the passed frequency band.
- **Drive:** This lets you specify an amount of filter overdrive, to add fullness and saturation artifacts to your sound.
- **Punch:** This control lets you add a range of percussive attack to the start of each note. At the lowest setting, dynamics are unchanged. At higher settings, the sound becomes more aggressive and more readily pops through the mix.
- **Velocity (Vel):** This control sets the relationship between incoming note Velocity and filter Cutoff. When set at the center, velocity does not affect cutoff. When moved to the right, the cutoff rises as note velocity increases. When moved to the left, cutoff lowers as note velocity increases.
- **Resonance (Res):** This lets you set the amount of resonance in the filter, which is an emphasis centered on the chosen cutoff frequency.
  - » At lower settings, the filter cuts frequencies smoothly. As you increase Res, the emphasis at the cutoff frequency becomes more pronounced, able to mimic resonances such as those in voices or acoustic instruments, as well as many classic synthesis effects.
  - » At the highest settings, the filter can self-oscillate, emitting a pitched tone at the current cutoff frequency. This filter oscillation can be treated somewhat like an extra oscillator, especially in conjunction with the Key parameter.
- **Key:** This control sets the relationship between incoming note pitch and filter cutoff. In physical instruments, higher notes tend to produce higher harmonics, brightening slightly as you go up the scale.

» On a synthesized instrument, if the filter stays static, setting the proper tone in the lower note ranges may cause inappropriate dullness in the higher notes. Therefore, with the Key parameter, we can compensate for this and create a more natural-sounding range of timbres up and down the keyboard.

» When Key is set all the way to the left, the filter is unaffected by note pitch. In the middle, cutoff follows note pitch subtly, allowing high notes to shine. When set all the way right, filter cutoff follows note pitch closely in a relative fashion, moving upward and downward in semitone values as notes are received. This lets you use the filter as an additional pitched oscillator or resonator when filter Res is set high.

4.4.2.3 Envelopes

Envelope generators are a vital part of sound synthesis, giving us the ability to shape the amplitude and timbre of our sounds within the time-scale of each note. Sub Zero Bass has two envelope generators per part, labeled "Amp Envelope" (so named because it is hard-wired to amplitude) and "Filter Envelope".

Both envelope modules are triggered when a note is played. Each envelope then outputs a control signal that follows the shape set by the following controls:

- **Delay (Δ - Filter Envelope only):** This control lets you specify a length of time (in ms) for the Env to pause before starting its attack phase after a note is played. This can assist in creating evolving sounds, where cycles of modulation occur at differing times over the length of a note.
- **Attack (A):** This control lets you set the time required for the envelope to go from zero (silence) to full amplitude, in a range from 0 ms to 20 seconds.
- **Decay (D):** This control lets you set the time required to drop from full amplitude to the sustain level, in a range from 0 ms to 20 seconds.
- **Sustain (S):** This control lets you set the signal level that is maintained from the end of the decay period, until the key is released, in a range from -∞ dB (silence) to 0.0 dB (full amplitude).
- **Release (R):** This control lets you set the time required to fall back to silence after the key is released, in a range from 0 ms to 30 seconds.

4.4.2.4 Envelope Graphical Display

Each envelope has a corresponding graphical display that represents the shape created by the settings of its parameters. There are handles on the corners and slopes of each envelope that you can click and drag, letting you shape the ADSR envelope and the curve between its points visually.

If you wish to lengthen any phase of the envelope beyond the time limits of the current display, simply drag the point toward the right of the graph, and the time scale adjusts to properly display the new setting.

4.4.2.5 Pitch Controls

The pitch control knobs let you manipulate the way that Sub Zero Bass plays the parts in the currently loaded preset.

- **Sample Start Mod:** Lets you specify an amount of negative or positive velocity-controlled offset, applied to the point in the part which playback begins. At settings above and below the default of 0, lower-velocity notes trigger a smaller amount of sample offset, and higher-velocity notes trigger a larger amount of offset.
- **Pitch Fine Tune:** Adjusts the tuning, in cents, for the selected pad. Variable from -100 to 100 cents.
- **Transpose:** Adjusts the transposition in semitones for the selected pad. Variable from -48 to +48.

4.4.2.6 Part Global Settings

The following Global parameters let you configure Sub Zero Bass's overall behavior and capabilities specific to the part you're editing:

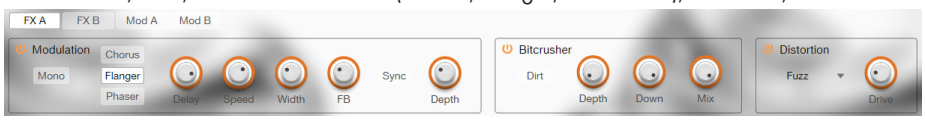
- **Volume:** This control lets you set the total output volume, in a range from -∞ dB (silence) to +10.0 dB (ten decibels above unity gain).
- **Velocity:** This control lets you set the degree to which Sub Zero Bass's volume is affected by note velocity, from zero (no velocity sensitivity) to 1.0 (full velocity sensitivity).
- **Poly, Mono, and Glide:** Enable Poly mode to allow polyphonic playing (more than one note at a time). Enable Mono mode to play just one note at a time. When in Mono mode, you can enable Glide to cause the pitch to sweep smoothly from that of the currently held note to that of the next note, when played legato (one note played while the previous note is held). The Glide knob lets you set the rate of pitch change over time, from 1 ms to 1 second.

4.4.2.7 FX and Mods

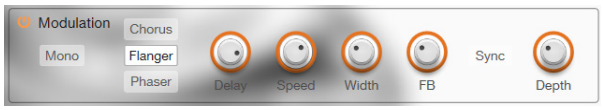
Sub Zero Bass offers built-in effects processors to add dimension to your sounds. They are arranged in four banks: FX A, FX B, Mod A, and Mod B. You can enable or disable each effect by clicking its name.

FX A

The first bank, FX A, includes Modulation (Chorus, Flanger, and Phaser), Bitcrusher, and Distortion effects.



Modulation



This processor creates time-based modulation effects. Choose from the following modes by clicking the [Chorus], [Flanger], or [Phaser] button.

Chorus: This processor creates effects similar to that of multiple identical instruments playing the same part simultaneously. The synth signal is fed through a short, modulated delay, which is then mixed with the dry signal. Chorus offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Delay:** This control lets you set the length of the modulated delay. Higher settings create full-bodied chorusing effects, while lower settings create more pronounced harmonics, akin to the effects of a Flanger.
- **Speed:** This control lets you set the speed at which the delay line is modulated. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the delay line is modulated. Lower settings produce subtler chorusing effects, while higher settings produce more pronounced changes in timbre over time.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the chorused signal (all the way right).

Flanger: This processor creates resonant, hollow-sounding sweeping effects. The synth signal is fed through a short, modulated delay, which is mixed with the dry signal. While similar to the workings of a Chorus effect, Flangers get their signature sound by employing smaller delay times than those used in chorusing, combined with a feedback system that can add extra resonance to the sweep. Flanger offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Delay:** This control lets you set the length of the modulated delay (in ms), which changes the pitch of the resultant resonance. Higher settings create lower-pitched resonance, while lower settings create resonances at a higher pitch.
- **Speed:** This control lets you set the speed at which the delay line is modulated. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the delay line is modulated. Lower settings produce subtler flanging effects, while higher settings produce more pronounced changes in timbre over time.
- **Feedback (FB):** This control lets you set the amount of output signal to feed back into the Flanger. Higher amounts of Feedback add to the resonance of the sweeping effect.
- **Sync:** Engage this option to enable setting Flanger modulation speed to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate on a continuous scale.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the flanged signal (all the way right).

Phaser: This processor creates dreamy, otherworldly sweeping effects. The synth signal is fed through a series of all pass filters that alter its phase. When mixed with the dry signal, this creates a series of peaks and valleys in the frequency response that changes depending on the degree of phase shift applied. Phaser offers the following controls:

- **Mono:** Engage this option to sum the wet (affected) signal to mono.
- **Shift:** This control lets you specify the amount of phase shift to apply. Lower settings focus the phasing effect in the lower frequencies, while higher settings focus the effect in higher frequencies.
- **Speed:** This control lets you set the speed of modulation applied to the phase shift amount. Lower settings create slow, sweeping effects, while higher settings create faster, more aggressive modulation.
- **Width:** This control lets you set the degree to which the phase shift amount is modulated. Lower settings produce subtler effects, while higher settings produce more pronounced changes in timbre over time.
- **Feedback (FB):** This control lets you set the amount of output signal to feed back into the Phaser. Higher amounts of Feedback add to the resonance of the sweeping effect.
- **Sync:** Engage this option to enable setting Phaser modulation speed to a rhythmic value (such as 1/8th-note or 1/4-note) relative to Song tempo. Disengage to set Rate on a continuous scale.
- **Depth:** This control lets you blend between the dry signal (all the way left) and the phase-shifted signal (all the way right).

Bitcrusher

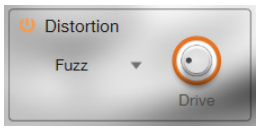
Perfect for audio abuse, Bitcrusher combines overdrive, bit-depth reduction, downsampling and clipping into a single plug-in. Bit depth reduction and downsampling are both digital resolution-reduction techniques, but each has its own sonic effect. When used in combination, they create a wide variety of tonal options.

The following parameters are available in Bitcrusher:

- **Dirt:** Enable this to introduce a high-frequency instability in the Bit Depth reduction effect. Good for creating aggressive sounds.
- **Depth:** Lets you specify the level of Bit Depth reduction to apply, from 24-bit to 1-bit.
- **Down:** Lets you specify the level of downsampling to apply.
- **Mix:** Lets you blend between the dry (0%) and effected (100%) signals.

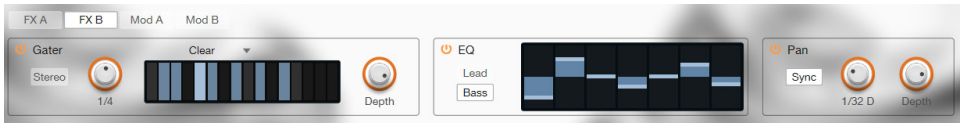
Distortion

This processor adds grit and character to your sounds. The distortion FX parameters within Sub Zero Bass are quite simple: Use the dropdown menu to choose from a variety of distortion types, from fizzy transistor fuzzes to thick, warm tube overdrives. Then, set the amount of distortion with the Drive knob.



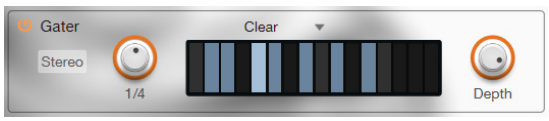
FX B

The second bank, FX B, includes Gater, Lead/Bass EQ, and Panning FX options.



Gater

The Gater will add a rhythmic gating effect, able to create a series of syncopated breaks in the synth signal. A variety of presets are provided, each with a different rhythmic gating pattern. However, the fun really begins when you create your own.

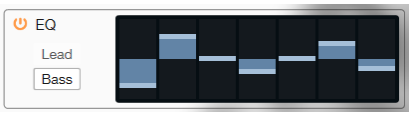


The Gater offers the following controls:

- **Stereo:** Engaging this option creates a separate beat grid for each side of the stereo field. When engaged, you'll see two rows of beat steps, the top row specifying gate steps for the left channel, and the bottom row gating the right channel.
- **Beats:** This control lets you set the length of the gating cycle, in rhythmic values (such as 1 bar or 1/2-note) relative to Song tempo. For example, at a setting of 1 bar, the 16 steps in the cycle repeat every bar, effectively representing 16th-notes. At a 1/2-note setting, the 16 steps repeat each half-bar, representing 32nd-note values.
- **Gate Presets:** There are nine provided gate presets to choose from.
- **Gate Steps:** This grid lets you specify which steps in the cycle let signals pass through, and which gate the signal to silence. Click on a step to enable or disable gating for that step.
- **Depth:** This control lets you blend between the gated and dry signals, allowing for rhythmic gating effects while retaining the continuity of the synth sound.

EQ

FX B includes a graphic equalizer effect, perfect for quick tonal shaping. Set the EQ bands to emphasize or attenuate bands of frequencies to suit your needs. When a band is in the center of its range, it neither adds or subtracts. When moved above the center, it emphasizes the chosen frequency. Moved below the center, it attenuates that frequency.



Choose between Lead mode, with frequencies chosen to suit aggressive, up-front sounds, or Bass mode, with wider ranging frequencies that work better for basses and mellower chordal parts.

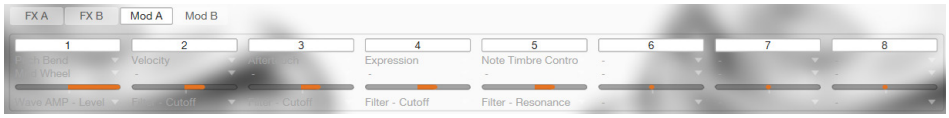
Pan

This is an auto-pan effect, which pans the synth signal left and right over time.

Pan offers the following controls:

- **Sync:** Enable this option to set pan speed to a rhythmic value (such as 1/4-note or 16th-note) relative to Song tempo. Disable this option to set pan speed along a continuous range.
- **Speed:** This control lets you set the speed at which the signal is panned left and right.
- **Depth:** This control lets you set the degree to which the signal is panned. Lower settings give a subtly panned effect, while higher settings pan the signal more radically, all the way to fully left and right in each cycle.

4.4.2.8 Modulation Matrix



If you would like to map additional modulation parameters, Sub Zero Bass provides 16 configurable modulation routings, split between two banks of eight (Mod A and Mod B). Modulation signals can be routed from a selection of incoming MIDI controller signals, including MPE data (such as Pitch Bend, Mod Wheel, Aftertouch, Poly Pressure, Note Timbre, and Pitch Control), modulation generators (such as the LFOs and envelopes), or the pitch or velocity of played notes.

These modulation signals can be used to vary most of the parameters throughout Sub Zero Bass, including modulation sources themselves (such as LFO 2 modulating the rate of LFO 1, or the Decay of Env 2). The following list describes the functions of each option within the Modulation Matrix, from top to bottom:

- **Bypass:** Each modulation slot has a bypass button at the top, which lets you enable or disable the flow of the modulation signal.
- **Main Source Selector:** Use this selector to specify the modulation signal you'd like to route to the Modifier Selector or Target.
- **Modifier Selector ("Via"):** Similar to the Main Source Selector, you can use this selector to specify the modulation signal you'd like to route to the Modifier Selector or Target. Additionally, you can use this selector in conjunction with the Main Source to further experiment with routing.
  - » For example, if you assign a modulation source to the input selector only, that signal is routed directly to the chosen destination. In some cases, you'll want to govern the flow of one mod source before it reaches its destination, using the signal from another mod source. On the other hand, you may want to control the output level of LFO 1 (routed to a parameter such as oscillator pitch) with the Mod Wheel. In this case, you'd choose Mod Wheel with the input selector, and LFO 1 with the modifier selector below.
- **Slider:** The slider underneath the selectors controls the amplitude and polarity of the modulation signal. Set at its center, no modulation occurs. Move the handle right of center to send an increasing amount of the modulation signal, at its normal (positive) polarity, to the chosen destination. Move it left of center to send the signal to its destination with a negative value.
  - » If the parameter you wish to modulate is set to a high value, you may want to send a negative modulation signal to it, driving the setting downward and causing more audible effects. Positive-going modulation signals are more efficient when modulating parameters set to low values.
- **Target Selector:** A selector at the bottom of each modulation slot lets you choose the destination of the chosen modulation signals. Signals can be mapped to any of the available destinations within Sub Zero Bass's Part Edit page.

Be sure to check out **www.fender.com** for the latest feature lists, compatibility, and computer requirements.

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