



# Final Cut Pro

## Logic Effects



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# Overview

## Logic effects included with Final Cut Pro

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

The most common processing options include EQs, levels (dynamic processors), modulations, distortions, spaces (reverbs), and echo (delays).

Further advanced features include precise signal meters and analyzers, noise reduction, bass enhancement, and vocal effects.

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

Effect category	Included effects
Distortion	<a href="#">Bitcrusher</a> <a href="#">Clip Distortion</a> <a href="#">Distortion</a> <a href="#">Distortion II</a> <a href="#">Overdrive</a> <a href="#">Phase Distortion</a> <a href="#">Ringshifter</a>
Echo	<a href="#">Delay Designer</a> <a href="#">Modulation Delay</a> <a href="#">Stereo Delay</a> <a href="#">Tape Delay</a>
EQ	<a href="#">AutoFilter</a> <a href="#">Channel EQ</a> <a href="#">Fat EQ</a> <a href="#">Linear Phase EQ</a>

<b>Effect category</b>	<b>Included effects</b>
Levels	<a href="#">Adaptive Limiter</a> <a href="#">Compressor</a> <a href="#">Enveloper</a> <a href="#">Expander</a> <a href="#">Gain</a> <a href="#">Limiter</a> <a href="#">Multichannel Gain</a> <a href="#">Multipressor</a> <a href="#">Noise Gate</a> <a href="#">Spectral Gate</a> <a href="#">Surround Compressor</a>
Modulation	<a href="#">Chorus</a> <a href="#">Ensemble</a> <a href="#">Flanger</a> <a href="#">Phaser</a> <a href="#">Scanner Vibrato</a> <a href="#">Tremolo</a>
Spaces	<a href="#">PlatinumVerb</a> <a href="#">Space Designer</a>
Specialized	<a href="#">Correlation Meter</a> <a href="#">Denoiser</a> <a href="#">Direction Mixer</a> <a href="#">Exciter</a> <a href="#">MultiMeter</a> <a href="#">Stereo Spread</a> <a href="#">SubBass</a> <a href="#">Test Oscillator</a>
Vocal	<a href="#">DeEsser</a> <a href="#">Pitch Correction</a> <a href="#">Pitch Shifter</a> <a href="#">Vocal Transformer</a>

# Add Logic effects to clips in Final Cut Pro

You can add Logic effects to audio clips using the Effects browser. After you add an effect, you can access the effect's interface and advanced controls from the Audio inspector.

## Add Logic effects to a clip

1. In Final Cut Pro, click the Effects button in the top-right corner of the timeline (or press Command-5).



2. In the Effects browser, select a type of audio effect, then select an effect from the Logic category on the right.

You can do any of the following to help make your selection:

- *Preview what an effect sounds like using the audio from the currently selected timeline clip:* Move the pointer over the audio effect thumbnail.
- *Preview changing an effect's primary control:* Hold down the Option key while moving the pointer over the audio effect thumbnail.
- *Filter the list of effects that appear:* Type text in the Effects browser search field.

3. Do one of the following:

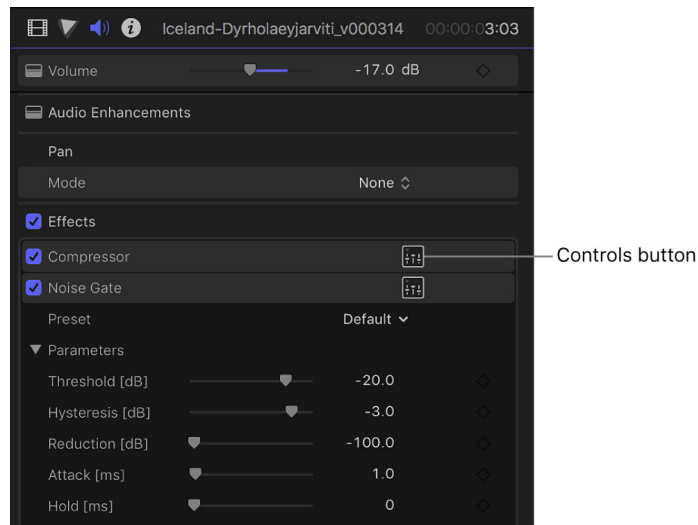
- Drag the effect to an audio clip (or a video clip with audio) in the timeline.
- Double-click the effect icon to apply the effect to the selected clip.

The effect appears in the Effects section of the Audio inspector and in the Audio Animation editor.

## Show and adjust Logic effect controls

1. In the Final Cut Pro timeline, select the clip with the effect you want to adjust.
2. To open the Audio inspector, click the Inspector button on the right side of the toolbar (or press Command-4), then click the Audio button (speaker icon) at the top.
3. Locate the effect in the Effects section of the Audio inspector.

4. To show the effect's controls, click the Controls button (to the right of the effect's name).



5. Select effect parameters and adjust their settings as needed.

You can preview your adjustments by using the skimmer or playing the clip in the timeline.

To return the effect's values to their default settings, click the effect's Reset button ↶.

## More Final Cut Pro resources

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

### Final Cut Pro website

For general information and updates, as well as the latest news on Final Cut Pro, go to:

[apple.com/finalcutpro](https://apple.com/finalcutpro)

### Apple Support

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

[apple.com/support](https://apple.com/support)

For software updates, documentation, discussion forums, and answers to the most frequently asked questions for Final Cut Pro, go to:

[apple.com/support/finalcutpro](https://apple.com/support/finalcutpro)



# Distortion effects

## Intro to distortion effects in Final Cut Pro

You can use Distortion effects to re-create the sound of analog or digital distortion and to radically transform your audio.

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

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

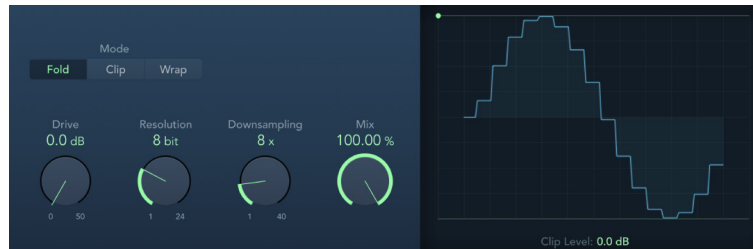
Distortion effects include parameters for *tone*, which let you shape the way the distortion alters the signal (often as a frequency-based filter), and for *gain*, which let you control how much the distortion alters the output level of the signal.

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

# Bitcrusher controls in Final Cut Pro

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

For information about adding the Bitcrusher effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- **Mode buttons:** Set the distortion mode to Fold, Clip, or Wrap. Signal peaks that exceed the clip level are processed.

**Note:** The Clip Level parameter has a significant impact on the behavior of all three modes. This is reflected in the waveform display, so try each mode button and adjust the Clip Level slider to get a feel for how this works.

- **Fold button:** Set a softer distortion by halving the level of the center portion of the signal above the threshold. The start and end levels of the clipped signal are unchanged.
- **Clip button:** Enable to cause an abrupt distortion when the clipping threshold is exceeded. Clipping that occurs in most digital systems is closest to Cut mode.
- **Wrap button:** Set a less severe distortion by offsetting the start, mid, and end levels of the signal above the threshold. This parameter smooths signal levels when they cross the threshold. The center portion of the clipped signal is also softer than in Cut mode.
- **Drive knob and field:** Set the amount of gain applied to the input signal.  
**Note:** Raising the Drive level also tends to increase the amount of clipping at the effect output.
- **Resolution knob and field:** Set the bit rate (between 1 and 24 bits) to alter the calculation precision of the process. Lower values increase the number of sampling errors, generating more distortion. At extremely low bit rates, the amount of distortion can be greater than the level of the usable signal.
- **Downsampling knob and field:** Reduce the sample rate. A value of 1x has no effect on the signal, a value of 2x halves the sample rate, and a value of 10x reduces the sample rate to one-tenth of the original. (For example, if you set Downsampling to 10x, a 44.1 kHz signal is sampled at just 4.41 kHz.)  
**Note:** Downsampling has no impact on the playback speed or pitch of the signal.
- **Mix knob and field:** Set the balance between the dry and crushed signal.
- **Waveform display:** Shows the impact of parameters on the distortion process.
- **Clip Level handle and field:** Set the point (below the clipping threshold of the channel) at which the signal starts clipping. Drag the green dot at the top left, or drag vertically in the field, to set the clip level.

# Clip Distortion controls in Final Cut Pro

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

Clip Distortion has an unusual combination of serially connected filters. The incoming signal is amplified by the Drive value, passes through a highpass filter, then is subjected to nonlinear distortion. Following the distortion, the signal passes through a lowpass filter. The effect signal is then recombined with the original signal, and this mixed signal is sent through a further lowpass filter. All three filters have a slope of 6 dB/octave.

This unique combination of filters allows for gaps in the frequency spectra that can sound good with this sort of nonlinear distortion.

For information about adding the Clip Distortion effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

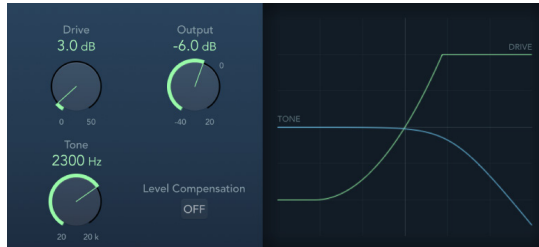


- *Input Gain knob and field:* Set the amount of gain applied to the plug-in input signal. This behaves like a preamplifier for the Drive parameter.
- *Drive knob and field:* Set the amount of additional gain (distortion) applied to the input signal. After being amplified by Drive, the signal passes through a highpass filter.
- *Clip circuit display:* Shows the impact of all parameters.
  - *Tone handle and field:* Set the cutoff frequency (in hertz) of the highpass filter.
  - *Symmetry handle and field:* Set the amount of nonlinear (asymmetrical) distortion applied to the signal.
  - *Clip Filter handle and field:* Set the cutoff frequency (in hertz) of the first lowpass filter.
- *Mix knob and field:* Set the ratio between the effect (wet) signal and original (dry) signals, following the clip filter.
- *High Shelving knob and field:* Set the frequency (in hertz) of the high shelving filter. If you set the High Shelving frequency to around 12 kHz, you can use it like the treble control on a mixer channel strip or a stereo hi-fi amplifier. Unlike these types of treble controls, however, you can boost or cut the signal by up to  $\pm 30$  dB with the Gain parameter.
- *LP Filter knob and field:* Set the cutoff frequency (in hertz) of the lowpass filter. This processes the mixed signal.
- *Gain knob and field:* Set the amount of gain applied to the output of signals above the high shelving filter frequency.
- *Output Gain knob and field:* Set the amount of gain applied to the plug-in output signal.

## Distortion controls in Final Cut Pro

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

For information about adding the Distortion effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Drive knob and field:* Set the amount of saturation applied to the signal.
- *Tone knob and field:* Set the frequency for the high cut filter. Filtering the harmonically rich distorted signal produces a softer tone.
- *Output knob and field:* Set the output level.
- *Level Compensation button:* Turn on to reference the overall processing of the signal to 0 dB. This compensates for increases in loudness caused by adding distortion.
- *Display:* Shows the impact of parameters on the signal.

## Distortion II controls in Final Cut Pro

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

For information about adding the Distortion II effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Pre Gain knob and field:* Set the amount of gain applied to the input signal.
- *Drive knob and field:* Set the amount of saturation applied to the signal.
- *Tone knob and field:* Boost the integrated high shelf filter gain both pre- and post-distortion, to achieve a different tone.

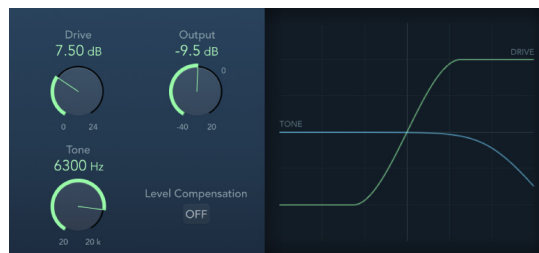
- *Pop-up menu:* Choose the type of distortion.
  - *Growl:* Emulates a two-stage tube amplifier similar to the type found in a Leslie 122 speaker cabinet, which is often used with the Hammond B3 organ.
  - *Bity:* Emulates the sound of a bluesy (overdriven) guitar amp.
  - *Nasty:* Produces hard distortion, suitable for creating very aggressive sounds.
  - *Class AB soft:* Emulates a softer crossover distortion encountered in transistorized amplifiers when driving a higher load.
  - *Class AB hard:* Emulates a harder crossover distortion encountered in transistorized amplifiers when driving a higher load.

*Crossover*, in this case, refers to switching between two matched transistors. In Class AB designs, both transistors remain on at all times, reducing the effect of hard switching between them.

## Overdrive controls in Final Cut Pro

Overdrive emulates the distortion produced by a field effect transistor (FET), commonly used in solid-state musical instrument amplifiers and hardware effects devices. When saturated, FETs generate a warmer-sounding distortion than bipolar transistors, such as those emulated by the [Distortion effect](#).

For information about adding the Overdrive effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



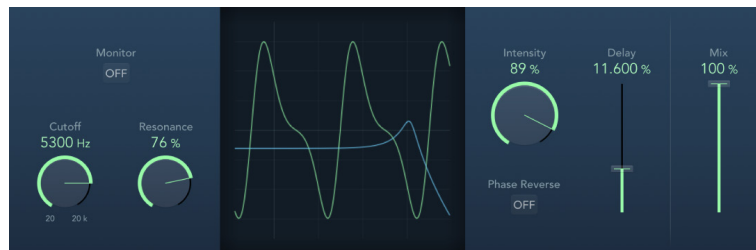
- *Drive knob and field:* Set the saturation amount for the simulated FET transistor.
- *Tone knob and field:* Set the frequency of the high cut filter. Filtering the harmonically rich distorted signal produces a softer tone.
- *Output knob and field:* Set the output level.
- *Level Compensation button:* Turn on to reference the overall processing of the signal to 0 dB. This compensates for increases in loudness caused by using overdrive.
- *Display:* Shows the impact of parameters on the signal.

# Phase Distortion controls in Final Cut Pro

The Phase Distortion effect is based on a modulated delay line, similar to a chorus or flanger effect (see [Intro to modulation effects in Final Cut Pro](#)). Unlike these effects, however, the delay time is not modulated by a low-frequency oscillator (LFO), but rather by a lowpass-filtered version of the input signal itself, using an internal side chain. This means that the incoming signal modulates its own phase position.

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

For information about adding the Phase Distortion effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Monitor button:* Turn on to hear the input signal in isolation. Turn off to hear the mixed signal.
- *Cutoff knob and field:* Set the (center) cutoff frequency of the lowpass filter.
- *Resonance knob and field:* Emphasize frequencies surrounding the cutoff frequency.
- *Display:* Shows the impact of parameters on the signal.
- *Intensity knob and field:* Set the amount of modulation applied to the signal.
- *Phase Reverse button:* Turn on to reduce the delay time on the right channel when input signals that exceed the cutoff frequency are received. Available only for stereo instances of the Phase Distortion effect.
- *Delay slider and field:* Set the maximum delay time.
- *Mix slider and field:* Set the percentage of the effect signal mixed with the original signal.

## Ringshifter

### Intro to Ringshifter in Final Cut Pro

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

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

The frequency shifter moves the frequency content of the input signal by a fixed amount and, in doing so, alters the frequency relationship of the original harmonics. The resulting sounds range from sweet and spacious phasing effects to strange robotic timbres.

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

For information about adding the Ringshifter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

## Ringshifter interface in Final Cut Pro

The Ringshifter interface consists of six main sections.



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

## Ringshifter mode buttons in Final Cut Pro

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

For information about adding the Ringshifter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



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

## Ringshifter Oscillator controls in Final Cut Pro

In both frequency shifter modes and the ring modulator OSC mode, the Ringshifter effect's internal sine wave oscillator is used to modulate the amplitude of the input signal.

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

For information about adding the Ringshifter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).





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

## Ringshifter Delay controls in Final Cut Pro

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

For information about adding the Ringshifter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Time knob and field*: Set the delay time. This is in Hz when running freely, or in note values (including triplet and dotted notes) when the Sync button is active.
- *Sync button*: Synchronize the delay to the project tempo. You can choose musical note values with the Time knob.
- *Level knob and field*: Set the level of the delay added to the ring-modulated or frequency-shifted signal. A Level value of 0 passes the effect signal directly to the output (bypass).

## Ringshifter Envelope Follower controls in Final Cut Pro

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

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

For information about adding the Ringshifter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Power button:* Turn the envelope follower on or off.
- *Sens (Sensitivity) slider and field:* Determine how responsive the envelope follower is to the input signal. At lower settings, the envelope follower reacts only to the most dominant signal peaks. At higher settings, the envelope follower tracks the signal more closely, but may react less dynamically.
- *Attack slider and field:* Set the response time of the envelope follower.
- *Decay slider and field:* Control the time it takes the envelope follower to return from a higher to a lower value.

## Ringshifter LFO modulation controls in Final Cut Pro

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

For information about adding the Ringshifter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



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

## Ringshifter Output controls in Final Cut Pro

The Ringshifter effect's Output controls are used to set the balance between the effect and input signals and also to set the width and feedback of Ringshifter.

For information about adding the Ringshifter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



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

# Echo effects

## Intro to echo effects in Final Cut Pro

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

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

You can use delays to double individual sounds to resemble a group of instruments playing the same melody, to create echo effects, to place the sound in a large “space,” to generate rhythmic effects, or to enhance the stereo position of an audio clip.

Echo effects are generally used as individual audio clip effects. They are rarely used on an overall mix, unless you’re trying to achieve an unusual effect.

## Delay Designer

### Intro to Delay Designer in Final Cut Pro

Delay Designer is a *multitap* delay. Unlike traditional delay units that offer only one or two delays (or taps) that may or may not be fed back into the circuit, Delay Designer provides up to 26 individual taps. These taps are all fed from the source signal and can be freely edited to create delay effects that have never been heard before.

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

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

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

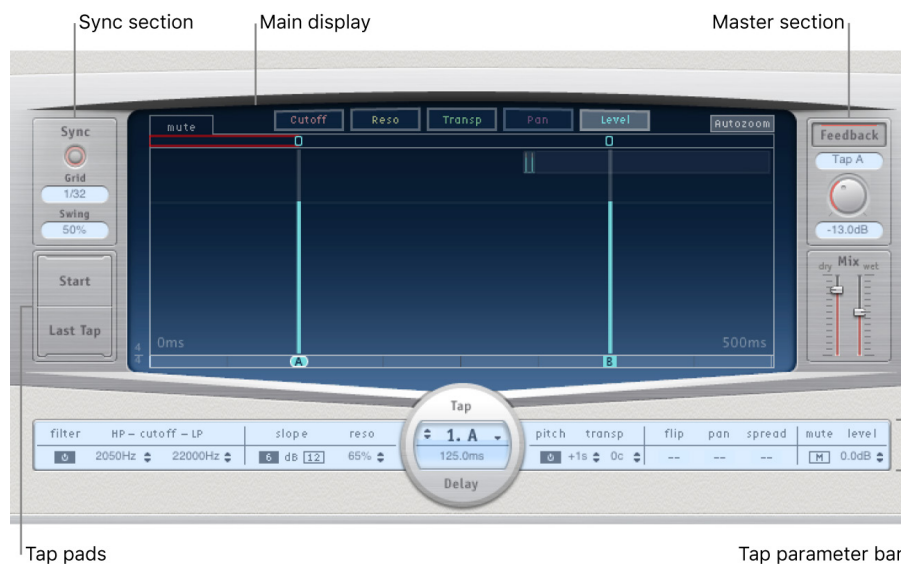
As the name implies, Delay Designer offers significant sound design potential. You can use it for everything from a basic echo effect to an audio pattern sequencer. You can create complex, evolving, moving rhythms by synchronizing the placement of taps. This leads to further musical possibilities when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

You can use Delay Designer with mono, stereo, or surround clips. See [Work with Delay Designer in surround in Final Cut Pro](#) for details on using it in surround.

For information about adding the Delay Designer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

## Delay Designer interface in Final Cut Pro

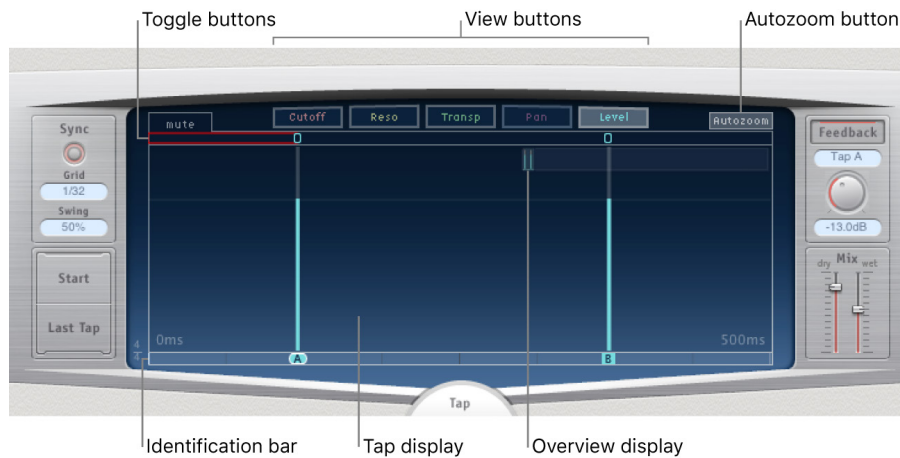
The Delay Designer interface consists of five main sections.



- **Main display:** Provides a graphic representation of all taps. You can see, and edit, the parameters of each tap in this area. See [Delay Designer main display in Final Cut Pro](#).
- **Tap parameter bar:** Offers a numeric overview of the current parameter settings for the selected tap. You can view and edit the parameters of each tap in this area. See [Delay Designer Tap parameter bar in Final Cut Pro](#).
- **Tap pads:** You can use these two pads to create taps in Delay Designer. See [Create taps in Delay Designer in Final Cut Pro](#).
- **Sync section:** This is used for syncing tempo in Logic Pro and is disabled for use with Final Cut Pro.
- **Master section:** This area contains the global Mix and Feedback controls. See [Delay Designer Master section controls in Final Cut Pro](#).

## Delay Designer main display in Final Cut Pro

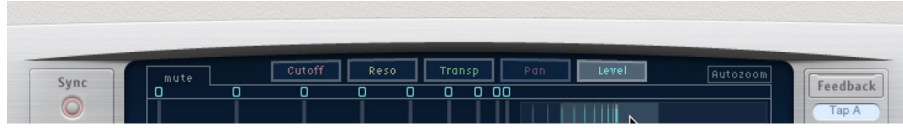
Delay Designer's main display is used to view and edit tap parameters. You can freely determine the parameter shown, and quickly zoom or navigate through all taps.



- *View buttons*: Determine the parameter or parameters represented in the Tap display. See [Delay Designer view buttons in Final Cut Pro](#).
- *Autozoom button*: Zoom the Tap display out, making all taps visible. Turn Autozoom off if you want to zoom the display in (by dragging vertically on the Overview display) to view specific taps.
- *Overview display*: Shows all taps in the time range. See [Navigate the Delay Designer Tap display in Final Cut Pro](#).
- *Toggle buttons*: Click to enable or disable the parameters of a particular tap. The parameter being toggled is chosen with the view buttons. The label at the left of the toggle bar always indicates the parameter being toggled. For more information, see [Use the Delay Designer tap toggle buttons in Final Cut Pro](#).
- *Tap display*: Represents each tap as a shaded line. Each tap contains a bright bar (or dot for stereo panning) that indicates the value of the parameter. You can directly edit tap parameters in the Tap display area. For more details, see [Edit taps in the Delay Designer Tap display in Final Cut Pro](#).
- *Identification bar*: Shows an identification letter for each tap. It also serves as a time position indicator for each tap. You may freely move taps backward or forward in time along this bar/timeline. See [Move and delete taps in Delay Designer in Final Cut Pro](#).

## Delay Designer view buttons in Final Cut Pro

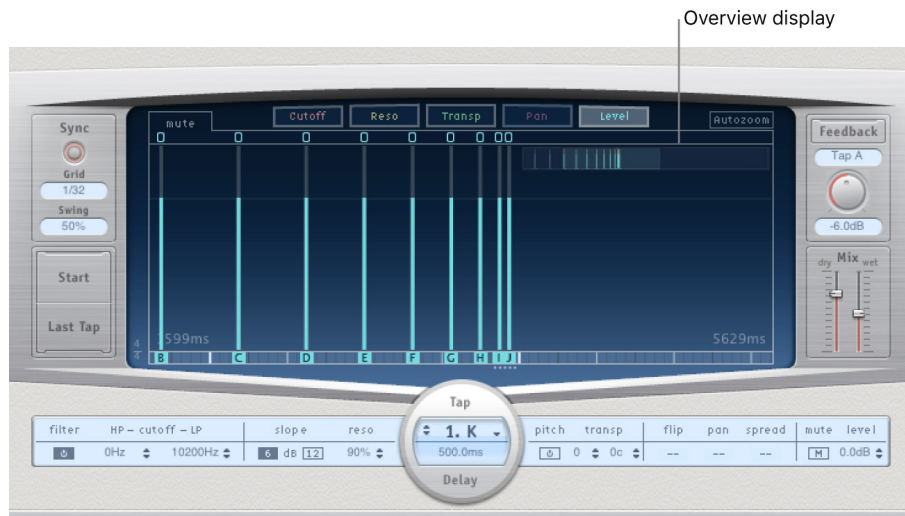
The view buttons determine which control is represented in Delay Designer's Tap display.



- **Cutoff button:** Show the highpass and lowpass filter cutoff frequencies of taps.
- **Reso (Resonance) button:** Show the filter resonance value of each tap.
- **Transp (Transpose) button:** Show the pitch transposition of each tap.
- **Pan button:** Show the pan parameter of each tap.
  - For mono to stereo channels, each tap contains a line showing its pan position.
  - For stereo to stereo channels, each tap contains a dot showing its stereo balance. A line extending outwards from the dot indicates the tap's stereo spread.
  - For surround channels, each tap contains a line representing its surround angle (for details, see [Work with Delay Designer in surround in Final Cut Pro](#)).
- **Level button:** Show the relative volume level of each tap.
  - **Tip:** You can temporarily switch the Tap display to Level view from one of the other view modes by pressing Command-Option.

## Navigate the Delay Designer Tap display in Final Cut Pro

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



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

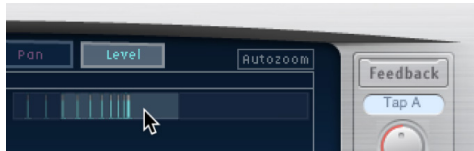
## Zoom the Tap display

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

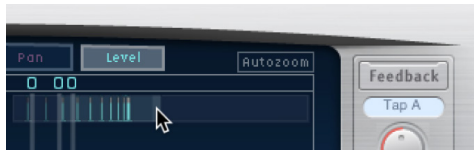
For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Do one of the following:

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



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



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

## Move to different sections of the Tap display

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Horizontally drag the middle of the bright rectangle in the Overview display.

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



## Create taps in Delay Designer in Final Cut Pro

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

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

After a tap has been created, you can freely adjust its position, or you can remove it if it was created accidentally. For details, see [Move and delete taps in Delay Designer in Final Cut Pro](#).

### Create taps with the Tap pad

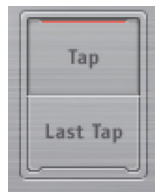
1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Click the upper pad (Start).

*Note:* Whenever you click the Start pad, it automatically erases all existing taps. To avoid erasing your initial taps, create subsequent taps by clicking in the Identification bar.

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



3. Click the Tap button to begin recording new taps.
4. Click the Tap button to create new taps.

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

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

This adds the final tap, ending tap recording and assigning the last tap as the *feedback tap* (for an explanation of the feedback tap, see [Delay Designer Master section controls in Final Cut Pro](#)).

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

### Create taps in the Identification bar

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Click at the appropriate position.



### Copy taps in the Identification bar

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

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

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

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

The Identification bar shows the letter of each visible tap. The Tap Delay field of the Tap parameter bar displays the letter of the currently selected tap, or the letter of the tap being edited when multiple taps are selected (for details, see [Select taps in Delay Designer in Final Cut Pro](#)).

## Select taps in Delay Designer in Final Cut Pro

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



### Select a tap

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.  
For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).
2. Do one of the following:
  - Click a tap in the Tap display.
  - Click the appropriate tap letter in the Identification bar.
  - Click one of the arrows to the left of the tap name to select the next or previous tap.
  - Click the pop-up menu to the right of the tap name, and choose the appropriate tap letter.



### Select multiple taps

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.  
For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).
2. Do one of the following:
  - Drag across the background of the Tap display to select multiple taps.
  - Shift-click specific taps in the Tap display to select multiple nonadjacent taps.

## Move and delete taps in Delay Designer in Final Cut Pro

In the Delay Designer effect, you can move a tap backward or forward in time, or completely remove it.

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

### Move a selected tap in time

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Select the tap in the Identification bar, and drag it to the left to go forward in time, or to the right to go backward in time.

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

*Note:* Editing the Delay Time parameter in the Tap Delay field of the Tap parameter bar also moves a tap in time. For more details about the Tap Delay field and editing taps, see [Delay Designer Tap parameter bar in Final Cut Pro](#).

### Delete taps

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Do one of the following:

- *Delete a tap:* Select it and press the Delete or Backspace key.

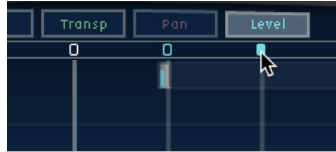
You can also select a tap letter in the Identification bar and drag it downward, out of the Tap display. This method also works when more than one tap is selected.



- *Delete all selected taps:* Control-click (or right-click) a tap, and choose "Delete tap(s)" from the shortcut menu.

## Use the Delay Designer tap toggle buttons in Final Cut Pro

In the Delay Designer effect, each tap has its own toggle button in the Toggle bar. These buttons offer you a quick way to graphically activate and deactivate parameters. The specific parameter being toggled by the toggle buttons depends on the current view button selection.



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

### Temporarily switch the mute state of taps

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.  
For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).
2. Option-Command-click a toggle button, regardless of the current view mode.  
When you release the Option and Command keys, the toggle buttons return to their standard functionality in the active View mode.

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

## Edit taps in the Delay Designer Tap display in Final Cut Pro

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

### Edit a tap parameter in the Tap display

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.  
For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).
2. Click the view button of the parameter you want to edit.

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

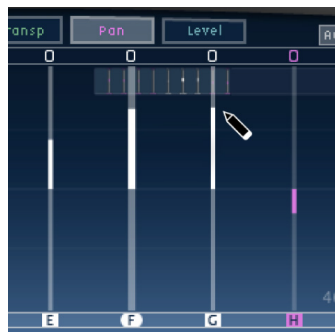


If you have chosen multiple taps, the values of all selected taps are changed relative to each other.

*Note:* The method outlined above is slightly different for the Filter Cutoff and Pan parameters. See [Edit Delay Designer filter cutoff in Final Cut Pro](#) and [Edit Delay Designer pan in Final Cut Pro](#).

### Set the values of multiple taps

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.  
For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).
2. Command-drag horizontally and vertically across several taps in the Tap display.  
Parameter values change to match the mouse position as you drag across the taps. Command-dragging across several taps allows you to draw value curves, much like using a pencil to create a curved line on a piece of paper.



## Align Delay Designer tap values in Final Cut Pro

You can use Delay Designer's Tap display to graphically align tap parameter values that are represented as vertical lines.

### Align the values of several taps

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Command-click in the Tap display, and move the pointer while holding down the Command key.

This results in a line trailing behind the pointer.



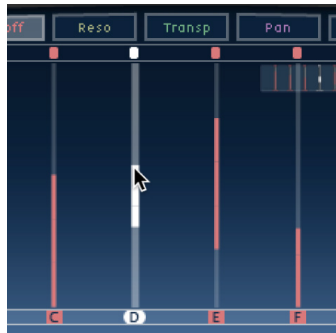
3. Click the appropriate position to mark the end point of the line.

The values of taps that fall between the start and end points are aligned along the line.



## Edit Delay Designer filter cutoff in Final Cut Pro

Whereas the steps outlined in [Edit taps in the Delay Designer Tap display in Final Cut Pro](#) apply to most graphically editable parameters, the Cutoff and Pan parameters work in a slightly different fashion.



In Cutoff view, each tap actually shows two parameters: highpass and lowpass filter cutoff frequency. You can adjust the filter cutoff values independently by dragging the specific cutoff frequency line—the upper line is lowpass and the lower line is highpass—or you can adjust both cutoff frequencies by dragging between them.

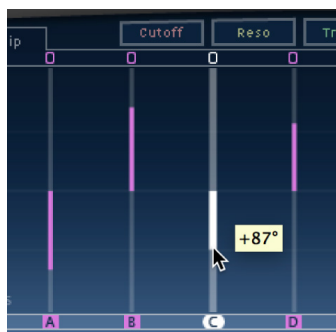
When the highpass filter cutoff frequency value is lower than that of the lowpass cutoff frequency, *only one line is shown*. This line represents the frequency band that passes through the filters—in other words, the filters act as a bandpass filter. In this configuration, the two filters operate *serially*,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

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

## Edit Delay Designer pan in Final Cut Pro

The way the Delay Designer effect's Pan parameter is represented in Pan view is entirely dependent on the input channel configuration—mono to stereo, stereo to stereo, or surround.

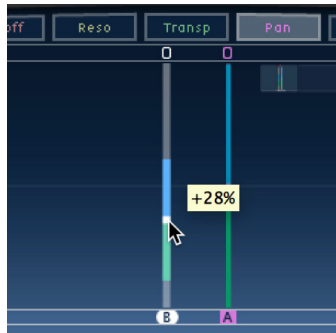
*Note:* Pan is not available in mono configurations.





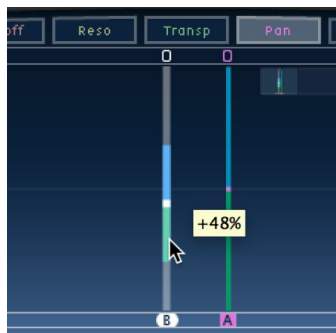
In mono input/stereo output configurations, all taps are initially panned to the center. To edit the pan position, drag vertically *from the center* of the tap in the direction you wish to pan the tap, or taps. A white line extends outward from the center in the direction you have dragged, reflecting the pan position of the tap, or taps.

Lines above the center position indicate pans to the left, and lines below the center position denote pans to the right. Left (blue) and right (green) channels are easily identified.



In stereo input/stereo output configurations, the Pan parameter adjusts the stereo balance, not the position of the tap in the stereo field. The Pan parameter appears as a *dot* on the tap, which represents stereo balance. Drag the dot up or down the tap to adjust the stereo balance.

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



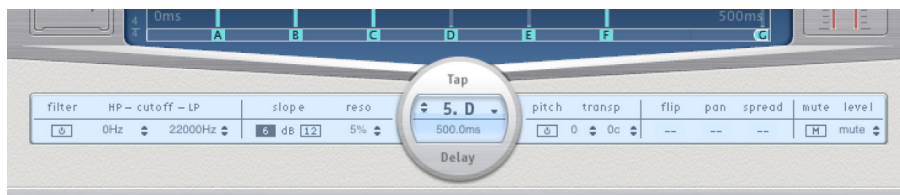
In surround configurations, the bright line represents the surround angle. For more information, see [Work with Delay Designer in surround in Final Cut Pro](#).

## Delay Designer Tap parameter bar in Final Cut Pro

The Delay Designer effect's Tap parameter bar provides instant access to all parameters of the chosen tap. The Tap parameter bar also provides access to several parameters that are not available in the Tap display, such as Transpose and Flip.

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

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



- **Filter On/Off button:** Enable or disable the highpass and lowpass filters for the selected tap.
- **HP-Cutoff-LP fields:** Set the cutoff frequencies (in Hz) for the highpass and lowpass filters.
- **Slope buttons:** Determine the steepness of the highpass and lowpass filter slope. Click the 6 dB button for a gentler filter slope, or click the 12 dB button for a steeper, more pronounced filtering effect.  
*Note:* You cannot set the slope of the highpass and lowpass filters independently.
- **Reso (Resonance) field:** Set the amount of filter resonance for both filters.
- **Tap Delay fields:** Show the number and name of the selected tap in the upper section and the delay time in the lower section.
- **Pitch On/Off button:** Enable or disable pitch transposition for the selected tap.
- **Transp (Transpose) fields:** The left field sets the amount of pitch transposition in semitones. The right field fine-tunes each semitone step in cents (1/100th of a semitone).
- **Flip buttons:** Swap the left and right side of the stereo or surround image. Clicking these buttons reverses the tap position from left to right, or vice versa. For example, if a tap is set to 55% left, clicking the flip button changes it to 55% right.
- **Pan field:** Control the pan position for mono input signals, stereo balance for stereo input signals, and surround angle when used in surround configurations.
  - Pan displays a percentage between 100% (full left) and -100% (full right), which represents the pan position or balance of the tap. A value of 0% represents the center panorama position.
  - When used in surround, a surround panner replaces the percentage representation. For more information, see [Work with Delay Designer in surround in Final Cut Pro](#).
- **Spread field:** When a stereo-to-stereo or stereo-to-surround instance of Delay Designer is used, Spread sets the width of the stereo spread for the selected tap.
- **Mute button:** Mute or unmute the selected tap.
- **Level field:** Determine the output level for the selected tap.

## Delay Designer tap shortcut menu in Final Cut Pro

Control-click (or right-click) a tap in Delay Designer's Tap display to open a shortcut menu containing the following commands:

- *Copy sound parameters*: Copies all parameters (except the delay time) of the selected tap or taps into the Clipboard.
- *Paste sound parameters*: Pastes the tap parameters from the Clipboard into the selected tap or taps. If there are more taps in the Clipboard than are selected in the Tap display, the extra taps in the Clipboard are ignored.
- *Reset sound parameters to default values*: Resets all parameters of all selected taps (except the delay time) to the default values.
- *2 x delay time*: Doubles the delay time of all selected taps. For example, the delay times of three taps are set as follows: Tap A = 250 ms, Tap B = 500 ms, Tap C = 750 ms. If you select these three taps and choose the "2 x delay time" shortcut menu command, the taps will be changed as follows: Tap A = 500 ms, Tap B = 1000 ms, Tap C = 1500 ms. In other words, a rhythmic delay pattern will unfold half as fast. (In musical terms, it will be played in half time.)
- *1/2 x delay time*: Halves the delay time of all selected taps. Using the example above, choosing the "1/2 x delay time" shortcut menu command changes the taps as follows: Tap A = 125 ms, Tap B = 250 ms, Tap C = 375 ms. In other words, a rhythmic delay pattern will unfold twice as fast. (In musical terms, it will be played in double time.)
- *Delete tap(s)*: Deletes all selected taps.

## Reset Delay Designer tap values in Final Cut Pro

You can use Delay Designer's Tap display and Tap parameter bar to reset tap parameters to their default values.

1. In the Final Cut Pro timeline, select a clip with the Delay Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Do one of the following:

- In the Tap display, Option-click a tap to reset the chosen parameter to its default setting.

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

- In the Tap parameter bar, Option-click a parameter value to reset it to the default setting.

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

## Delay Designer Master section controls in Final Cut Pro

The Delay Designer effect's Master section incorporates two global functions: delay feedback and dry/wet mix.

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

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



- *Feedback button*: Enable or disable the feedback tap.
- *Feedback Tap pop-up menu*: Choose a tap as the feedback tap.
- *Feedback Level knob*: Set the feedback level. You can vary the feedback tap output level before it is routed back into Delay Designer's input.
  - A value of 0% equals no feedback.
  - A value of 100% sends the feedback tap back into Delay Designer's input at full volume.

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

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

## Work with Delay Designer in surround in Final Cut Pro

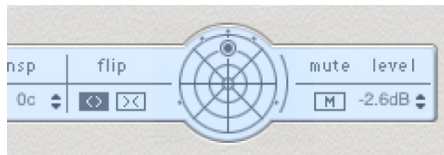
Delay Designer's design is optimized for use in surround configurations. With 26 taps that can be freely positioned in the surround field, you can create some truly amazing rhythmic and spatial effects.

Delay Designer always processes each input channel independently.

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

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

*Note:* In the Tap display's Pan view mode, you can only adjust the *angle* of taps. You must use the surround panner on the Tap parameter bar to adjust diversity.



To easily move the surround position, you can:

- Command-drag to adjust diversity.
- Option-Command-drag to adjust the angle.
- Option-click the blue dot to reset the angle and diversity.

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

## Modulation Delay controls in Final Cut Pro

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

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

For information about adding the Modulation Delay effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Mod Intensity knob and field:* Set the modulation amount.
- *Time knob and field:* Set the basic delay time. Set to the far left position to create flanger effects, to the center for chorus effects, and to the far right to hear clearly discernible delays.
- *Feedback knob and field:* Set the amount of effect signal routed back to the input. Use a high Feedback value for strong modulations. If you want to double the signal, don't use Feedback. Negative values invert the phase of the feedback signal, resulting in more chaotic effects.
- *De-Warble button:* Turn on to make sure the pitch of the modulated signal remains constant.
- *Constant Mod button:* Turn on to make sure the modulation width remains constant, regardless of the modulation rate.  
*Note:* When Constant Mod is turned on, higher modulation frequencies reduce the modulation width.
- *D-Mode button:* Turn on to introduce a spatial filtering effect that resembles a well-known vintage processor.  
*Note:* Available only in stereo or mono-to-stereo instances.
- *LFO 1 and LFO 2 Rate knobs and fields:* Set the modulation rate for the left and right stereo channels. In surround instances, the center channel is assigned the middle value of the left and right LFO Rate knobs. The other channels are assigned values between the left and right LFO rates.  
*Note:* The right LFO Rate knob is available only in stereo and surround instances, and it can be set separately only if Link L & R is not turned on.

- *Link L & R button*: Link the modulation rates of the left and right stereo channels. Adjusting either Rate knob affects the other channel in stereo instances, or other channels in surround instances.
- *Mix slider and fields*: Determine the balance between the two LFOs.
- *Phase knob and field*: Control the phase relationship between individual channel modulations. Available only in stereo and surround instances.
  - At 0°, the extreme values of the modulation are achieved simultaneously for all channels.
  - At 180° or -180°, you achieve the greatest possible distance between the modulation phases of the channels.

*Note*: The Phase controls are available only if Link L & R is turned on.

- *Distribution pop-up menu*: Choose how phase offsets between individual channels are distributed in the surround field. Choose Circular, Left↔Right, Front↔Rear, Random, or New Random. Available only in surround instances.

*Note*: When you load a setting that uses the Random option, the saved phase offset value is recalled. If you want to randomize the phase setting again, choose New Random from the Distribution pop-up menu.

- *Filter button*: Turn on to introduce an additional allpass filter into the signal path. This filter shifts the phase angle of a signal, influencing its stereo image.
- *Low Cut and High Cut sliders and fields*: Set the frequency at which the phase shift crosses 90°—the halfway point of the total 180°—for each of the stereo channels. In surround instances, the other channels are automatically assigned values that fall between the two settings.
- *Volume Mod slider and field*: Determine the impact of LFO modulation on the amplitude of the effect signal.
- *Output Mix slider and field*: Set the balance between dry and wet signals.

## Stereo Delay controls in Final Cut Pro

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

*Note*: If you use Stereo Delay on mono channel strips, the track or bus switches to two-channel operation from the point of insertion—all Insert slots after the chosen slot are stereo.

For information about adding the Stereo Delay effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



The controls for the left and right delays are identical. The following descriptions apply to both channels. The Global and Output Mix controls are described after the channel controls.

## Channel controls

- *Input pop-up menu:* Choose the input signal for the two stereo sides. Options include Off, Left, Right, L + R, and L - R.
- *Delay Time knob and field:* Set the delay time in milliseconds or in note values when synchronized with the clip. Notes (and dots) are displayed around the Delay Time knob when synchronized with the clip. Click the notes or dots (or rotate the knob) to choose an exact synchronization value.  
*Note:* Clicking note or dot values resets the Deviation parameter value. Choose a value from the Note pop-up menu to retain the current Deviation value.
- *Note pop-up menu:* Choose the grid resolution for the delay time when Tempo Sync is turned on.
- *Deviation field:* Set the amount of deviation from the grid.
- *Delay Time : 2 and x 2 buttons:* Halve or double the current delay time.
- *Low Cut and High Cut slider and field:* Cut frequencies below the Low Cut value and above the High Cut value from the source signal.
- *Feedback knob and field:* Set the amount of feedback for the left and right delay signals.
- *Feedback Phase button:* Invert the phase of the corresponding channel feedback signal.
- *Crossfeed Left to Right (or Right to Left) knob and field:* Transfer the feedback signal of the left channel to the right channel, and vice versa.
- *Crossfeed Phase button:* Invert the phase of the crossfed feedback signals.



## Global and Output Mix controls

- *Routing pop-up menu:* Choose the internal signal routing. Options include Customized, Straight, Crossfeed, 90/10, 10/90, Ping Pong L/R, Pan L/R, and Rotate L/R.
  - *Tempo Sync button:* Synchronize delay repeats with the clip. Set note values with the Note pop-up menu or Delay Time knob.
  - *Stereo Link button:* Turn Stereo Link on to make the same adjustments for both channels. Adjusting one channel value adjusts the other. Relative values are maintained.
    - Press Command to temporarily flip stereo linking, allowing you to adjust a control in a linked fashion even when Stereo Link is not turned on.
    - Command-drag a slider to adjust a single channel when Stereo Link is turned on.
- Note:* Stereo Link can be automatically turned off when you choose a new routing or setting. To turn it back on, click the Stereo Link button.
- *Output Mix sliders and fields:* Independently control the level of the left and right channel signals.

## Tape Delay controls in Final Cut Pro

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

For information about adding the Tape Delay effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Tempo Sync button*: Synchronize delay repeats with the clip. Set note values with the Note pop-up menu or the Delay Time knob.
- *Delay Time knob and field*: Set the delay time in milliseconds. Notes (and dots) are displayed around the Delay Time knob when the delay time is synchronized with the clip. Click the notes or dots (or rotate the knob) to choose an exact synchronization value.  
*Note*: Clicking note or dot values resets the Deviation parameter value. Choose a value from the Note pop-up menu to retain the current Deviation value.
- *Delay Time : 2 and x 2 buttons*: Halve or double the current delay time.
- *Note pop-up menu*: Set the grid resolution for the delay time.
- *Deviation field*: Set the amount of deviation from the grid.
- *Smoothing slider and field*: Even out the LFO and flutter effect. See the LFO and Flutter parameter descriptions, below.
- *Clip Threshold knob*: Set the level of the distorted tape saturation signal. Higher values produce no additional audible distortion. Lower values result in an aggressive distortion. This behavior is influenced by high Feedback values, which result in eventual distortion, irrespective of the Clip Threshold value. However, aggressive distortion and signal breakup are achieved far more rapidly when a low Clip Threshold level is used.
- *Spread knob and field*: Set the width of the effect signal in stereo instances. This parameter is not available in mono instances.
- *Tape Head Mode buttons*: Click either Clean or Diffuse to emulate a different tape head position. This affects the behavior of other parameters, such as Flutter and Feedback.
- *Low Cut and High Cut sliders and fields*: Cut frequencies below the Low Cut value and above the High Cut value to shape the sound of taps (delay repeats) with the highpass and lowpass filters. The filters are located in the feedback circuit, which means that the filtering effect increases in intensity with each delay repeat. If you want an increasingly muddy and confused tone, move the High Cut slider toward the left. For ever thinner echoes, move the Low Cut slider toward the right. If you can't hear the effect, check the Dry and Wet controls *and* the filter settings.
- *LFO Rate knob and field*: Set the speed of the LFO.
- *LFO Intensity knob and field*: Set the amount of LFO modulation. A value of 0 turns off delay modulation.
- *Flutter Rate and Flutter Intensity knobs and fields*: Simulate the speed irregularities of tape transports used in analog tape delay units.
  - *Flutter Rate*: Set the speed variation.
  - *Flutter Intensity*: Set the intensity of the flutter effect.
- *Feedback knob*: Set the amount of delayed and filtered signal that is routed back to the input. Set Feedback to the lowest possible value to generate a single echo. Set Feedback to 100% to endlessly repeat the signal. The levels of the original signal and taps (echo repeats) tend to accumulate and may cause distortion. Use the Character controls to change the color of these overdriven signals.
- *Freeze button*: Capture current delay repeats and sustain them until Freeze is turned off.
- *Dry and Wet sliders and fields*: Independently control the amount of original and effect signal.

# Equalizers

## Intro to equalizers in Final Cut Pro

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

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

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

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

Multiband EQs (such as the Channel EQ, Fat EQ, or Linear Phase EQ) combine several filters in one unit, enabling you to control a large part of the frequency spectrum.

Multiband EQs allow you to independently set the frequency, bandwidth, and Q factor of each frequency spectrum band. This provides extensive, and precise, tone-shaping on any audio source, be it an individual audio signal or an overall mix.

Final Cut Pro includes a variety of single band and multiband EQs.

# AutoFilter

## Intro to AutoFilter in Final Cut Pro

AutoFilter is a versatile filter effect with several unique features. You can use it to create classic, analog-style synthesizer effects, or as a tool for creative sound design.

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

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

For information about adding the AutoFilter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

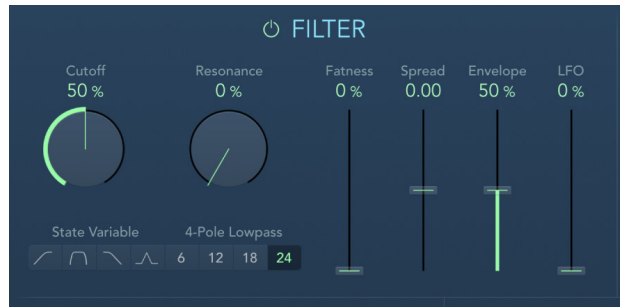


The AutoFilter window is divided into sections for Filter, Envelope, Distortion, LFO, and Out (output) controls. An extended controls section is also available.

- *Filter controls:* Control the tonal color of the filtered sound. See [AutoFilter Filter controls in Final Cut Pro](#).
- *Envelope controls:* Define how the filter cutoff frequency is modulated over time. See [AutoFilter Envelope controls in Final Cut Pro](#).
- *Distortion controls:* Distort the signal both before and after the filter. See [AutoFilter Distortion controls in Final Cut Pro](#).
- *LFO controls:* Define how the filter cutoff frequency is modulated by the LFO. See [AutoFilter LFO controls in Final Cut Pro](#).
- *Out (output) controls:* Set the level of both the dry and effect signal. See [AutoFilter output controls in Final Cut Pro](#).

## AutoFilter Filter controls in Final Cut Pro

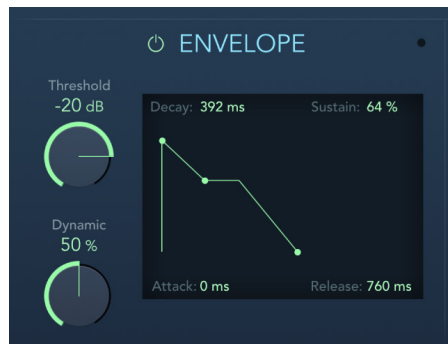
The AutoFilter effect's Filter controls precisely tailor the tonal color.



- *On/Off button:* Turn the Filter section on or off.
- *Cutoff knob and field:* Set the cutoff frequency for the filter. Higher frequencies are attenuated, whereas lower frequencies are allowed to pass through in a lowpass filter. The reverse is true in a highpass filter. When the State Variable filter is set to bandpass (BP) mode, the filter cutoff determines the center frequency of the frequency band that is allowed to pass.
- *Resonance knob and field:* Boost or cut signals in the frequency band that surrounds the cutoff frequency. Very high Resonance values cause the filter to begin oscillating at the cutoff frequency. This self-oscillation occurs before you reach the maximum Resonance value.
- *State Variable buttons:* Switch the filter between highpass (HP), bandpass (BP), lowpass (LP), and peak (PK) modes.
- *4-Pole Lowpass buttons:* Set the slope of the lowpass filter to 6, 12, 18, or 24 dB per octave.  
*Note:* Clicking one of these buttons automatically chooses the lowpass (LP) filter and slope, overriding any active State Variable filter button.
- *Fatness slider and field:* Boost the level of low-frequency content. When you set Fatness to its maximum value, adjusting Resonance has no effect on frequencies below the cutoff frequency. This parameter is used to compensate for a weak or "brittle" sound caused by high resonance values, when in the lowpass filter mode.
- *Spread slider and field:* Set the stereo spread width.
- *Envelope slider and field:* Determine the impact of the envelope on cutoff frequency.
- *LFO slider and field:* Determine the impact of the LFO on cutoff frequency.

## AutoFilter Envelope controls in Final Cut Pro

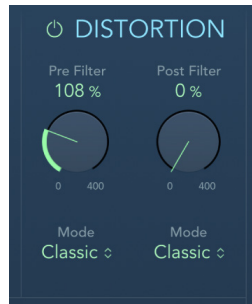
In the AutoFilter effect, the envelope is used to shape the filter cutoff over time. When the input signal exceeds the set threshold level, the envelope is triggered.



- *On/Off button:* Turn the Envelope section on or off.
- *Threshold knob and field:* Set an input level that—if exceeded—triggers the envelope or LFO that dynamically modulates filter cutoff frequency. See [AutoFilter LFO controls in Final Cut Pro](#) and [AutoFilter Filter controls in Final Cut Pro](#).  
*Note:* Retriggering of the envelope or LFO occurs only if Retrigger is set to On. See [AutoFilter extended controls in Final Cut Pro](#).
- *Dynamic knob and field:* Determine the input signal modulation amount. You can modulate the peak value of the envelope section by adjusting this control.
- *Attack handle and field:* Drag the handle horizontally (or drag in the field vertically) to set the envelope attack time.
- *Decay handle and field:* Drag the handle horizontally (or drag in the field vertically) to set the envelope decay time. This handle sets both the decay time and the sustain level.
- *Sustain handle and field:* Drag the handle vertically (or drag in the field vertically) to set the envelope sustain level. If the input signal falls below the threshold level before the envelope sustain phase, the release phase is triggered.
- *Release handle and field:* Drag the handle horizontally (or drag in the field vertically) to set the envelope release time. This is triggered as soon as the input signal falls below the threshold.

## AutoFilter Distortion controls in Final Cut Pro

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



- *On/Off button:* Turn the Distortion section on or off.
- *Pre Filter knob and field:* Set the amount of distortion applied before the filter section processes the signal.
- *Post Filter knob and field:* Set the amount of distortion applied after the filter section processes the signal.
- *Mode pop-up menus:* Choose the distortion type. The options are Classic, Tube, and Scream.

## AutoFilter LFO controls in Final Cut Pro

The AutoFilter effect's low-frequency oscillator (LFO) is used as a modulation source for filter cutoff.

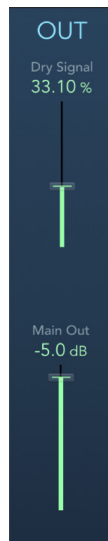


- *On/Off button:* Turn the LFO section on or off.
- *Sync button:* Synchronize the LFO with the clip tempo. You can set bar values, triplet values, and more with the Rate knob and field.
- *Rate knob and field:* Set the speed of LFO modulation. See [AutoFilter extended controls in Final Cut Pro](#).
- *Sync Phase knob and field:* When Sync is active, rotate to set the phase relationship between the LFO rate and the project tempo. This control is dimmed when Beat Sync is turned off.

- *Stereo Phase knob and field:* Set the phase relationship of the LFO modulations between the two channels (stereo only).
- *Retrigger button:* Turn on to retrigger the LFO waveform from the start of the cycle each time.
- *Waveform buttons:* Select the shape of the LFO waveform. Choose descending sawtooth, ascending sawtooth, triangle, pulse wave, or random.
- *Pulse Width knob and field:* Alter the curve shape of the selected waveform.

## AutoFilter output controls in Final Cut Pro

The AutoFilter effect's Out (output) controls are used to set the wet/dry balance and the overall level.



- *Dry Signal slider and field:* Set the amount of original, dry signal added to the filtered signal.
- *Main Out slider and field:* Set the overall output level. This compensates for higher levels caused by the use of distortion or by the filtering process itself.

## AutoFilter extended controls in Final Cut Pro

The extended controls provide enhanced control of AutoFilter functions. To display the extended controls in the Audio inspector, hold the pointer over AutoFilter in the Effects section, then click Show.

- *Rate Modulation slider and field:* Set the LFO frequency, independent of the input signal level. Typically, when the input signal exceeds the threshold, the modulation width of the LFO increases from 0 to the Rate Mod. value. This parameter allows you to override this behavior.
- *Retrigger button:* Start the LFO waveform at 0 each time the threshold is exceeded.
- *Region Gate On and Off buttons:* Turn on to keep the filter open for the entire region length. Turn off to use other AutoFilter parameters to control filter time.
- *LFO Decay/Delay slider and field:* Set the time required for the LFO to move from zero to the maximum value.



# Channel EQ

## Intro to Channel EQ in Final Cut Pro

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

You can use Channel EQ to shape the sound of an individual clip. The Analyzer and graphic controls make it easy to view and change the audio signal in real time.

## Using Channel EQ

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

You can reduce or eliminate unwanted frequencies, and you can raise quieter frequencies to make them more pronounced. You can adjust the center frequencies of bands 2 through 7 to affect a specific frequency—either one you want to emphasize, such as the root note of the music, or one you want to eliminate, such as hum or other noise. While doing so, change the Q parameter so that only a narrow range of frequencies are affected, or widen the range to alter a broad area.

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

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

To increase the resolution of the EQ curve display in the most interesting area around the zero line, drag the dB scale, on the left side of the graphic display, upward. Drag downward to decrease the resolution.

## Using the Channel EQ Analyzer

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

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

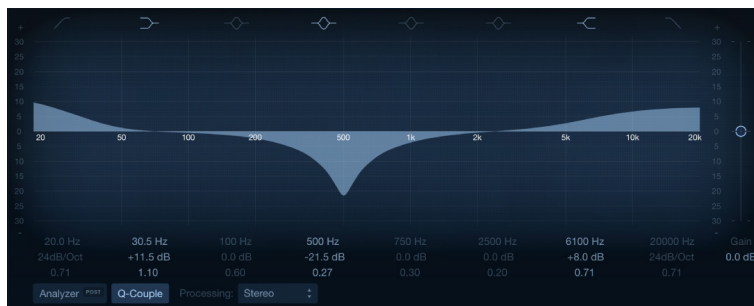
As soon as the Analyzer is activated, you can change the scaling with the Analyzer Top control, on the right side of the graphic display. The visible area represents a dynamic range of 60 dB. Drag vertically to set the maximum value to anywhere between +20 dB and -80 dB. The Analyzer display is always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

*Note:* When choosing a resolution, be aware that higher resolutions require significantly more processing power. High resolution is necessary when trying to obtain accurate analysis of very low bass frequencies, for example. It is recommended that you disable the Analyzer or close the Channel EQ window after setting the appropriate EQ controls. This frees up CPU resources for other tasks.

## Channel EQ controls in Final Cut Pro


The central area of the Channel EQ window includes the graphic display and controls for shaping each EQ band. The master Gain slider is shown at the right. The Analyzer and Q-Couple controls are shown at the lower left.

For information about adding the Channel EQ effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



## Channel EQ controls

- *Band 1 On/Off button*: A highpass filter that allows high frequencies to pass and reduces the level of low frequencies near the cutoff (set) frequency. When band 1 is active, you can directly change band parameters in the graphic display.
  - *Band 1 background*: Drag to change the frequency value.
  - *Band 1 dot*: Drag to change the frequency and Q values.
  - *Band 1 horizontal line*: Drag to change the Q value.
  - *Band 1 vertical line*: Drag to change the slope.
  - *Band 1 vertical and horizontal lines*: Drag to change the slope and Q value.
- *Band 2 On/Off button*: A low shelving filter that adjusts the level of low frequencies and has a minimal impact on frequencies above the cutoff (set) frequency. When band 2 is active, you can directly change band parameters in the graphic display.
  - *Band 2 background or dot*: Drag to change the frequency and gain values.
  - *Band 2 horizontal line*: Drag to change the gain value.
  - *Band 2 vertical line*: Drag to change the Q value.
  - *Band 2 vertical and horizontal lines*: Drag to change the gain and Q values.
- *Bands 3–6 On/Off buttons*: Parametric bell filters with three controls. Frequency sets a center frequency. Q sets the width of the frequency band around the center frequency. Gain sets the level of the band. When one of these bands is active, you can directly change band parameters in the graphic display.
  - *Band 3–6 background or dot*: Drag to change the frequency and gain values.
  - *Band 3–6 horizontal line*: Drag to change the gain value.
  - *Band 3–6 vertical line*: Drag to change the Q value.
  - *Band 3–6 vertical and horizontal lines*: Drag to change the gain and Q values.
- *Band 7 On/Off button*: A high shelving filter that adjusts the level of high frequencies and has a minimal impact on frequencies below the cutoff (set) frequency. When band 7 is active, you can directly change band parameters in the graphic display.
  - *Band 7 background or dot*: Drag to change the frequency and gain values.
  - *Band 7 horizontal line*: Drag to change the gain value.
  - *Band 7 vertical line*: Drag to change the Q value.
  - *Band 7 vertical and horizontal lines*: Drag to change the gain and Q values.
- *Band 8 On/Off button*: A lowpass filter that allows low frequencies to pass and reduces the level of high frequencies near the cutoff (set) frequency. When band 8 is active, you can directly change band parameters in the graphic display.
  - *Band 8 background*: Drag to change the frequency value.
  - *Band 8 dot*: Drag to change the frequency and Q value.
  - *Band 8 horizontal line*: Drag to change the Q value.
  - *Band 8 vertical line*: Drag to change the slope.
  - *Band 8 vertical and horizontal lines*: Drag to change the slope and Q value.

- *Graphic display:* Shows the current curve of each EQ band. The scale is shown in dB. The color of each band matches the corresponding button above the display. Each colored band (and corresponding Frequency, Gain, and Q field) is highlighted as you move the pointer across it. To select a band for editing, click a curve line segment, the (center frequency) handle, or in the colored space between the zero line and EQ curve.
-  **Tip:** Hold down Command while performing any of the operations below to limit dragging to vertical or horizontal movement.
  - Drag anywhere in the colored band to adjust gain and the center frequency.
  - Drag the vertical lines that encompass the band to adjust the Q (bandwidth) only.
  - Drag the horizontal line in the band to adjust the gain only. If Q-Coupling is enabled, both the gain and the bandwidth are adjusted.
  - Drag the intersection of vertical and horizontal lines to adjust the gain and Q simultaneously.
  - Drag the handle to adjust the center frequency only.

*Note:* Horizontally dragging the frequency handle in band 1 and band 8 adjusts both the frequency and Q.

  - Hold down Option-Command to adjust the Q and center frequency simultaneously. A "Q" icon is shown at the pointer position.
- *Frequency field:* Adjust the frequency of the selected band.
- *Gain/Slope field:* Set the amount of gain for the selected band. For bands 1 and 8, this changes the slope of the filter.
- *Q field:* Adjust the Q factor or resonance of the affected range around the center frequency in the selected band.
 

*Note:* The Q parameter of band 1 and band 8 has no effect when the slope is set to 6 dB/Oct. When the Q parameter is set to an extremely high value, such as 100, these filters affect only a very narrow frequency band and can be used as notch filters.
- *Master Gain slider and field:* Set the overall output level of the signal. Use it after boosting or cutting individual frequency bands.
- *Analyzer button:* Turn the Analyzer on or off.
- *Analyzer Pre/Post button:* Set to display the frequency curve before or after EQ is applied, when the Analyzer is on. See [Intro to Channel EQ in Final Cut Pro](#).
- *Q-Couple button:* Turn on Gain-Q coupling, which automatically adjusts the Q (bandwidth) when you raise or lower the gain on any EQ band. This preserves the perceived bandwidth of the bell curve. See [Channel EQ extended controls](#), below.
- *Processing pop-up menu:* Choose which channel or channels to process (Stereo, Left Only, Right Only, Mid Only, or Side Only).

## Channel EQ extended controls

To display the extended controls in the Audio inspector, hold the pointer over Channel EQ in the Effects section, then click Show.

- *Analyzer Resolution pop-up menu:* Choose the sample resolution for the Analyzer. Choose "Resolution low (2048 points)," "Resolution medium (4096 points)," or "Resolution high (8192 points)."
- *Analyzer Mode buttons:* Choose Peak or RMS.
- *Analyzer Decay slider and field:* Set the decay rate (in dB per second) of the Analyzer curve. This is shown as a peak decay in Peak mode or an averaged decay in RMS mode.
- *Gain-Q Couple Strength pop-up menu:* Choose the amount of Gain-Q coupling.
  - *Strong:* Preserves most of the perceived bandwidth.
  - *Light or medium:* Allows some change as you raise or lower the gain.
  - *Asymmetric:* These settings feature a stronger coupling for negative gain values than for positive values, so the perceived bandwidth is more closely preserved when you cut, rather than boost, gain.

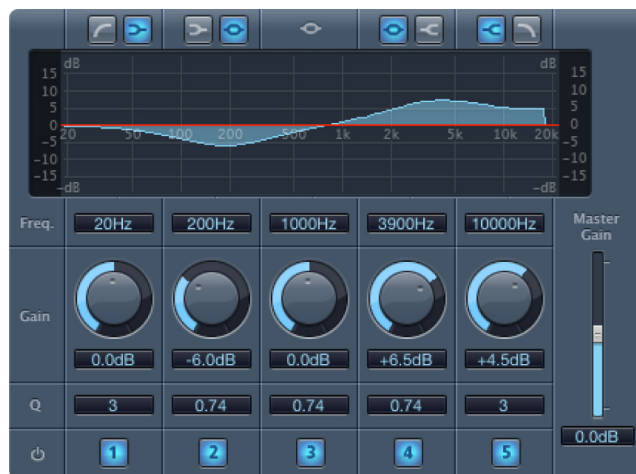
*Note:* If you play back automation of the Q parameter with a different Gain-Q Couple Strength setting, the actual Q values are different than when the automation was recorded.

- *Oversampling checkbox:* Select to double the EQ sampling rate. This results in enhanced precision, with a corresponding increase in CPU overhead.

## Fat EQ controls in Final Cut Pro

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

For information about adding the Fat EQ effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Band Type buttons*: Located above the graphic display. For bands 1–2 and 4–5, click one of the paired buttons to select the EQ type for the corresponding band.
  - *Band 1*: Click the highpass or low shelving button.
  - *Band 2*: Click the low shelving or parametric button.
  - *Band 3*: Always acts as a parametric EQ band.
  - *Band 4*: Click the parametric or high shelving button.
  - *Band 5*: Click the high shelving or lowpass button.
- *Graphic display*: Shows the EQ curve of each frequency band.
- *Frequency fields*: Set the frequency for each band.
- *Gain knobs*: Set the amount of gain for each band.
- *Q fields*: Set the Q or bandwidth of each band—the range of frequencies around the center frequency that are altered. At low Q factor values, the EQ covers a wider frequency range. At high Q values, the effect of the EQ band is limited to a narrow frequency range. The Q value can significantly influence how audible your changes are—if you’re working with a narrow frequency band, you generally need to cut or boost more drastically to notice the difference.
 

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

## Linear Phase EQ

### Intro to Linear Phase EQ in Final Cut Pro

The high-quality Linear Phase EQ effect is similar to the Channel EQ effect, sharing the same controls and eight-band layout. Linear Phase EQ uses a different underlying technology, however, that perfectly preserves the phase of the audio signal. This phase coherency is assured even when you apply the wildest EQ curves to the sharpest signal transients.

A further difference is that Linear Phase EQ uses a fixed amount of CPU resources, regardless of how many bands are active. Linear Phase EQ also introduces greater amounts of latency. Therefore, it is strongly recommended that you use it for mastering previously recorded audio.

### Using Linear Phase EQ

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

You can reduce or eliminate unwanted frequencies, and you can raise quieter frequencies to make them more pronounced. You can adjust the center frequencies of bands 2 through 7 to affect a specific frequency—either one you want to emphasize, such as the root note of the music, or one you want to eliminate, such as hum or other noise. While doing so, change the Q parameter so that only a narrow range of frequencies are affected, or widen the range to alter a broad area.

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

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

To increase the resolution of the EQ curve display in the most interesting area around the zero line, drag the dB scale, on the left side of the graphic display, upward. Drag downward to decrease the resolution.

## Using the Linear Phase EQ Analyzer

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

The bands derived from FFT analysis are divided in accordance with the frequency linear principle—there are more bands in higher octaves than in lower ones.

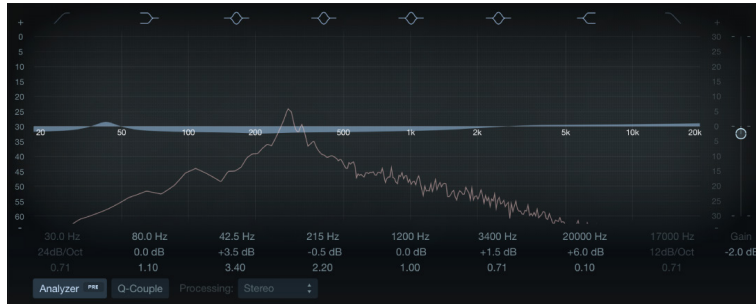
As soon as the Analyzer is activated, you can change the scaling with the Analyzer Top control, on the right side of the graphic display. The visible area represents a dynamic range of 60 dB. Drag vertically to set the maximum value to anywhere between +20 dB and -40 dB. The Analyzer display is always dB-linear.

*Note:* When choosing a resolution, be aware that higher resolutions require significantly more processing power. High resolution is necessary when trying to obtain accurate analysis of very low bass frequencies, for example. It's recommended that you disable the Analyzer or close the Linear Phase EQ window after setting the appropriate EQ parameters. This frees up CPU resources for other tasks.

## Linear Phase EQ controls in Final Cut Pro

The central area of the Linear Phase EQ window includes the graphic display and controls for shaping each EQ band. The master Gain slider is shown at the right. The Analyzer and Q-Couple controls are shown at the lower left.

For information about adding the Linear Phase EQ effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).




### Linear Phase EQ controls

- **Band 1 On/Off button:** A highpass filter that allows high frequencies to pass and reduces the level of low frequencies near the cutoff (set) frequency. When band 1 is active, you can directly change band parameters in the graphic display.
  - **Band 1 background:** Drag to change the frequency value.
  - **Band 1 dot:** Drag to change the frequency and Q values.
  - **Band 1 horizontal line:** Drag to change the Q value.
  - **Band 1 vertical line:** Drag to change the slope.
  - **Band 1 vertical and horizontal lines:** Drag to change the slope and Q value.
- **Band 2 On/Off button:** A low shelving filter that adjusts the level of low frequencies and has a minimal impact on frequencies above the cutoff (set) frequency. When band 2 is active, you can directly change band parameters in the graphic display.
  - **Band 2 background or dot:** Drag to change the frequency and gain values.
  - **Band 2 horizontal line:** Drag to change the gain value.
  - **Band 2 vertical line:** Drag to change the Q value.
  - **Band 2 vertical and horizontal lines:** Drag to change the gain and Q values.
- **Bands 3–6 On/Off buttons:** Parametric bell filters with three controls. Frequency sets a center frequency. Q sets the width of the frequency band around the center frequency. Gain sets the level of the band. When a band is active, you can directly change band parameters in the graphic display.
  - **Band 3–6 background or dot:** Drag to change the frequency and gain values.
  - **Band 3–6 horizontal line:** Drag to change the gain value.
  - **Band 3–6 vertical line:** Drag to change the Q value.
  - **Band 3–6 vertical and horizontal lines:** Drag to change the gain and Q values.



- *Band 7 On/Off button*: A high shelving filter that adjusts the level of high frequencies and has a minimal impact on frequencies below the cutoff (set) frequency. When band 7 is active, you can directly change band parameters in the graphic display.
    - *Band 7 background or dot*: Drag to change the frequency and gain values.
    - *Band 7 horizontal line*: Drag to change the gain value.
    - *Band 7 vertical line*: Drag to change the Q value.
    - *Band 7 vertical and horizontal lines*: Drag to change the gain and Q values.
  - *Band 8 On/Off button*: A lowpass filter that allows low frequencies to pass and reduces the level of high frequencies near the cutoff (set) frequency. When band 8 is active, you can directly change band parameters in the graphic display.
    - *Band 8 background*: Drag to change the frequency value.
    - *Band 8 dot*: Drag to change the frequency and Q value.
    - *Band 8 horizontal line*: Drag to change the Q value.
    - *Band 8 vertical line*: Drag to change the slope.
    - *Band 8 vertical and horizontal lines*: Drag to change the slope and Q value.
  - *Graphic display*: Shows the current curve of each EQ band. The scale is shown in dB. The color of each band matches the corresponding button above the display. Each colored band (and corresponding Frequency, Gain, and Q field) is highlighted as you move the pointer across it. To select a band for editing, click a curve line segment, the (center frequency) handle, or in the colored space between the zero line and EQ curve.
 

 **Tip**: Hold down Command while performing any of the operations below to limit dragging to vertical or horizontal movement.

    - Drag anywhere in the colored band to adjust gain and the center frequency.
    - Drag the vertical lines that encompass the band to adjust the Q (bandwidth) only.
    - Drag the horizontal line in the band to adjust the gain only. If Q-Coupling is enabled, both the gain and bandwidth are adjusted.
    - Drag the intersection of vertical and horizontal lines to adjust the gain and Q simultaneously.
    - Drag the handle to adjust the center frequency only.

*Note*: Horizontally dragging the frequency handle in band 1 and band 8 adjusts both the frequency and Q.

    - Hold down Option-Command to adjust the Q and center frequency simultaneously. A "Q" icon is shown at the pointer position.
  - *Frequency field*: Adjust the frequency of the selected band.
  - *Gain/Slope field*: Set the amount of gain for the selected band. For bands 1 and 8, this changes the slope of the filter.
  - *Q field*: Adjust the Q factor or resonance of the affected range around the center frequency in the selected band.
- Note*: The Q parameter of band 1 and band 8 has no effect when the slope is set to 6 dB/Oct. When the Q parameter is set to an extremely high value, such as 100, these filters affect only a very narrow frequency band and can be used as notch filters.

- *Master Gain slider and field:* Set the overall output level of the signal. Use it after boosting or cutting individual frequency bands.
- *Analyzer button:* Turn the Analyzer on or off.
- *Analyzer Pre/Post button:* Set to display the frequency curve before or after EQ is applied, when the Analyzer is on. See [Intro to Linear Phase EQ in Final Cut Pro](#).
- *Q-Couple button:* Turn on Gain-Q coupling, which automatically adjusts the Q (bandwidth) when you raise or lower the gain on any EQ band. This preserves the perceived bandwidth of the bell curve. See [Linear Phase EQ extended controls](#), below.
- *Processing pop-up menu:* Choose to process both sides of a stereo signal, the Left Only, Right Only, Mid Only, or Side Only signal.

## Linear Phase EQ extended controls

To display the extended controls in the Audio inspector, hold the pointer over Linear Phase EQ in the Effects section, then click Show.

- *Analyzer Resolution pop-up menu:* Choose the sample resolution for the Analyzer. Choose "Resolution low (2048 points)," "Resolution medium (4096 points)," or "Resolution high (8192 points)."
- *Analyzer Mode buttons:* Choose Peak or RMS.
- *Analyzer Decay slider and field:* Set the decay rate (in dB per second) of the Analyzer curve. These are shown as a peak decay in Peak mode or an averaged decay in RMS mode.
- *Gain-Q Couple Strength pop-up menu:* Choose the amount of Gain-Q coupling.
  - *Strong:* Preserves most of the perceived bandwidth.
  - *Light or medium:* Allows some change as you raise or lower the gain.
  - *Asymmetric:* These settings feature a stronger coupling for negative gain values than for positive values, so the perceived bandwidth is more closely preserved when you cut, rather than boost, gain.

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

# Levels effects

## Intro to levels effects in Final Cut Pro

The levels effects control the perceived loudness of your audio, add focus and punch to clips, and optimize the sound for playback in different situations.

The *dynamic range* of an audio signal is the range between the softest and loudest parts of the signal—technically, between the lowest and highest amplitudes. Dynamics processors enable you to adjust the dynamic range of individual audio clips. This can be to increase the perceived loudness or to highlight the most important sounds, while ensuring that softer sounds are not lost in the mix.

There are four types of dynamics processors included in Final Cut Pro, each used for different audio processing tasks.

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

By reducing the highest parts of the signal, called *peaks*, a compressor raises the overall level of the signal, increasing the perceived volume. This gives the signal more focus by making the louder (foreground) parts stand out, while keeping the softer background parts from becoming inaudible. Compression also tends to make sounds tighter or punchier because transients are emphasized, depending on attack and release settings, and because the maximum volume is reached more swiftly.

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

Compressors are typically used on dialogue clips to make the speech more intelligible in an overall mix. They are also commonly used on music and sound effects clips, but they are rarely used on ambience clips.

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

- **Expanders:** Expanders are similar to compressors, except that they raise, rather than lower, the signal when it exceeds the threshold. Expanders are used to add life to audio signals.

- *Limiters*: Limiters—also called *peak limiters*—work in a similar way to compressors in that they reduce the audio signal when it exceeds a set threshold. The difference is that whereas a compressor gradually lowers signal levels that exceed the threshold, a limiter quickly reduces any signal louder than the threshold to the threshold level. The main use of a limiter is to prevent clipping while preserving the maximum overall signal level.
- *Noise gates*: Noise gates alter the signal in a way that is opposite to that used by compressors or limiters. Whereas a compressor lowers the level when the signal is louder than the threshold, a noise gate lowers the signal level whenever it falls below the threshold. Louder sounds pass through unchanged, but softer sounds, such as ambient noise or the decay of a sustained instrument, are cut off. Noise gates are often used to eliminate low-level noise or hum from an audio signal.

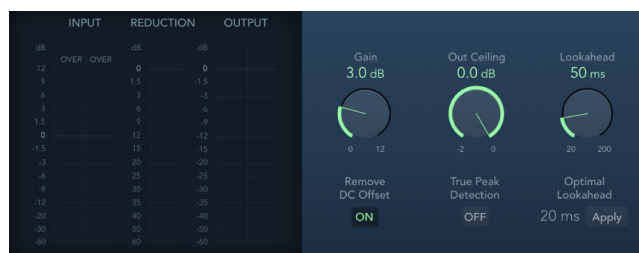
## Adaptive Limiter controls in Final Cut Pro

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

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

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

For information about adding the Adaptive Limiter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Input meters*: Show input levels in real time. The Margin field shows the peak input level. You can reset the Margin field by clicking it.
- *Reduction meter*: Shows the amount of gain reduction. The Margin field shows the peak reduction level. You can reset the Margin field by clicking it.
- *Output meters*: Show output levels of the limited signal. The Margin field shows the peak output level. You can reset the Margin field by clicking it.

- *Gain knob and field*: Set the amount of gain after input scaling.
- *Out Ceiling knob and field*: Set the maximum output level, or ceiling. The signal does not rise above this.
- *Lookahead knob and field*: Set the playback buffer size (how far in the future the file is analyzed for peaks). Also see the description of the Optimal Lookahead control, below. Values lower than the optimal buffer size are indicated in red.
- *Remove DC Offset button*: Turn on to activate a highpass filter that removes direct current (DC) from the signal. DC can be introduced by lower-quality audio hardware.
- *True Peak Detection button*: Turn on to detect inter-sample peaks in the signal.
- *Optimal Lookahead field and button*: Use the Apply button to set the optimal playback buffer size. This changes the value shown for the Lookahead parameter.

*Note*: The Optimal Lookahead value is automatically determined and cannot be changed directly.

## Compressor

### Intro to the Compressor effect in Final Cut Pro

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

You can use Compressor with individual clips, including voice, instrumental, and effects clips.

### Setting the Compressor threshold and ratio

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

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

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

## Setting suitable Compressor envelope times

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

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

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

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

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

## Setting the Compressor knee

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

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

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

## Setting other Compressor parameters

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

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

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

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

# Compressor effect controls in Final Cut Pro

Compressor offers the following controls.

For information about adding the Compressor effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



## Main Compressor controls

- **Circuit Type buttons:** Select the type of circuit emulated by Compressor. The interface is updated when you select a circuit type. The choices are Platinum Digital, Studio VCA, Studio FET, Classic VCA, Vintage VCA, Vintage FET, and Vintage Opto.
  - FET (Field Effect Transistor) compressors are known for their fast transient response. They can deliver a clean or colored tone (particularly in the midrange), and can be pushed to a somewhat “crunchy” tone on transients. FET compressors are ideal for drums, vocals, guitars, and other signals with a fast attack phase. FET compressors can only attenuate the signal.
  - VCA (Voltage Controlled Amplifier) compressors can respond slowly or quickly to incoming transients. They tend to produce a clean tone and are well-suited for bass guitars and other low-frequency signals. VCA compressors can attenuate or amplify the signal.
  - Opto (Optical) compressors are known for their fast transient response and nonlinear release handling. They are very clean and are ideal for vocals and guitars. They are also often used as limiting amplifiers across busses or outputs.
- **Side Chain and Output buttons:** View Side Chain or Output parameters.
- **Gain Reduction meter/graph:** Click either the Meter or Graph button to change the real-time compression amount display.

- *Input Gain knob and field:* Set the level at the compressor input.
- *Input Gain meter:* Displays the level at the compressor input.
- *Threshold knob and field:* Set the threshold level—signals above this threshold value are reduced in level.
- *Ratio knob and field:* Set the compression ratio—the ratio of signal reduction when the threshold is exceeded.
- *Make Up knob and field:* Set the amount of gain applied to the compressed signal.
- *Auto Gain buttons:* The Off button disables autogain. The 0 dB and –12 dB buttons compensate for volume reductions caused by compression.
- *Knee knob and field:* Set the strength of compression at levels close to the threshold. Lower values result in more severe or immediate compression (hard knee). Higher values result in gentler compression (soft knee).
- *Attack knob and field:* Set the time it takes for Compressor to react when the signal exceeds the threshold.
- *Release knob and field:* Set the time it takes for Compressor to stop reducing the signal after the signal level falls below the threshold. This control works in conjunction with the Auto button when Auto is turned on.
- *Auto button:* Make the release time dynamically adjust to the audio material. The automatic release time adjustment and compression results change when different Release parameter values are used.
- *Output Gain knob and field:* Set the overall level of the compressor output.
- *Output Gain meter:* Displays the overall level of the compressor output.

## Output controls

- *Limiter button:* Turn the integrated limiter on or off. Limiting prevents the Compressor output from exceeding the threshold value.
- *Limiter Threshold knob and field:* Set the threshold level for the limiter.
- *Lookahead button:* Turn on to pre-buffer incoming audio, enabling perfect compression of very fast transients.
- *Distortion knob:* Choose whether to apply clipping above 0 dB, and the type of clipping. Soft, Hard, and Clip reduce the signal around the 0 dB line in different ways, resulting in a smoothed or squared-off distortion of the signal peaks.
- *Mix knob and field:* Set the balance between dry (source) and wet (effect) signals. This enables you to either reduce signal peaks (dry) or increase the level of softer signals (wet).



## Side Chain controls

- *Detection buttons:* Choose the way stereo side-chain signals are handled at the side-chain input.
  - *Max button:* Turn on to compress both channels if either stereo channel exceeds or falls below the threshold.
  - *Sum button:* When Sum is turned on, the combined level of both channels must exceed the threshold before compression occurs.
  - *Peak and RMS buttons:* Use in conjunction with the Max and Sum buttons. Click Peak or RMS to determine whether signal peaks or a signal average is used for detection. These can help avoid artifacts such as clicks in the processed signal, depending on the type of audio material and parameter settings (notably Attack).
- *Filter buttons:* Turn the filter on or off. Turn on Listen to monitor the side-chain signal.
- *Filter mode knob:* Choose the type of filter used to process the incoming side-chain signal. Filtering the side-chain input signal can enhance the precision of trigger signals, resulting in more surgical compression. The choices are LP (lowpass), BP (bandpass), HP (highpass), ParEQ (parametric), and HS (high shelving).
- *Frequency knob and field:* Set the center frequency for the side-chain filter.
- *Q knob and field:* Set the width of the frequency band affected by the side-chain filter.
- *Gain knob and field:* Set the amount of gain applied to the side-chain signal.

## Enveloper

### Intro to Enveloper in Final Cut Pro

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

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

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

Boosting the release phase also accentuates any reverb applied to the affected clip. Conversely, attenuating the release phase makes audio originally drenched in reverb sound drier. This is particularly useful when working with drum loops, but it has many other applications as well. Let your imagination be your guide.

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

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

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

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

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

## Enveloper controls in Final Cut Pro

Enveloper offers the following controls.

For information about adding the Enveloper effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



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

# Expander controls in Final Cut Pro

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

For information about adding the Expander effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



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

*Note:* As Expander is a genuine upward expander—in contrast to a downward expander, which increases the dynamic range below the threshold—the Ratio slider features a value range of 1:1 to 0.5:1.

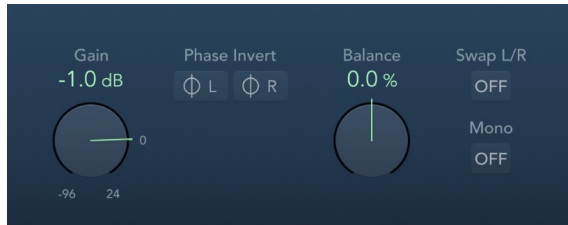
- *Knee slider and field:* Determine the strength of expansion at levels close to the threshold. Lower values result in more severe or immediate expansion (hard knee). Higher values result in a gentler expansion (soft knee).
- *Gain slider and field:* Set the amount of output gain.
- *Auto Gain button:* Compensate for the level increase caused by expansion. When Auto Gain is on, the signal sounds softer, even when the peak level remains the same.

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

# Gain controls in Final Cut Pro

Gain amplifies (or reduces) the signal by a specific decibel amount.

For information about adding the Gain effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Gain knob and field:* Set the amount of gain.
- *Phase Invert Left and Right buttons:* Invert the phase of the left and right channels, respectively.

Inverting phase is useful for dealing with time alignment problems, particularly those caused by simultaneous recording with multiple microphones. When you invert the phase of a signal heard in isolation, it sounds identical to the original. When the signal is heard in conjunction with other signals, however, phase inversion may have an audible effect. For example, if you place microphones above and below a snare drum, inverting the phase of either microphone can improve (or ruin) the sound. As always, rely on your ears.
- *Balance knob and field:* Adjust the balance of the incoming signal between the left and right channels.
- *Swap L/R button:* Swap the left and right output channels. Swapping occurs after the Balance parameter in the signal path. The Swap L/R button is disabled when Mono is turned on.
- *Mono button:* Output the summed mono signal on both the left and right channels.

*Note:* The Gain effect is available in mono, mono-to-stereo, and stereo instances. Only one Phase Invert button is available in mono and mono-to-stereo modes. In mono mode, the Balance, Swap L/R, and Mono parameters are also disabled. A separate [Multichannel Gain](#) effect is also available in surround channels. This features per-channel Phase Invert and Mute buttons, and Level sliders for each channel.

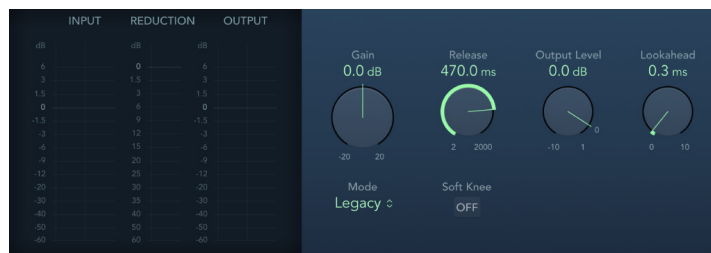
# Limiter controls in Final Cut Pro

Limiter works much like a compressor but with one important difference: Whereas a compressor proportionally reduces the signal when it exceeds the threshold, a limiter reduces any peak above the threshold to the threshold level, effectively limiting the signal to this level.

Limiter is used primarily when mastering. Typically, you apply Limiter as the very last process in the mastering signal chain, where it raises the overall volume of the signal so that it reaches, but does not exceed, 0 dB.

Limiter is designed in such a way that if set to 0 dB Gain and 0 dB Output Level, it has no effect on a normalized signal. If the signal clips, Limiter reduces the level before clipping can occur. Limiter cannot, however, fix audio that is clipped during recording.

For information about adding the Limiter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Input meters:* Show input levels in real time. The Margin field shows the highest input level. Click the Margin field to reset it.
- *Reduction meter:* Shows the amount of limiting in real time.
- *Output meters:* Show output levels of the limited signal. The Margin field shows the highest output level. Click the Margin field to reset it.
- *Gain knob and field:* Set the amount of gain applied to the input signal.
- *Release knob and field:* Set the time it takes for Limiter to stop processing, after the signal falls below the threshold level.
- *Output Level knob and field:* Set the output level of the signal.
- *Lookahead knob and field:* Adjust how far ahead (in milliseconds) Limiter analyzes the audio signal. This enables it to react earlier to peak volumes by adjusting the amount of reduction.

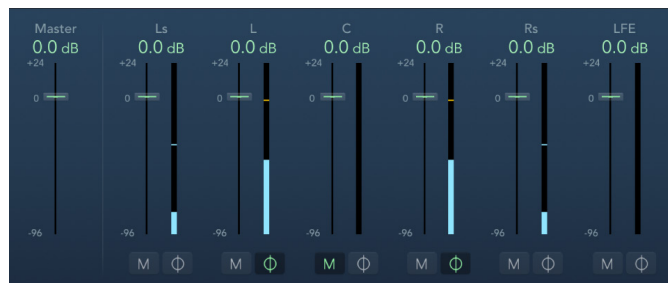
*Note:* Lookahead causes latency, but this has no perceptible effect when you use Limiter as a mastering effect on prerecorded material. Set it to higher values if you want the limiting effect to occur before the maximum level is reached, thus creating a smoother transition.

- *Mode pop-up menu*: Choose between Legacy and Precision algorithms. Use Precision for hard limiting, but be aware that this can introduce distortion artifacts.
- *Soft Knee button (Legacy mode)*: Turn on to limit the signal only when it reaches the threshold. The transition to full limiting is nonlinear, producing a softer, less abrupt effect, and reducing distortion artifacts that can be produced by hard limiting (in Precision mode).
- *True Peak Detection button (Precision mode)*: Turn on to detect intersample peaks in the signal.

## Multichannel Gain controls in Final Cut Pro

Multichannel Gain allows you to independently control the gain and phase of each channel in a surround mix.

For information about adding the Multichannel Gain effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Master slider and field*: Set the master gain for the combined channel output.
- *Channel gain sliders and fields*: Set the amount of gain for the respective channel.
- *Mute buttons*: Mute the selected channel.
- *Phase Invert buttons*: Invert the phase of the selected channel.

## Multipressor

### Intro to Multipressor in Final Cut Pro

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

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

Multipressor allows the use of higher compression ratios on specific frequency bands, so it can achieve a higher average volume without causing audible artifacts.

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

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

## Using Multipressor

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

## Setting compression parameters

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

## Setting downward expansion parameters

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

## Setting Peak/RMS and Envelope parameters

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

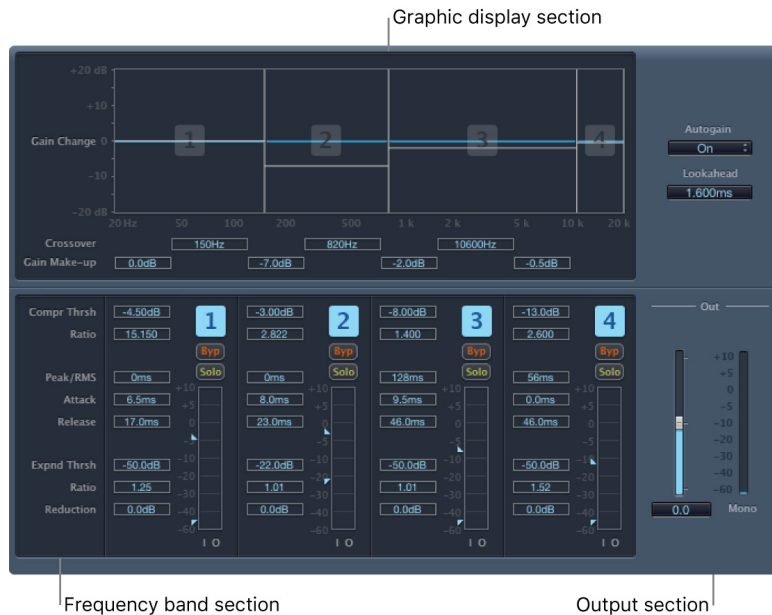
## Setting output parameters

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

# Multipressor controls in Final Cut Pro

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

For information about adding the Multipressor effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



## Multipressor graphic display section

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

## Multipressor frequency band section

- *Compr Thrsh (Compression Threshold) fields:* Set the compression threshold for the selected band. Setting the parameter to 0 dB results in no compression of the band.
- *Compr (Compression) Ratio fields:* Set the compression ratio for the selected band. Setting the parameter to 1:1 results in no compression of the band.
- *Exprd Thrsh (Expand Threshold) fields:* Set the expansion threshold for the selected band. Setting the parameter to its minimum value (-60 dB) means that only signals that fall below this level are expanded.
- *Exprd (Expand) Ratio fields:* Set the expansion ratio for the selected band.



- *Expnd (Expand) Reduction fields:* Set the amount of downward expansion for the selected band.
- *Peak/RMS fields:* Enter a smaller value for shorter peak detection, or a larger value for RMS detection, in milliseconds.
- *Attack fields:* Set the amount of time before compression starts for the selected band, after the signal exceeds the threshold.
- *Release fields:* Set the time required before compression stops on the selected band, after the signal falls below the threshold.
- *Band On/Off buttons (1, 2, 3, and 4):* Enable or disable each band (1 to 4). When enabled, the button is highlighted, and the corresponding band appears in the graphic display area above.
- *Byp (Bypass) buttons:* Enable to bypass the selected frequency band.
- *Solo buttons:* Enable to hear compression on only the selected frequency band.
- *Level meters:* The bar on the left shows the input level, and the bar on the right shows the output level.
- *Threshold arrows:* Two arrows appear to the left of each Level meter.
  - The upper arrow adjusts the Compression Threshold (Compr Thrsh).
  - The lower arrow adjusts the Expansion Threshold (Expnd Thrsh).

### Multipressor output section

- *Auto Gain pop-up menu:* When you choose On, it references the overall processing of the signal to 0 dB, making the output louder.
- *Lookahead value field:* Adjusts how far the effect looks forward in the incoming audio signal, in order to react earlier to peak volumes, and therefore achieve smoother transitions.
- *Out slider:* Sets the overall gain at the Multipressor output.
- *Level meter:* Shows the overall output level.

## Noise Gate

### Intro to Noise Gate in Final Cut Pro

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

Noise Gate works by allowing signals above the threshold level to pass unimpeded, while reducing signals below the threshold level. This effectively removes lower-level parts of the signal, while allowing the desired parts of the audio to pass.

## Use Noise Gate in Final Cut Pro

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

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

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

The Hysteresis slider provides another option for preventing chattering, without needing to define a minimum Hold time. Use it to set the range between the threshold values that open and close Noise Gate. This is useful when the signal level hovers around the Threshold level, causing Noise Gate to switch on and off repeatedly, producing the undesirable chattering effect. The Hysteresis slider essentially sets Noise Gate to open at the Threshold level and remain open until the level drops below another, lower, level. As long as the difference between these two values is large enough to accommodate the fluctuating level of the incoming signal, Noise Gate can function without creating chatter. This value is always negative. Generally, -6 dB is a good place to start.

In some situations, you may find that the level of the signal you want to keep and the level of the noise signal are close, making it difficult to separate them. For example, when you are recording a drum kit and using Noise Gate to isolate the sound of the snare drum, the hi-hat may also open the gate in many cases. To remedy this, use the side-chain controls to isolate the desired trigger signal with the High Cut and Low Cut filters.

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

### Use the side-chain filters

1. In the Final Cut Pro timeline, select a clip with the Noise Gate effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Click the Monitor button to hear how the High Cut and Low Cut filters will affect the incoming trigger signal.

3. Drag the High Cut slider to set the upper frequency.

Trigger signals above this are filtered.

4. Drag the Low Cut slider to set the lower frequency.

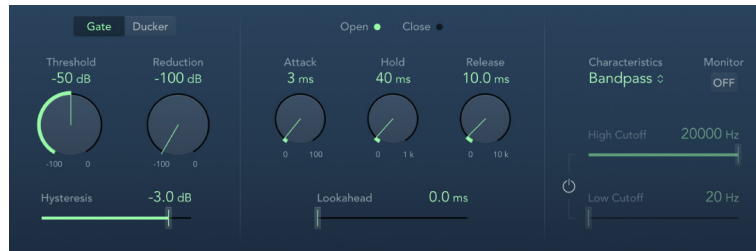
Trigger signals below this are filtered.

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

## Noise Gate controls in Final Cut Pro

Noise Gate has the following controls.

For information about adding the Noise Gate effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Gate and Ducker buttons:* Set the operating mode. See [Use Noise Gate in Final Cut Pro](#).
  - *Threshold knob and field:* Set the threshold level. Signals that fall below the threshold are reduced in level.
  - *Reduction/Ducking knob and field:* Set the amount of signal reduction.
  - *Hysteresis slider and field:* Set the difference (in decibels) between the threshold values that open and close the gate. This prevents the gate from rapidly opening and closing when the input signal level is close to the threshold level.
  - *Open and Close indicators:* Show current gate state.
  - *Attack knob and field:* Set the time it takes to fully open the gate after the signal exceeds the threshold.
  - *Hold knob and field:* Set the time the gate remains open after the signal falls below the threshold.
  - *Release knob and field:* Set the time it takes to reach maximum attenuation after the signal falls below the threshold.
  - *Lookahead slider and field:* Control how far ahead Noise Gate analyzes the incoming signal, allowing the effect to respond more quickly to peak levels.
  - *Characteristics pop-up menu:* Choose a Bandpass or Band Reject filter type.
  - *Monitor button:* Turn on to hear the side-chain signal, including the effect of the High Cut and Low Cut filters (if enabled).
  - *Filter button:* Turn on to adjust the High Cutoff and Low Cutoff parameters.
  - *High Cutoff slider and field:* Set the upper cutoff frequency for the side-chain signal.
  - *Low Cutoff slider and field:* Set the lower cutoff frequency for the side-chain signal.
- Note:* When no external side chain is selected, the input signal is used as the side chain.

# Spectral Gate

## Intro to Spectral Gate in Final Cut Pro

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

It works by dividing the incoming signal into two frequency ranges—above and below a central frequency band that you specify with the Center Freq. and Bandwidth controls. The signal ranges above and below the defined band can be individually processed with the Low Level and High Level controls and the Super Energy and Sub Energy controls. See [Use Spectral Gate in Final Cut Pro](#).

## Use Spectral Gate in Final Cut Pro

One way to familiarize yourself with the operation of Spectral Gate would be to start with a drum loop. Set Center Freq. to its minimum value (20 Hz) and Bandwidth to its maximum value (20,000 Hz) so that the entire frequency range is processed. Turn up Super Energy and Sub Energy, one at a time, then try different Threshold settings. This should give you a good sense of how different Threshold levels affect the sound of Super Energy and Sub Energy. When you come across a sound that you like or consider useful, narrow Bandwidth drastically, gradually increase Center Freq., and then use the Low Level and High Level sliders to mix in some treble and bass from the original signal. At lower Speed settings, turn up CF Modulation or BW Modulation.

### Acquaint yourself with Spectral Gate

1. In the Final Cut Pro timeline, select a clip with the Spectral Gate effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Set the frequency band you want to process by using the Center Freq. and Bandwidth controls.

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

3. After the frequency band is defined, use the Threshold slider to set the appropriate level.

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

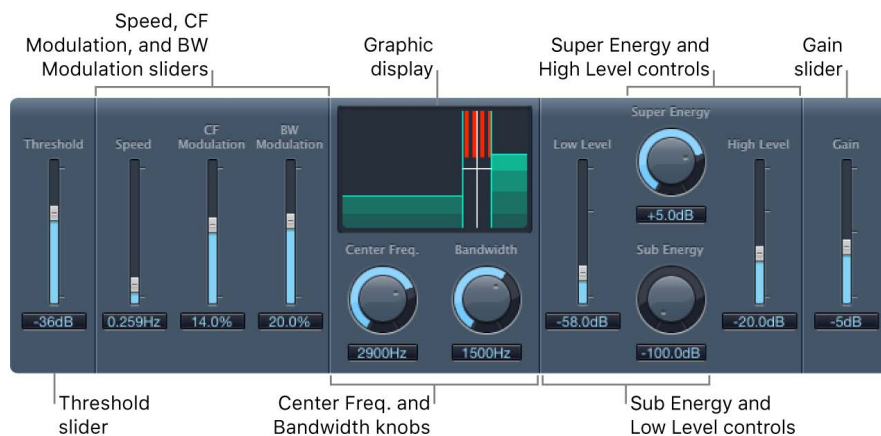
4. Use the Super Energy knob to control the level of the frequencies above the Threshold, and use the Sub Energy knob to control the level of the frequencies below the Threshold.
5. To mix the frequencies that fall outside the frequency band (defined by the Center Freq. and Bandwidth parameters) with the processed signal, do any of the following:
  - Use the Low Level slider to blend the frequencies below the defined frequency band with the processed signal.
  - Use the High Level slider to blend frequencies above the defined frequency band with the processed signal.

6. To modulate the defined frequency band, use the Speed, CF Modulation, and BW Modulation sliders.
  - Speed determines the modulation frequency.
  - CF Modulation defines the intensity of the center frequency modulation.
  - BW Modulation controls the amount of bandwidth modulation.
7. After making your adjustments, use the Gain slider to adjust the final output level of the processed signal.

## Spectral Gate controls in Final Cut Pro

Spectral Gate includes the following controls.

For information about adding the Spectral Gate effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Threshold slider and field:* Set the threshold level for division of frequency ranges. When the threshold is exceeded, the frequency band defined by the Center Freq. and Bandwidth parameters is divided into upper and lower frequency ranges.
- *Speed slider and field:* Set the modulation frequency for the defined frequency band.
- *CF (Center Frequency) Modulation slider and field:* Set the intensity of center frequency modulation.
- *BW (Bandwidth) Modulation slider and field:* Set the amount of bandwidth modulation.
- *Graphic display:* Shows the frequency band defined by the Center Freq. and Bandwidth parameters.
- *Center Freq. (Frequency) knob and field:* Set the center frequency of the band that you want to process.
- *Bandwidth knob and field:* Set the width of the frequency band that you want to process.
- *Super Energy knob and field:* Control the level of the frequency range above the threshold.
- *High Level slider and field:* Blend the frequencies of the original signal—above the selected frequency band—with the processed signal.

- *Sub Energy knob and field:* Control the level of the frequency range below the threshold.
- *Low Level slider and field:* Blend the frequencies of the original signal—below the selected frequency band—with the processed signal.
- *Gain slider and field:* Set the output level of Spectral Gate.

## Surround Compressor

### Intro to Surround Compressor in Final Cut Pro

Surround Compressor, based on Compressor, is specifically designed for compression of complete surround mixes. It is commonly inserted in a surround clip that carries multichannel audio.

You can adjust the compression ratio, knee, attack, and release for the main, side, surround, and LFE channels, depending on the chosen surround format. All channels include an integrated limiter and provide independent threshold and output level controls.

You can link channels by assigning them to one of three groups. When you adjust the threshold or output parameter of any grouped channel, the parameter adjustment is mirrored by all channels assigned to the group.



Surround Compressor is divided into three sections:

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

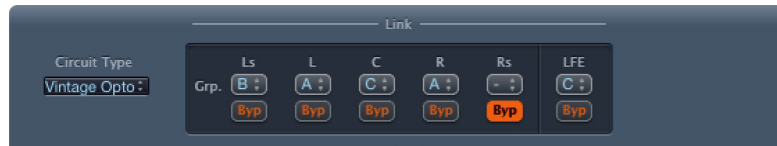
For details, see [Surround Compressor controls in Final Cut Pro](#).

# Surround Compressor controls in Final Cut Pro

Surround Compressor provides the following controls.

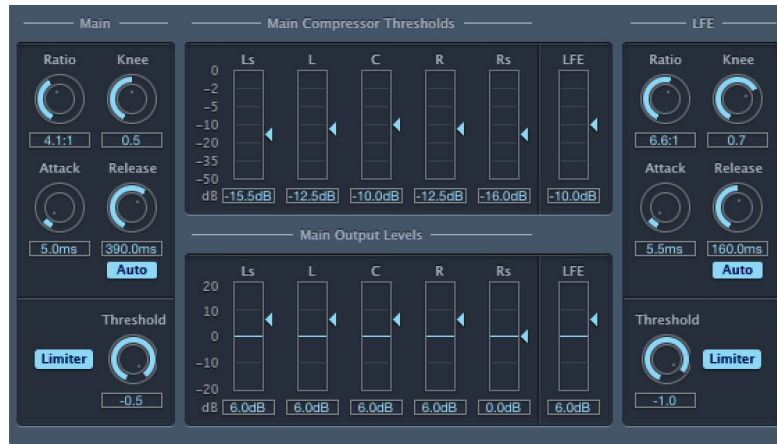
For information about adding the Surround Compressor effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

## Link controls



- *Circuit Type pop-up menu*: Choose the type of circuit emulated by the Surround Compressor. The choices are Platinum, Classic A\_R, Classic A\_U, VCA, FET, and Opto (optical).
- *Grp. (Group) pop-up menus*: Set group membership for each channel (A, B, C, or no group (indicated by -)). Moving the Threshold or Output Level slider for any grouped channel moves the sliders for all channels assigned to that group.
  - 💡 **Tip**: Press Option and Command while moving the Threshold or Output Level slider of a grouped channel to temporarily unlink the channel from the group. This allows you to make independent threshold settings while maintaining the side-chain detection link necessary for a stable surround image.
- *Byp (Bypass) buttons*: Independently bypass the corresponding channel, unless the channels are grouped. If the channel belongs to a group, all channels in the group are bypassed.
- *Detection pop-up menu*: Choose whether Surround Compressor uses the maximum level of each signal (Max) or the summed level of all signals (Sum) to exceed or fall below the threshold.
  - If Max is chosen, and any of the surround channels exceed or fall below the Threshold, those channels (or grouped channels) are compressed.
  - If Sum is chosen, the combined level of all channels must exceed the Threshold before compression occurs.

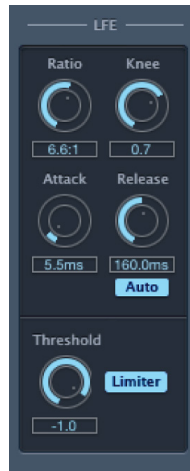
## Main controls



- *Ratio knob and field:* Set the ratio of signal reduction when the threshold is exceeded.
- *Knee knob and field:* Determine the ratio of compression at levels close to the threshold.
- *Attack knob and field:* Set the amount of time it takes to reach full compression, after the signal exceeds the threshold.
- *Release knob and field:* Set the amount of time it takes to return to 0 compression, after the signal falls below the threshold.
- *Auto button:* Make the release time dynamically adjust to the audio material.
- *Limiter button:* Turn limiting for the main channels on or off.
- *Threshold knob and field:* Set the threshold for the limiter on the main channels.
- *Main Compressor Thresholds sliders and fields:* Set the threshold level for each channel—including the LFE channel, which also has independent controls.
- *Main Output Levels sliders and fields:* Set the output level for each channel—including the LFE channel, which also has independent controls.



## LFE controls



- *Ratio knob and field:* Set the compression ratio for the LFE channel.
- *Knee knob and field:* Set the knee for the LFE channel.
- *Attack knob and field:* Set the attack time for the LFE channel.
- *Release knob and field:* Set the release time for the LFE channel.
- *Auto button:* Make the release time dynamically adjust to the audio material.
- *Threshold knob and field:* Set the threshold for the limiter on the LFE channel.
- *Limiter button:* Turn limiting on or off for the LFE channel.

# Modulation effects

## Intro to modulation effects in Final Cut Pro

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

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

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

You can also control the ratio of the affected (wet) signal and the original (dry) signal. Some modulation effects include feedback controls, which add part of the effect's output back into the effect input.

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

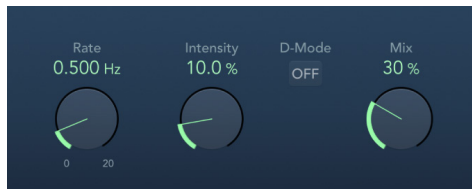
More complex Final Cut Pro modulation effects, such as Ensemble, mix several delayed signals with the original signal.

## Chorus controls in Final Cut Pro

The Chorus effect delays the original signal, and the delay time is modulated with a low-frequency oscillator (LFO). The delayed, modulated signal is then mixed with the original, dry signal.

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

For information about adding the Chorus effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Rate knob and field:* Set the frequency, or speed, of the LFO.
- *Intensity knob and field:* Set the modulation amount.
- *D-Mode button:* Turn on to introduce a spatial filtering effect that resembles a well-known vintage processor.
- *Mix knob and field:* Determine the balance between dry and wet signals.

## Ensemble controls in Final Cut Pro

Ensemble can add richness and movement to sounds, particularly when you use a high number of voices. It is useful for thickening parts, but you can also use it for strong pitch variations between voices, resulting in a detuned quality to processed material. Ensemble combines up to eight chorus effects. Two standard low-frequency oscillators (LFOs) and one random LFO enable you to create complex modulations. The graphic display represents the modulation rate and intensity of all LFOs and lets you directly adjust waveforms.

For information about adding the Ensemble effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *LFO1, LFO2, and Random On/Off buttons:* Enable or disable LFO 1, LFO 2, or the random LFO independently.
- *Rate fields:* Set the frequency of LFO 1, LFO 2, and random modulation.
- *Intensity fields:* Set the amount of LFO 1, LFO 2, and random modulation.

- *Graphic display:* Shows and lets you edit the shape and intensity of LFO 1, LFO 2, and random modulations. Move the pointer to the left third to adjust LFO 1, the center third to adjust LFO 2, or the right third to adjust the random LFO.
  - Modulation is represented by a horizontal line and waveform. The line represents the LFO intensity. The waveform represents the LFO rate. Drag the line or circular handle vertically (or use the corresponding field) to set the modulation intensity. Drag the waveform or circular handle horizontally (or use the corresponding field) to set the modulation rate.
  - *LFO1 parameters:* Drag the green handle to set the modulation rate and intensity.
  - *LFO2 parameters:* Drag the blue handle to set the modulation rate and intensity.
  - *Random LFO parameters:* Drag the white handle to set the modulation rate and intensity.
- *Voices knob and field:* Set the number of chorus instances (voices) generated in addition to the original signal.
- *Stereo Spread slider and field:* Distribute voices across the stereo or surround field. You can set a value of 200% to artificially expand the stereo or surround base. Note that monaural compatibility may suffer if you do this.
 

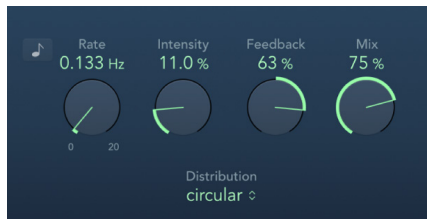
*Note:* When Ensemble is used in surround, the input signal is converted to mono before processing. In essence, you insert the Ensemble effect as a multi-mono instance.
- *Phase knob and field:* Control the phase relationship between the individual voice modulations. The value you choose here depends on the number of voices, which is why it is shown as a percentage value rather than in degrees. The value 100 (or -100) indicates the greatest possible distance between the modulation phases of all voices.
- *Volume Compensation knob and field:* Compensate for effects signal volume changes caused by adjusting the Voices value.
- *Output Mix knob and field:* Set the balance between dry and wet signals.

## Flanger controls in Final Cut Pro

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

Flanging is typically used to add a spacey or underwater quality to input signals.

For information about adding the Flanger effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- **Sync button:** Synchronize the modulation speed with the project tempo. Choose musical note values with the Rate knob.
- **Rate knob and field:** Set the frequency, or speed, of the low-frequency oscillator (LFO).
- **Intensity knob and field:** Set the modulation amount.
- **Feedback knob and field:** Set the amount of the effect signal that is routed back to the input. This can change the tonal color and make the sweeping effect more pronounced. Negative Feedback values invert the phase of the routed signal.
- **Mix knob and field:** Determine the balance between dry and wet signals.

## Phaser controls in Final Cut Pro

The Phaser effect combines the original signal with a copy that is slightly out of phase with the original. This means that the amplitudes of the two signals reach their highest and lowest points at slightly different times. The timing differences between the two signals are modulated by two independent low-frequency oscillators (LFOs). In addition, Phaser includes a filter circuit and a built-in envelope follower that tracks volume changes in the input signal, generating a dynamic control signal. This control signal alters the sweep range.

Sonically, phasing is used to create whooshing, sweeping sounds that wander through the frequency spectrum. It is a commonly used guitar effect, but it is suitable for a range of signals.

For information about adding the Phaser effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- **Stages knob and field:** Choose phaser algorithms (even numbers) or comb filtering (odd numbers).
  - The 4, 6, 8, 10, and 12 settings switch between five different phaser algorithms. All are modeled on analog circuits, with each designed for a specific application.
  - The 5, 7, 9, and 11 settings don't generate actual phasing effects. The more subtle comb filtering effects produced by odd-numbered settings can, however, be useful.

- *Sweep Mode pop-up menu*: Choose a mode that determines the impact of incoming signal levels on the frequency range. Set the frequency range with the Ceiling and Floor controls.
- *Ceiling and Floor sliders and fields*: Determine the frequency range affected by LFO modulations. Drag the green slider area between Ceiling and Floor to move the entire range.
- *Rate 1 and 2 knobs and fields*: Set the speed for each LFO.
- *Sync buttons*: Synchronize the modulation speed of each LFO with the clip. Choose musical note values with the Rate 1 and Rate 2 knobs.
- *Phase knob and field*: Control the phase relationship between individual channel modulations. Available only in stereo and surround instances. At 0°, extreme modulation values are achieved simultaneously for all channels. At 180° or –180°, there is the greatest possible distance between channel modulation phases.
- *Mix slider and fields (LFO section)*: Determine the ratio between the two LFOs.
- *Distribution pop-up menu*: Choose how phase offsets between individual channels are distributed in the surround field. Choose Circular, Left↔Right, Front↔Rear, Random, or New Random. Available only in surround instances.

*Note*: When you load a setting that uses the Random option, the saved phase offset value is recalled. If you want to randomize the phase setting again, choose New Random from the Distribution pop-up menu.

- *Level knob and field*: Determine the amount of effect signal routed back to the input.
- *Warmth button*: Enable or disable a distortion circuit, suitable for warm overdrive effects.
- *Low Cut and High Cut sliders and fields*: Set the cutoff frequency of the lowpass (LP) and highpass (HP) filters.
- *Filter button*: Enable or disable the filter section.
- *Mix slider and field (Out section)*: Determine the balance of dry and wet signals. Negative values result in a phase-inverted mix of the effect and direct (dry) signal.

## Scanner Vibrato controls in Final Cut Pro

Scanner Vibrato simulates the scanner vibrato section of a Hammond organ. Scanner Vibrato is based on an analog delay line consisting of several lowpass filters. The delay line is scanned by a multipole capacitor that has a rotating pickup. It is a unique effect that cannot be simulated with simple low-frequency oscillators (LFOs).

You can choose between three different vibrato and chorus types. The stereo version of the effect features two additional controls—Stereo Phase and Rate Right. These set the modulation speed independently for the left and right channels.

For information about adding the Scanner Vibrato effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

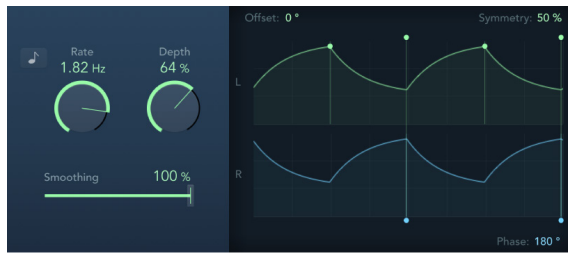


- *Type knob:* Choose from three Vibrato positions (V1, V2, and V3) or three Chorus positions (C1, C2, and C3).
  - In each of the Vibrato positions, only the delay line signal is heard. Each vibrato type has a different intensity.
  - In the three Chorus positions (C1, C2, and C3), the signal of the delay line is mixed with the original signal. Mixing a vibrato signal with an original, statically pitched signal results in a chorus effect. This organ-style chorus sounds different from the Chorus plug-in.
  - In the C0 setting, neither the chorus nor the vibrato is enabled.
- *Depth knob and field:* Set the intensity of the chosen chorus effect type. If a vibrato effect type is chosen, this control has no effect.
- *Stereo Phase knob and field:* Determine the phase relationship between left and right channel modulations. If you set the knob to Free, you can set the modulation speed of the left and right channels independently.
- *Rate Left knob and field:* Set the modulation speed of the left channel when Stereo Phase is set to Free. If Stereo Phase is set to a value between 0° and 360°, Rate Left sets the modulation speed for both the left and right channels, and the Rate Right knob has no function.
- *Rate Right knob and field:* Set the modulation speed of the right channel when Stereo Phase is set to Free.

## Tremolo controls in Final Cut Pro

Tremolo modulates the amplitude of the incoming signal, resulting in periodic volume changes. Tremolo is commonly used in vintage guitar combo amps, where it is sometimes incorrectly referred to as *vibrato*. The graphic waveform display shows all parameters except Rate.

For information about adding the Tremolo effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- **Sync button:** Synchronize the modulation speed with the project tempo. Choose musical note values with the Rate knob.
- **Rate knob and field:** Set the frequency of the low-frequency oscillator (LFO).
- **Depth knob and field:** Determine the modulation amount.
- **Smoothing slider and field:** Change the shape of the LFO waveform. Also see the description of the Symmetry parameter, below.
- **Distribution pop-up menu:** Choose how phase offsets between individual channels are distributed in the surround field. Choose Circular, Left↔Right, Front↔Rear, Random, or New Random. Available only in surround instances.
- **Offset field:** Set the amount of left or right movement for the modulation (cycle). This results in small or large tremolo variations.
- **Symmetry field:** Skew the balance toward the upward or downward phase of waveform cycles.
 

*Note:* If Symmetry is set to 50% and Smoothing to 0%, the LFO waveform becomes rectangular. The timing of the highest volume signal is then equal to the timing of the lowest volume signal, with the switch between both states occurring abruptly.
- **Phase field:** Control the phase relationship between individual channel modulations in stereo or surround signals. At 0, modulation values are reached simultaneously for all channels. At values of 180 or -180, there is the greatest possible distance between the modulation phases of the channels.
- **Waveform display:** Shows and lets you edit modulation Offset, Symmetry, and Phase. Modulation is represented by a repeating waveform cycle for each channel. The height represents the LFO intensity, or depth. Adjust the Depth knob to change the LFO intensity. Two green handles and two blue handles let you control Offset, Symmetry, and Phase values.
  - **Offset:** Drag the field or the green handle at the left of any waveform cycle to set the left or right modulation movement.
  - **Symmetry:** Drag the field or the green handle at the right of any waveform cycle to set the balance between upward and downward waveform phases.
  - **Phase:** Drag the field or either blue handle of any waveform cycle to control the phase relationship between channel modulations.



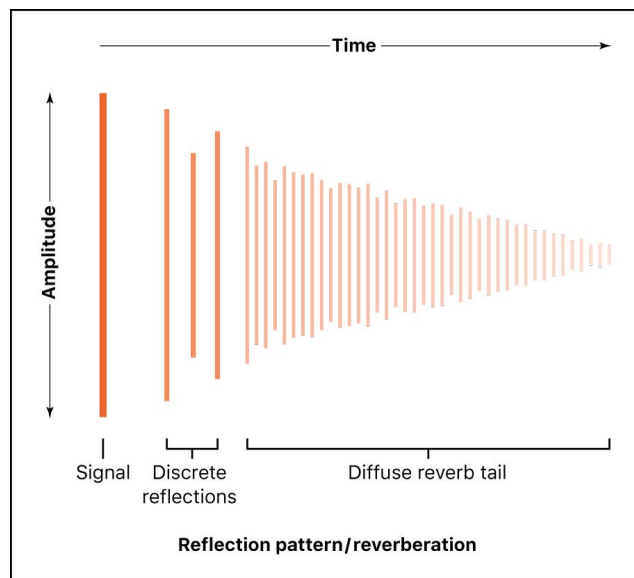
# Spaces effects

## Intro to spaces effects in Final Cut Pro

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

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

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



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

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

Today's computers make it possible to sample the reverb characteristics of real spaces, using convolution reverbs. These room characteristic sample recordings are known as *impulse responses*.

Convolution reverbs work by convolving (combining) an audio signal with the impulse response recording of a room's reverb characteristics. Final Cut Pro includes a Logic convolution reverb called Space Designer. See [Intro to Space Designer in Final Cut Pro](#).

## PlatinumVerb

### Intro to PlatinumVerb in Final Cut Pro

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

For information about adding the PlatinumVerb effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



The PlatinumVerb interface is divided into four sections:

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

For details, see [PlatinumVerb controls in Final Cut Pro](#).

## PlatinumVerb controls in Final Cut Pro

PlatinumVerb offers the following controls.

For information about adding the PlatinumVerb effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

### Early Reflections controls



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

The optimum Predelay setting depends on the type of input signal—or more precisely, the envelope of the input signal. Percussive signals generally require shorter predelays than signals where the attack fades in gradually. A good working method is to use the longest possible Predelay value before you start to hear unwanted side effects, such as an audible echo. When you reach this point, reduce the Predelay setting slightly.

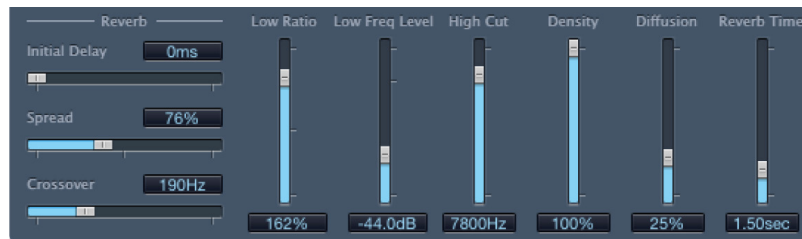
- *Room Shape slider and field:* Define the geometric form of the room. The numeric value (3 to 7) represents the number of corners in the room. The graphic display visually represents this setting.
- *Room Size slider and field:* Determine the dimensions of the room. The numeric value indicates the length of the room's walls—the distance between two corners.

- *Stereo Base slider and field*: Define the distance between the two virtual microphones that are used to capture the signal in the simulated room.

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

- *ER Scale slider and field (Extended controls area)*: Scale the early reflections along the time axis, influencing the Room Shape, Room Size, and Stereo Base parameters simultaneously.

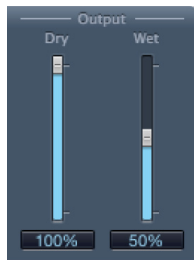
## Reverb controls



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

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

## Output controls



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

# Space Designer Convolution Reverb

## Intro to Space Designer in Final Cut Pro

Space Designer is a *convolution* reverb effect. You can use it to place your audio signals in exceptionally realistic re-creations of real-world acoustic environments.

Space Designer generates reverb by convolving, or combining, an audio signal with an impulse response (IR) reverb sample. An impulse response is a recording of a room's reverb characteristics—or, to be more precise, a recording of all reflections in a given room, following an initial signal spike. The actual impulse response file is a standard audio file.

To understand how this works, imagine a situation where Space Designer is used on a vocal clip. An IR recorded in an actual opera house is loaded into Space Designer. This IR is convolved with your vocal clip, placing the singer inside the opera house.

Convolution can be used to place your audio signal inside any space, including a speaker cabinet, a plastic toy, a cardboard box, and so on. All you need is an IR recording of the space.

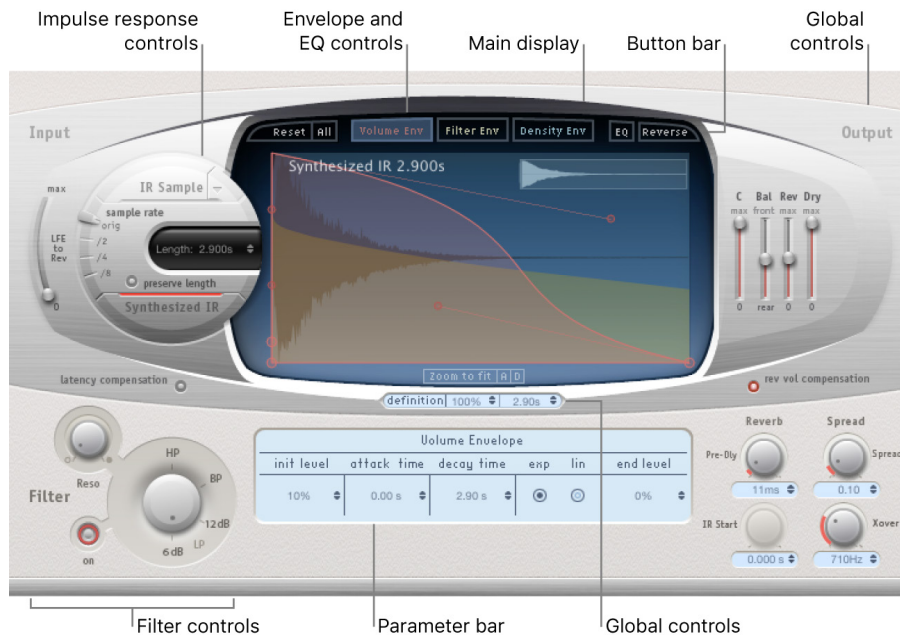
In addition to loading impulse responses, Space Designer includes an on-board impulse response synthesis facility. This enables you to create completely unique effects, particularly when the synthesized IR doesn't represent a real space.

Space Designer also offers features such as envelopes, filters, EQ, and stereo/surround balance controls, which provide precise control over the dynamics, timbre, and length of the reverberation.

Space Designer can operate as a mono, stereo, true stereo (meaning each channel is processed discretely), or surround effect.

# Space Designer interface in Final Cut Pro

The Space Designer interface consists of the following main sections.

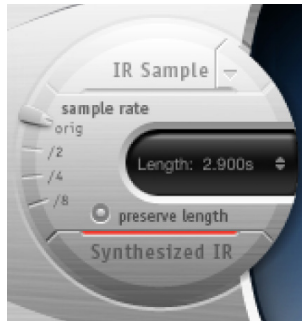


- **Impulse response controls:** Used to load, save, or manipulate (recorded or synthesized) impulse response files. The chosen IR file determines what Space Designer will use to convolve with your audio signal. See [Intro to Space Designer impulse responses in Final Cut Pro](#).
- **Envelope and EQ controls:** Use the view buttons in the button bar to switch the main display and parameter bar between envelope and EQ views. Use the main display to edit the displayed parameters graphically, and use the parameter bar to edit them numerically. See [Intro to Space Designer envelope and EQ controls in Final Cut Pro](#).
- **Filter controls:** Used to modify the timbre of the Space Designer reverb. You can choose from several filter modes, adjust resonance, and also adjust the filter envelope dynamically over time. See [Space Designer filter controls in Final Cut Pro](#).
- **Global controls:** After your IR is loaded, these controls determine how Space Designer operates on the overall signal and IR. Included are input and output controls, delay and volume compensation, predelay, and so on. See [Intro to Space Designer global controls in Final Cut Pro](#).

# Work with impulse responses

## Intro to Space Designer impulse responses in Final Cut Pro

Space Designer can use either recorded impulse response files or its own synthesized impulse responses. The circular area to the left of the main display contains the impulse response controls. These are used to determine the Impulse Response mode (IR Sample mode or Synthesized IR mode), load or create impulse responses, and set the sample rate and length.



- *IR Sample button and pop-up menu:* Click the IR Sample button to switch to IR Sample mode. In IR Sample mode, an impulse response sample is used to generate reverberation. Click the down arrow next to the IR Sample button to open the IR Sample pop-up menu, in which you can load and manipulate impulse response samples. See [Use the Space Designer IR Sample mode in Final Cut Pro](#).
- *Sample Rate slider and Preserve Length button:* The Sample Rate slider determines the sample rate of the loaded impulse response. Turn on Preserve Length to preserve the length of the impulse response when changing the sample rate. See [Space Designer IR Sample rate in Final Cut Pro](#).
- *Length field:* Adjust the length of the impulse response. See [Space Designer impulse response lengths in Final Cut Pro](#).
- *Synthesized IR button:* Click to activate Synthesized IR mode. A new synthesized impulse response is generated. This is derived from the values of the Length, envelope, Filter, EQ, and Spread parameters. See [Space Designer Synthesized IR mode in Final Cut Pro](#).

You may freely switch between a loaded impulse response sample and a synthesized impulse response without losing the settings of the other. See [Space Designer Synthesized IR mode in Final Cut Pro](#).

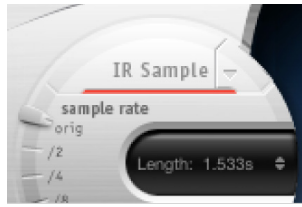
**Important:** To convolve (combine) audio in real time, Space Designer must first calculate any parameter adjustments to the impulse response. This requires a moment or two, following parameter edits, and is indicated by a blue progress bar. During this parameter edit processing time you can continue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





## Use the Space Designer IR Sample mode in Final Cut Pro

In IR Sample mode, Space Designer loads and uses an impulse response recording of an acoustic environment. This is convolved with the incoming audio signal to place it in the acoustic space provided by the IR.



### Activate IR Sample mode

1. In the Final Cut Pro timeline, select a clip with the Space Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Click the IR Sample button in the circular area to the left of the main display, and then select an impulse response file from any folder.

*Note:* If you already loaded an impulse response file, clicking the IR Sample button switches the mode from Synthesized IR to IR Sample mode.

### Manage the loaded IR file

1. In the Final Cut Pro timeline, select a clip with the Space Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

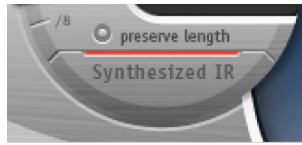
2. Click the down arrow next to the IR Sample button to open a pop-up menu with the following commands:
  - *Load IR:* Loads an impulse response sample without changing the envelopes.
  - *Load IR & Init:* Loads an impulse response sample and initializes the envelopes.
  - *Show in Finder:* Opens a Finder window that shows the location of the currently loaded IR file.

All impulse responses that ship with Final Cut Pro are installed in the /Library/Audio/Impulse Responses/Apple folder. Deconvolution files have the file extension .sdir.

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

## Space Designer Synthesized IR mode in Final Cut Pro

In Synthesized IR mode, Space Designer generates a synthesized impulse response based on the values of the Length, envelope, Filter, EQ, and Spread parameters. To switch to this mode, click the Synthesized IR button in the impulse response controls section.

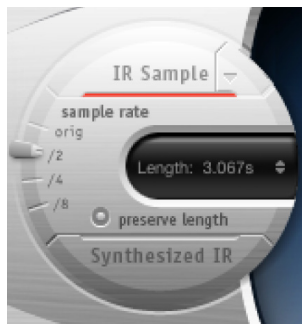


Repeated clicks of the *activated* Synthesized IR button randomly generate new impulse responses with slightly different reflection patterns. The current impulse response state (including parameter and other values that represent the reflection patterns and characteristics of the synthetic IR) is saved with the settings file.

*Note:* Clicking the Synthesized IR button while Space Designer is in IR Sample mode switches it back to the synthesized IR stored with the setting.

## Space Designer IR Sample rate in Final Cut Pro

The Space Designer effect's Sample Rate slider determines the sample rate of an impulse response.



- *Orig:* Space Designer uses the current project sample rate. When loading an impulse response, Space Designer automatically converts the sample rate of the impulse response to match the current project sample rate, if necessary. For example, this allows you to load a 44.1 kHz impulse response into a project running at 96 kHz, and vice versa.
- */2, /4, /8:* These settings are half-divisions of the preceding value—one-half, one-quarter, one-eighth. For example:
  - If the project sample rate is 96 kHz, the options are 48 kHz, 24 kHz, and 12 kHz.
  - If the project sample rate is 44.1 kHz, the options are 22.05 kHz, 11.025 kHz, and 5512.5 Hz.

Changing the sample rate upward increases—or changing it downward decreases—the frequency response (and length) of the impulse response, and to a degree the overall sound quality of the reverb. Upward sample rate changes are of benefit only if the original IR sample actually contains higher frequencies. When you are reducing the sample rate, use your ears to decide if the sonic quality meets your needs.

*Note:* Natural room surfaces—except concrete and tiles—tend to have minimal reflections in the higher frequency ranges, making the half-rate and full-rate IRs sound almost identical.

When you select half the sample rate, the impulse response becomes twice as long. The highest frequency that can be reverberated is halved. This results in a behavior that is much like doubling every dimension of a virtual room—multiplying a room’s volume by eight.

Another benefit of reducing the sample rate is that processing requirements drop significantly, making half-sample rate settings useful for large, open spaces.

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

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

If you are using Space Designer in a project that has a higher sample rate than the impulse response, you may also want to reduce the impulse response sample rate. Make sure Preserve Length is turned on. This cuts CPU power consumption without compromising reverb quality. There is no loss in reverb quality, because the impulse response does not benefit from the higher project sample rate.

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

## Space Designer impulse response lengths in Final Cut Pro

You can use the Space Designer effect’s Length parameter to set the length of the impulse response—sampled or synthesized.

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

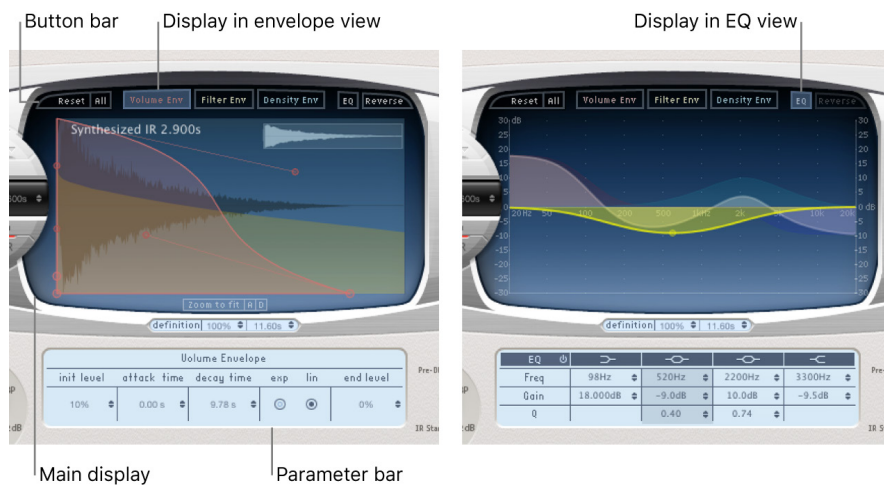
When you’re using an impulse response file, the Length parameter value can’t exceed the length of the actual impulse response *sample*. Longer impulse responses (sampled or synthesized) place a higher strain on the CPU.

# Envelope and EQ controls

## Intro to Space Designer envelope and EQ controls in Final Cut Pro

Space Designer's main interface area is used to show and edit envelope and EQ parameters. It consists of three components: the button bar at the top, the main display, and the parameter bar.

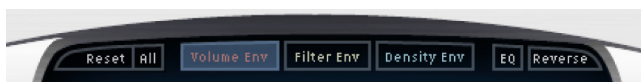
- The button bar is used to choose the current view or edit mode.
- The main display shows, and allows you to graphically edit, either the envelope or the EQ curve.
- The parameter bar displays, and allows you to numerically edit, either the envelope or the EQ curve.



## Space Designer button bar in Final Cut Pro

The Space Designer effect's button bar is used to switch the main display and parameter bar between envelope and EQ views. It also includes buttons that reset the envelopes and EQ or reverse the IR.

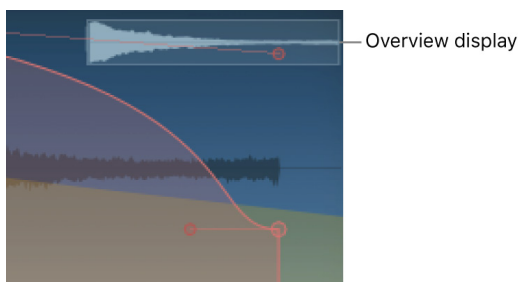
For information about adding the Space Designer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Reset button*: Reset the currently displayed envelope or EQ to its default values.
- *All button*: Reset all envelopes and the EQ to default values.
- *Volume Env button*: Display the volume envelope in the foreground of the main display. The other envelope curves are shown as transparencies in the background. See [Space Designer volume envelope controls in Final Cut Pro](#).
- *Filter Env button*: Display the filter envelope in the foreground of the main display. The other envelope curves are shown as transparencies in the background. See [Space Designer filter controls in Final Cut Pro](#).
- *Density Env button*: Display the density envelope in the foreground of the main display. The other envelope curves are shown as transparencies in the background. See [Space Designer Synthesized IR mode in Final Cut Pro](#).
- *EQ button*: Display the four-band parametric EQ in the main display. See [Work with the Space Designer EQ in Final Cut Pro](#).
- *Reverse button*: Reverse the impulse response and envelopes. When the impulse response is reversed, you are effectively using the tail rather than the front end of the sample. You may need to change the Pre-Dly and other parameter values when reversing.

## Space Designer envelope view controls in Final Cut Pro

When the Space Designer effect is set to display envelopes, the main display offers the following zoom and navigation controls (not shown in EQ view).



- *Overview display*: Indicates which portion of the impulse response file is currently visible in the main display, helping you to orient yourself when zooming.
- *Zoom to Fit button*: Click to display the entire impulse response waveform in the main display. Any envelope length changes are automatically reflected.
- *A and D buttons*: Click to limit the Zoom to Fit function to the attack and decay portions of the currently selected envelope shown in the main display. The A and D buttons are available only when you are viewing the volume and filter envelopes.

## Edit Space Designer envelopes in Final Cut Pro

In the Space Designer effect, you can edit the volume and filter envelopes of all IRs and the density envelope of synthesized IRs. All envelopes can be adjusted both graphically in the main display and numerically in the parameter bar.

Whereas some parameters are envelope-specific, all envelopes consist of the Attack Time and Decay Time parameters. The combined total of the Attack Time and Decay Time parameters is equal to the total length of the synthesized or sampled impulse response, unless the decay time is reduced. See [Space Designer impulse response lengths in Final Cut Pro](#).

The large nodes are value indicators of the parameters shown in the parameter bar below—Init Level, Attack Time, Decay Time, and so on. If you edit any numerical value in the parameter bar, the corresponding node moves in the main display.

### Move an envelope node graphically in Space Designer

1. In the Final Cut Pro timeline, select a clip with the Space Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Drag the node in one of the available directions.

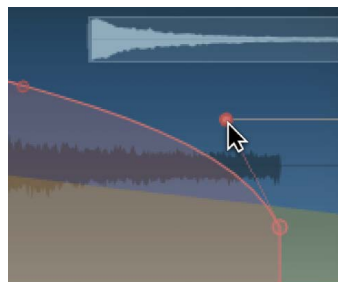
Two arrows are shown when you move the cursor over any node in the main display, indicating possible movements.

### Change Space Designer's envelope curve shape graphically

1. In the Final Cut Pro timeline, select a clip with the Space Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

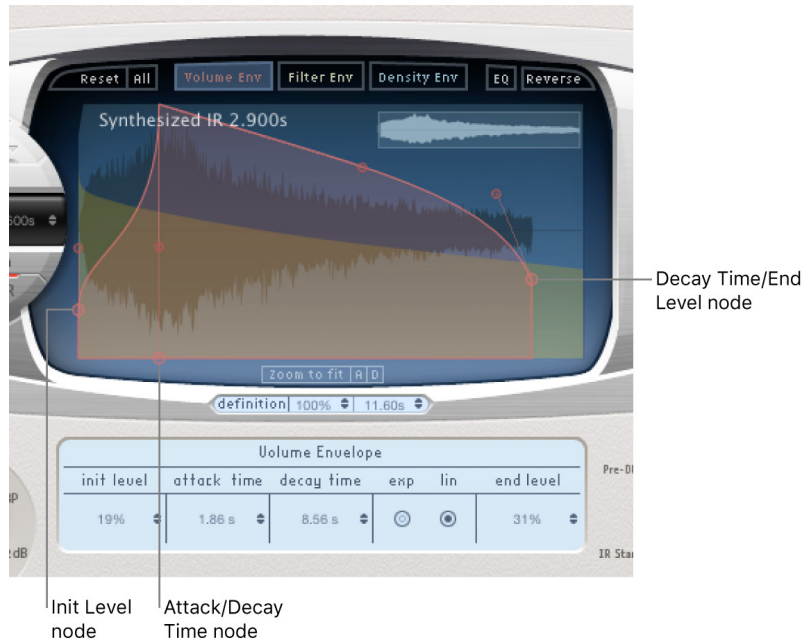
2. Drag the envelope curve in the main display.
3. Drag the small nodes attached to a line for fine adjustments to envelope curves. These nodes are tied to the envelope curve itself, so you can view them as envelope handles.



Move the nodes vertically or horizontally to change the shape of the envelope curve.

## Space Designer volume envelope controls in Final Cut Pro

The Space Designer effect's volume envelope is used to set the reverb's initial level and adjust how the volume changes over time. You can edit all volume envelope parameters numerically, and many can also be edited graphically (see [Edit Space Designer envelopes in Final Cut Pro](#)).



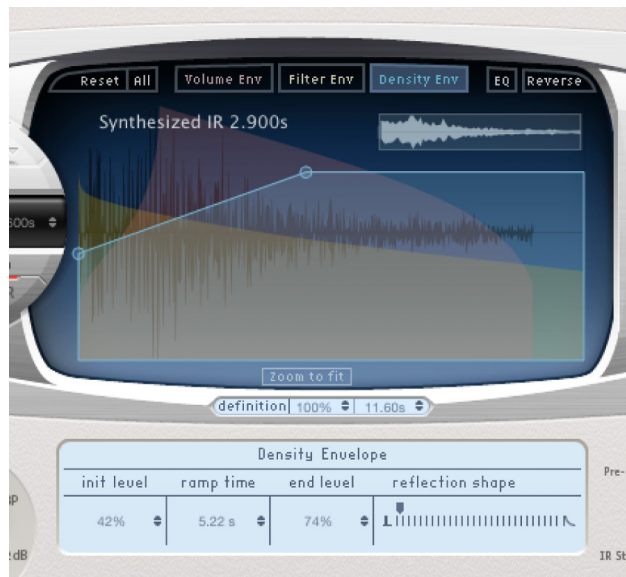
- **Init Level field:** Set the initial volume level of the impulse response attack phase. The value is expressed as a percentage of the full-scale volume of the impulse response file. The attack phase is generally the loudest point of the impulse response. Set Init Level to 100% to ensure maximum volume for the early reflections.
- **Attack Time field:** Determine the length of time before the decay phase of the volume envelope begins.
- **Decay Time field:** Set the length of the decay phase.
- **Volume decay mode buttons:** Set the volume decay curve type.
  - **Exp:** The output of the volume envelope is shaped by an exponential algorithm, to generate the most natural-sounding reverb tail.
  - **Lin:** The volume decay will be more linear (and less natural sounding).
- **End Level field:** Set the end volume level. The value is expressed as a percentage of the overall volume envelope.
  - If set to 0%, you can fade out the tail.
  - If set to 100%, you can't fade out the tail, and the reverb stops abruptly (if the end point falls within the tail).
  - If the end time falls outside the reverb tail, End Level has no effect.

## Space Designer density envelope controls in Final Cut Pro

The Space Designer effect's density envelope allows you to control the density of the synthesized impulse response over time. You can adjust the density envelope numerically in the parameter bar, and you can edit the Init Level, Ramp Time, and End Level parameters using the techniques described in [Edit Space Designer envelopes in Final Cut Pro](#).

*Note:* The density envelope is available only in Synthesized IR mode.

For information about adding the Space Designer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

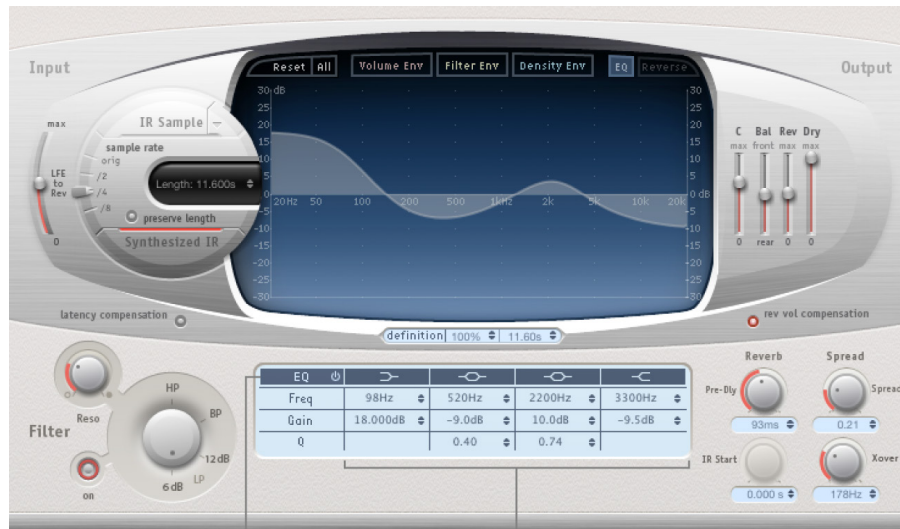


- *Init Level field:* Set the initial density (the average number of reflections in a given period of time) of the reverb. Lowering the density levels results in audible reflection patterns and discrete echoes.
- *Ramp Time field:* Adjust the length of time elapsed between the Initial and End Density levels.
- *End Level field:* Set the density of the reverb tail. If you select an End Level value that is too low, the reverb tail will sound grainy. You may also find that the stereo spectrum is affected by lower values.
- *Reflection Shape slider:* Determine the steepness (shape) of the early reflection clusters as they bounce off the walls, ceiling, and furnishings of the virtual space. Small values result in clusters with a sharp contour, and large values result in an exponential slope and a smoother sound. This is handy when re-creating rooms constructed of different materials. Reflection Shape, in conjunction with suitable settings for the envelopes, density, and early reflection, can help you create rooms of almost any shape and material.



## Work with the Space Designer EQ in Final Cut Pro

Space Designer features a four-band EQ that consists of two parametric mid-bands plus two shelving filters (one low shelving filter and one high shelving filter). You can edit the EQ parameters numerically in the parameter bar, or graphically in the main display.



EQ On/Off button

Individual EQ band buttons

- *EQ On/Off button*: Enable or disable the entire EQ section.
- *Individual EQ band buttons*: Enable or disable individual EQ bands.
- *Frequency fields*: Set the frequency for the selected EQ band.
- *Gain fields*: Adjust the gain cut or boost for the selected EQ band.
- *Q fields*: Set the Q factor for the two parametric bands. The Q factor can be adjusted from 0.1 (very narrow) to 10 (very wide).

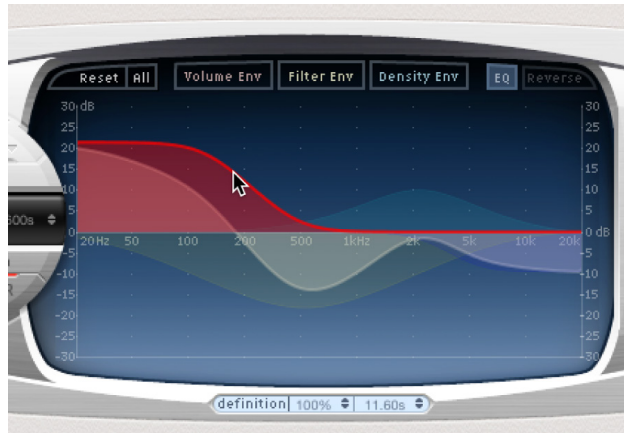
## Graphically edit an EQ curve in Space Designer

1. In the Final Cut Pro timeline, select a clip with the Space Designer effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. Enable the EQ and one or more bands with the EQ On/Off and EQ band buttons in the top row of the parameter bar.
3. Drag the pointer horizontally over the main display.

When the pointer is in the access area of a band, the corresponding curve and parameter area is automatically highlighted and a pivot point is displayed.



4. Drag horizontally to adjust the frequency of the band.
5. Drag vertically to increase or decrease the Gain of the band.
6. Vertically drag the (illuminated) pivot point of a parametric EQ band to raise or lower the Q value.

## Space Designer filter controls in Final Cut Pro

Space Designer's filter provides control over the timbre of the reverb.

You can select from several filter types and also have envelope control over the filter cutoff, which is independent from the volume envelope. Changes to filter settings result in a recalculation of the impulse response, rather than a straight change to the sound as it plays through the reverb.

For information about adding the Space Designer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

## Main filter controls

The main filter controls are found at the lower-left corner of the interface.

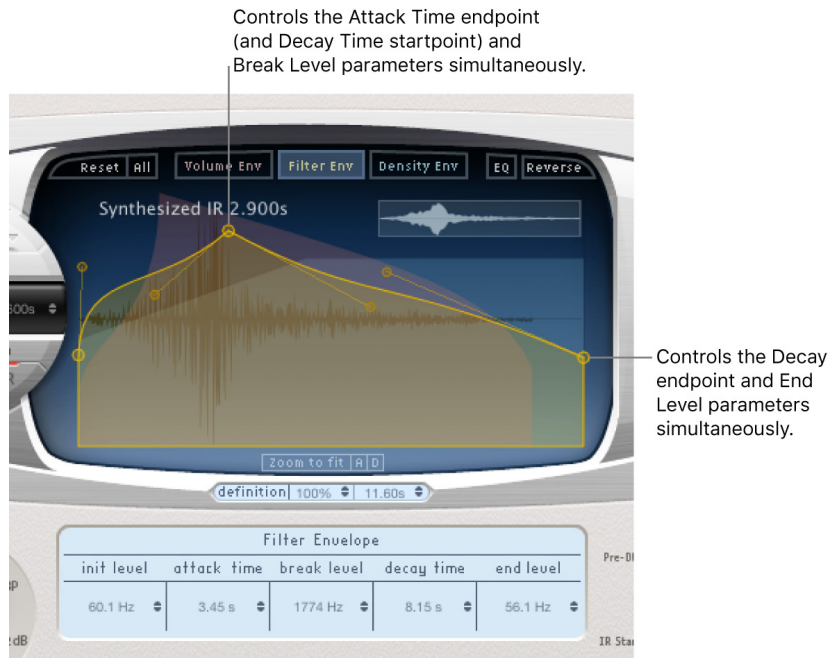


- *Filter On/Off button*: Turn the filter section on or off.
- *Filter Mode knob*: Determine the filter mode.
  - *6 dB (LP)*: Bright, good general-purpose filter mode. It can be used to retain the top end of most material, while still providing some filtering.
  - *12 dB (LP)*: Useful where you want a warmer sound, without drastic filter effects. It is handy for smoothing out bright reverbs.
  - *BP*: 6 dB per octave design. Reduces the low and high end of the signal, leaving the frequencies around the cutoff frequency intact.
  - *HP*: 12 dB per octave/two-pole design. Reduces the level of frequencies that fall below the cutoff frequency.
- *Reso (Resonance) knob*: Emphasize frequencies above, around, or below the cutoff frequency. The impact of the Resonance knob on the sound is highly dependent on the chosen filter mode, with steeper filter modes resulting in more pronounced tonal changes.

## Filter envelope controls

The filter envelope is shown in the main display when the Filter Env button is active. It provides control of the filter cutoff frequency over time. All filter envelope parameters can be adjusted either numerically in the parameter bar or graphically in the main display using the techniques discussed in [Edit Space Designer envelopes in Final Cut Pro](#).

*Note:* Activation of the filter envelope automatically enables the main filter.



- *Init Level field:* Set the initial cutoff frequency of the filter envelope.
- *Attack Time field:* Determine the time required to reach the Break Level (see below).
- *Break Level field:* Set the maximum filter cutoff frequency that the envelope reaches. It also acts as the separation point between the attack and decay phases of the overall filter envelope. In other words, when this level has been reached after the attack phase, the decay phase begins. You can create interesting filter sweeps by setting the Break Level to a value lower than the Init Level.
- *Decay Time field:* Determine the time required (after the Break Level point) to reach the End Level value.
- *End Level field:* Set the cutoff frequency at the end of the filter envelope decay phase.

# Global controls

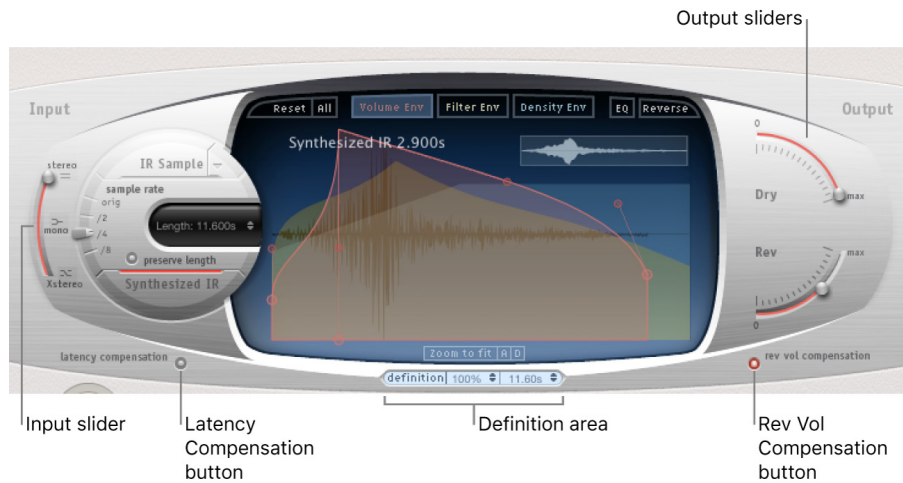
## Intro to Space Designer global controls in Final Cut Pro

Space Designer's global controls affect the overall output or behavior of the effect. The global controls are divided into two sections—those around the main display, and those below the main display.

For information about adding the Space Designer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

### Global controls: Upper section

These controls are found around the main display.



- *Input slider*: Determine how Space Designer processes a stereo or surround input signal. See [Space Designer Input slider in Final Cut Pro](#).
- *Latency Compensation button*: Turn Space Designer's internal latency compensation feature on or off. See [Space Designer Latency Compensation in Final Cut Pro](#).
- *Definition area*: Switch to a less defined IR set, in order to emulate reverb diffusion and save CPU resources. See [Space Designer Definition controls in Final Cut Pro](#).
- *Rev Vol Compensation button*: Engage Space Designer's internal IR volume matching function. See [Space Designer Rev Vol Compensation in Final Cut Pro](#).
- *Output sliders*: Adjust output levels. See [Space Designer output sliders in Final Cut Pro](#).

## Global controls: Lower section

These controls are found below the main display.



- *Pre-Dly (Predelay) knob*: Set the reverb's predelay time, or time between the original signal and the first reflections from the reverb. See [Space Designer Pre-Dly \(Predelay\) in Final Cut Pro](#).
- *IR Start knob*: Set the playback start point in the impulse response sample. See [Space Designer IR Start controls in Final Cut Pro](#).
- *Spread and Xover knobs (synthesized IRs only)*: Use Spread to adjust the perceived width of the stereo or surround field. Use Xover to set the crossover frequency in hertz. Any synthesized impulse response frequency that falls below this value will be affected by the Spread parameter. See [Space Designer spread controls in Final Cut Pro](#).

## Space Designer Input slider in Final Cut Pro

The Space Designer effect's Input slider behaves differently in stereo or surround instances. The slider does not appear in mono or mono-to-stereo instances.

- *In stereo instances*: The Input slider determines how a stereo signal is processed.
- *In surround instances*: The Input slider determines how much low-frequency effects (LFE) signal is mixed with the surround channels routed into the reverb.



**Stereo**      **Surround**

For information about adding the Space Designer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

## Input slider: Stereo mode

- *Stereo setting (top of slider):* The signal is processed on both channels, retaining the stereo balance of the original signal.
- *Mono setting (middle of slider):* The signal is processed in mono.
- *Xstereo setting (bottom of slider):* The signal is inverted, with processing for the right channel occurring on the left, and vice versa.
- *In-between positions:* A mixture of stereo to mono crossfeed signals is produced.

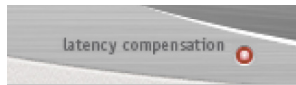
## Input slider: Surround mode

- *Surround Max setting (top of slider):* The maximum amount of LFE signal is mixed with the other surround channels.
- *Surround 0 setting (bottom of slider):* The entire LFE signal is passed through the reverb unprocessed.
- *In-between positions:* A mixture of LFE and surround channel information is processed.

## Space Designer Latency Compensation in Final Cut Pro

The complex calculations made by Space Designer take time, which results in a processing delay, or *latency*, between the direct input signal and the processed output signal. When activated, the Latency Compensation feature delays the direct signal (in the Output section) to match the processing delay of the effect signal.

*Note:* This is not related to latency compensation in the host application. This compensation feature occurs *entirely within* Space Designer.

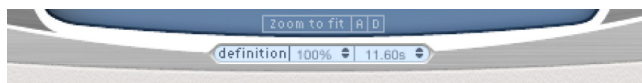


Space Designer's processing latency is 128 samples at the original sample rate, and it doubles at each lower sample rate division. If you set Space Designer's Sample Rate slider to "/2," the processing latency increases to 256 samples. Processing latency does not increase in surround mode or at sample rates above 44.1 kHz.

## Space Designer Definition controls in Final Cut Pro

The Space Designer effect's Definition parameter emulates the diffusion of natural reverb patterns. When used at values of less than 100% it also reduces CPU processing requirements.

*Note:* The Definition steppers are visible below the main display only when you have loaded CPU-intensive synthesized IRs.



Natural reverbs contain most of their spatial information in the first few milliseconds. Toward the end of the reverb, the pattern of reflections—signals bouncing off walls and so on—becomes more diffuse. In other words, the reflected signals become quieter and increasingly nondirectional, containing far less spatial information.

To emulate this phenomenon—as well as to conserve CPU power—you can configure Space Designer to use the full IR resolution only at the onset of the reverb, and to use a reduced IR resolution toward the end of the reverb.

The Definition parameter defines the crossover point—where the switch to the reduced IR resolution occurs. It is displayed in both milliseconds, indicating when the crossover occurs, and as a percentage—100% is equal to the length of the full-resolution IR.

## Space Designer Rev Vol Compensation in Final Cut Pro

The Space Designer effect's Rev Vol Compensation (Reverb Volume Compensation) feature attempts to match the perceived (not actual) volume differences between impulse response files.



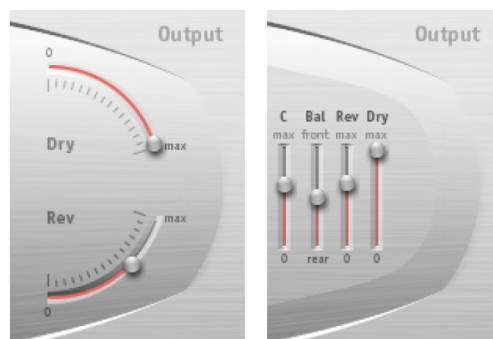
It is turned on by default and should generally be left on, although you may find that it isn't successful with all types of impulse responses. If this is the case, turn it off and adjust input and output levels accordingly.

## Space Designer output sliders in Final Cut Pro

The output controls enable you to adjust the balance between the direct (dry) and processed signals. The controls that are available depend on Space Designer's input configuration.

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

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



**Mono/Stereo**

**Surround**

For information about adding the Space Designer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



## Mono/Stereo output configuration controls

- *Dry slider*: Set the level of the non-effect (dry) signal.
- *Rev (Reverb) slider*: Adjust the output level of the effect (wet) signal.

## Surround output configuration controls

- *C (Center) slider*: Adjust the output level of the center channel independently of other surround channels.
- *Bal (Balance) slider*: Set the level balance between the front (L-C-R) and rear (Ls-Rs) channels.
  - In 7.1 ITU surround, the balance pivots around the Lm-Rm speakers, taking the surround angles into account.
  - With 7.1 SDDS surround, the Lc-Rc speakers are considered front speakers.
- *Rev (Reverb) slider*: Adjust the output level of the effect (wet) signal.
- *Dry slider*: Set the overall level of the non-effect signal.

## Space Designer Pre-Dly (Predelay) in Final Cut Pro

Predelay is the amount of time that elapses between the original signal and the initial early reflections of the reverberation.

For a room of any given size and shape, predelay determines the distance between the listener and the walls, ceiling, and floor. Space Designer allows you to adjust the Pre-Dly parameter beyond the range that would be considered natural.

In practice, an extremely short predelay tends to make it difficult to pinpoint the position of the signal source. It can also color the sound of the original signal. On the other hand, an excessively long predelay can be perceived as an unnatural echo. It can also divorce the original signal from its early reflections, leaving an audible gap between the original and reverb signals.

The ideal predelay setting for different sounds depends on the properties of—or more accurately, the envelope of—the original signal. Percussive signals generally require shorter predelays than signals where the attack fades in gradually, such as strings. A good rule of thumb is to use the longest predelay possible before unwanted side effects, such as an audible echo, begin materializing.

These guidelines are intended to help you design realistic-sounding spaces that are suitable for different signals. If you want to create unnatural sound stages or otherworldly reverbs and echoes, you can experiment with the Pre-Dly parameter.

## Space Designer IR Start controls in Final Cut Pro

The Space Designer effect's IR Start parameter enables you to shift the playback start point of the impulse response, which effectively cuts off the beginning of the impulse response.

This can be useful for eliminating level peaks at the beginning of the impulse response sample. Its use also affords a number of creative options, particularly when combined with the Reverse function. See [Space Designer button bar in Final Cut Pro](#).

*Note:* The IR Start parameter is not available or required in Synthesized IR mode because, by design, the Length parameter provides identical functionality.

## Space Designer spread controls in Final Cut Pro

The Space Designer effect's Spread and Xover knobs enhance the perceived width of the signal, without losing the directional information of the input signal normally found in the higher frequency range. Low frequencies are spread to the sides, reducing the amount of low-frequency content in the center—allowing the reverb to nicely wrap around the mix. The Spread and Xover knobs function only in Synthesized IR mode.

*Note:* As these parameters adjust stereo or surround processing, they have *no* impact when Space Designer is used as a mono plug-in.

For information about adding the Space Designer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Spread knob and field:* Extend the stereo or surround base to frequencies that fall below the frequency determined by the Xover (crossover) parameter.
  - At a Spread value of 0.00, no stereo or surround information is added (although the inherent stereo or surround information of the source signal and reverb are retained).
  - At a value of 1.00, the left and right channel divergence is at its maximum.
- *Xover knob and field:* Set the crossover frequency in hertz. Any synthesized impulse response frequency that falls below this value is affected by the Spread parameter (at values over 0).

## Automating Space Designer in Final Cut Pro

Space Designer can't be fully automated—unlike most other Logic effects, which can be. This is because Space Designer needs to reload the impulse response and recalculate the convolution before audio can be routed through it.

For more information on automating the following Space Designer parameters using keyframes, open Final Cut Pro and choose Help > Final Cut Pro Help.

- Direct Output
- Reverb Output

# Specialized effects and utilities

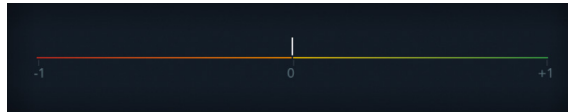
## Intro to specialized Logic effects in Final Cut Pro

Final Cut Pro includes specialized Logic effects and utilities designed to address tasks often encountered during audio production. For example, Denoiser eliminates or reduces noise below a threshold level. Exciter can add life to your recordings by generating artificial high-frequency components. SubBass generates an artificial bass signal that is derived from the incoming signal.

## Correlation Meter controls in Final Cut Pro

Correlation Meter displays the phase relationship of a stereo signal.

For information about adding the Correlation Meter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



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

## View pop-up menu

- In the View pop-up menu, choose a horizontal or vertical display and set a size.  
*Note:* You can also set a size by dragging the lower corners of the effect window.

# Denoiser

## Reduce noise with Denoiser in Final Cut Pro

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

If you use Denoiser too aggressively, however, the algorithm produces artifacts, which are usually less desirable than the existing noise. If using Denoiser produces these artifacts, you can use the three Smoothing knobs to reduce or eliminate them. See [Denoiser controls in Final Cut Pro](#).



1. In the Final Cut Pro timeline, select a clip with the Denoiser effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

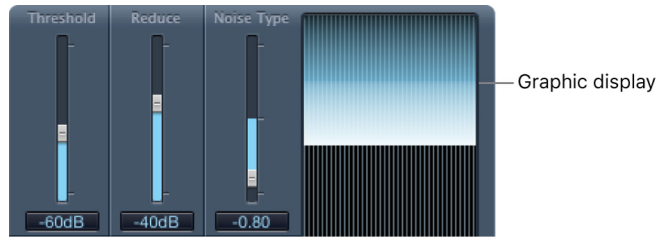
2. Locate a section of the audio where only noise is audible, and set the Threshold value so that only signals at or below this level are filtered out.
3. Play the audio signal, and set the Reduce value to the point where noise reduction is optimal but little of the appropriate signal is reduced.
4. If you encounter artifacts, use the Smoothing controls.

## Denoiser controls in Final Cut Pro

Denoiser offers the following controls.

For information about adding the Denoiser effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

## Main controls



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

*Note:* If the noise floor of your recording is very high (more than -68 dB), reducing it to a level of -83 to -78 dB should be sufficient, provided this doesn't introduce any audible side effects. This effectively reduces the noise by more than 10 dB, to less than half of the original (noise) volume.
- *Noise Type slider and field:* Determine the type of noise that you want to reduce.
  - A value of 0 equals white noise (equal frequency distribution).
  - Positive values change the noise type to pink noise (harmonic noise; greater bass response).
  - Negative values change the noise type to blue noise (hissy tape noise).
- *Graphic display:* Shows how the lowest volume levels of your audio material—which should be mostly, or entirely, noise—are reduced. Changes to parameters are instantly reflected here.

## Smoothing controls



- *Frequency knob and field:* Adjust how smoothing is applied to neighboring frequencies. If Denoiser recognizes that only noise is present on a certain frequency band, the higher you set the Frequency parameter, the more it changes the neighboring frequency bands to avoid glass noise.
- *Time knob and field:* Set the time required by Denoiser to reach (or release) maximum reduction. This is the simplest form of smoothing.
- *Transition knob and field:* Adjust how smoothing is applied to neighboring volume levels. If Denoiser recognizes that only noise is present in a certain volume range, the higher you set the Transition parameter, the more similar-level values are changed, in order to avoid glass noise.

# Direction Mixer

## Intro to Direction Mixer in Final Cut Pro

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

Direction Mixer works with any type of stereo recording, regardless of the miking technique used. For information about XY, AB, and MS recordings, see [Stereo miking techniques and the Direction Mixer](#).

### Using the Spread controls

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

When working with LR signals, keep in mind the following:

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

The following applies when working with MS signals:

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

### Using the Direction controls

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

The following applies when working with LR signals:

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

The following applies when working with MS signals:

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

## Direction Mixer controls in Final Cut Pro

Direction Mixer has the following controls.

For information about adding the Direction Mixer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Input buttons:* Click the LR button if the input signal is a standard *left/right* signal, and click the MS button if the signal is *middle and side* encoded.
- *Spread slider and field:* Determine the spread of the stereo base in LR input signals. Determine the level of the side signal in MS input signals.
- *Direction knob and field:* Determine the pan position for the middle—the center of the stereo base—of the recorded stereo signal.

## Stereo miking techniques and the Final Cut Pro Direction Mixer

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

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

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

### AB miking

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

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

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

## XY miking

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

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

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

## MS miking

In a Middle Side (MS) recording, two microphones are positioned as closely together as possible—usually placed on a stand or hung from the studio ceiling. One is a cardioid (or omnidirectional) microphone that directly faces the sound source you want to record—in a straight alignment. The other is a bidirectional microphone, with its axes pointing to the left and right of the sound source at 90° angles. The cardioid microphone records the middle signal to one side of a stereo recording. The bidirectional microphone records the side signal to the other side of a stereo recording. MS recordings made in this way can be decoded by Direction Mixer.

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

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

MS is ideal for all situations where you need to retain absolute mono compatibility. The advantage of MS recordings over XY recordings is that the stereo middle is positioned on the main recording direction (on-axis) of the cardioid microphone. This means that slight fluctuations in frequency response that occur off the on-axis—as is the case with every microphone—are less troublesome, because the recording always retains mono compatibility.



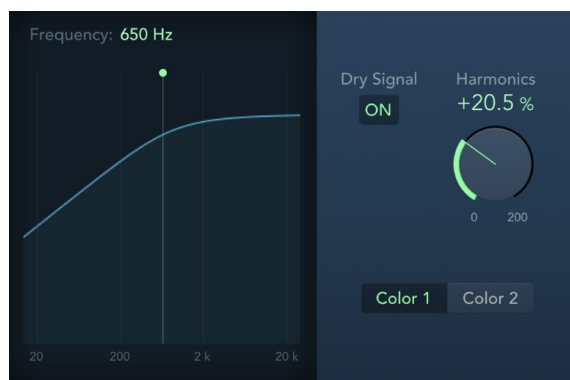
# Exciter controls in Final Cut Pro

Exciter generates high-frequency components that are not part of the original signal. It does this by utilizing a nonlinear distortion process that resembles the one used to produce overdrive and distortion effects.

Unlike this process, however, the Exciter distortion process involves passing the input signal through a highpass filter before feeding it into the harmonics (distortion) generator. Artificial harmonics are thus added to the original signal, and these added harmonics contain frequencies at least one octave above the threshold of the highpass filter. The distorted signal is then mixed with the original, dry signal.

You can use Exciter to add life to recordings, particularly audio tracks with a weak treble frequency range. You can also use Exciter to enhance guitar tracks.

For information about adding the Exciter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Frequency field:* Drag vertically to set the cutoff frequency of the highpass filter. The input signal passes through the filter before harmonic distortion is introduced.
- *Frequency display:* Shows the frequency range used as the source signal for the excite process. You can drag the green line or handle to set the cutoff frequency.
- *Dry Signal button:* Turn on to mix the original (pre-effect) signal with the effect signal. Turn off to hear only the effect signal.
- *Harmonics knob and field:* Set the ratio between the effect and original signals. If the Dry Signal button is turned off, this parameter has no effect.

*Note:* In most cases, it is preferable to select higher Frequency and Harmonics values, because human ears cannot easily distinguish between the artificial and original high frequencies.

- *Color 1 and Color 2 buttons:* Turn on Color 1 to generate a less dense harmonic distortion spectrum. Color 2 generates more intense harmonic distortion.

*Note:* Color 2 also introduces more intermodulation distortions, which can result in unpleasant artifacts.

# MultiMeter

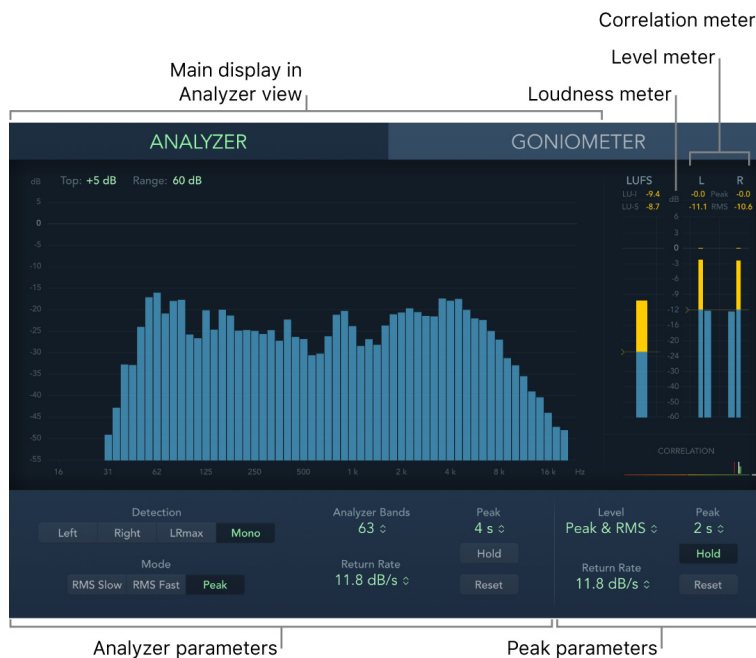
## Intro to MultiMeter in Final Cut Pro

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

- An Analyzer to view the level of each 1/3-octave or major second frequency band
- A Goniometer for judging phase coherency in a stereo sound field
- A Loudness meter that conforms to the AES 128 specification
- An integrated Level meter to view the signal level for each channel
- A Correlation meter to spot mono phase compatibility

You can view either the Analyzer or Goniometer results in the main display area. Use the Analyzer or Goniometer buttons to switch between modes and to set parameters. The Loudness/Level and Correlation meters are always visible in the Full display, as are several common parameters.

You can set a size by dragging the lower corners of the effect window. The View pop-up menu provides further Display, Meters, or Full items that show a partial view or the complete effect interface. Control-click in the main display when a partial view is enabled to switch between modes.



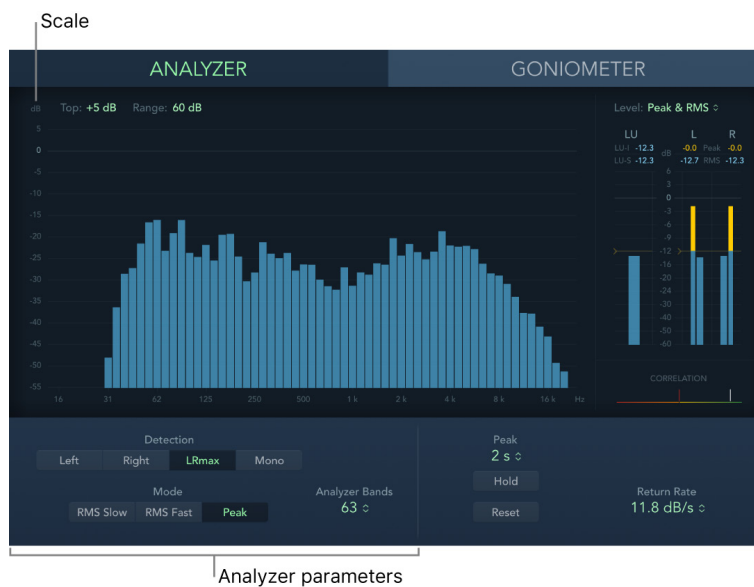
Although you can insert MultiMeter directly into any clip, it is more commonly used when you are working on the overall mix.

For information about adding the MultiMeter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

## MultiMeter Analyzer controls in Final Cut Pro

When Analyzer is selected in the MultiMeter effect, the main display shows the frequency spectrum of the input signal as 31 or 63 independent frequency bands. Each frequency band represents one-third of an octave or a major second. The Analyzer controls are used to activate Analyzer mode and to customize the appearance of the incoming signal in the main display.

For information about adding the MultiMeter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Analyzer button*: Switch the main display to the Analyzer.
- *Top and Range fields*: Change Analyzer display values by setting the maximum level (Top) and the overall dynamic range (Range).
- *Scale*: Indicates the scale of levels. Drag the scale vertically to adjust the Range value. Changing the scale is useful when analyzing highly compressed material because it makes it easier to identify small level differences.
- *Detection buttons*: Determine the channels shown in the Analyzer results in the main display.
  - *Left or Right*: Display the left or right channel.
  - *LRmax*: Display the maximum level of the stereo inputs.
  - *Mono*: Display the spectrum of the mono sum of both (stereo) inputs.
- *Mode buttons*: Determine how levels are displayed. You can select RMS Slow, RMS Fast, or Peak.
  - The two RMS modes show the effective signal average and provide a representative overview of perceived volume levels.
  - The Peak mode shows level peaks accurately.
- *Analyzer Bands pop-up menu*: Choose the number of bands shown in the Analyzer display. You can choose 31 Bands (Third-Octave) or 63 Bands (Major Second).

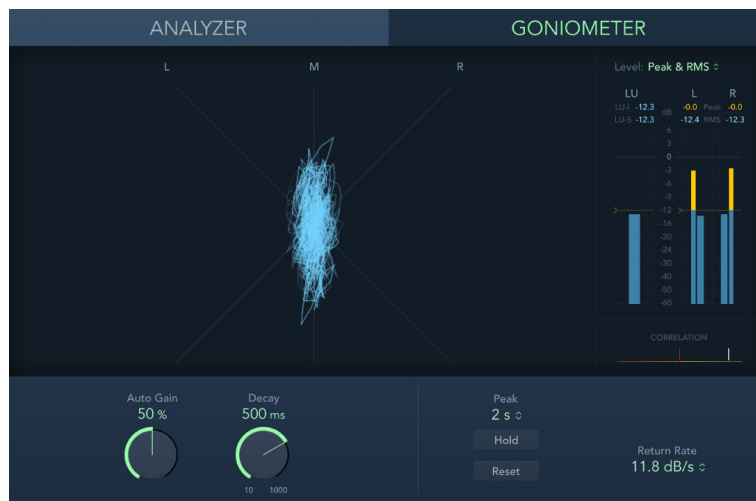
## MultiMeter Goniometer controls in Final Cut Pro

The MultiMeter effect's Goniometer helps you judge the coherence of the stereo image and determine phase differences between the left and right channels. Phase problems are easily spotted as trace cancelations along the center line, labeled *M* for *mid/mono*.

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

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

For information about adding the MultiMeter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



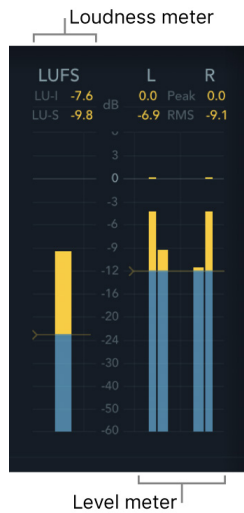
- *Goniometer button*: Switch the main display to the Goniometer.
- *Auto Gain knob and field*: Set the amount of display compensation for low input levels. You can set Auto Gain levels in 10% increments or turn Auto Gain off.

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

- *Decay knob and field*: Determine the time it takes for the Goniometer trace to fade to black.

## MultiMeter Level and Loudness controls in Final Cut Pro

The MultiMeter effect's Level meter displays the current signal level on a logarithmic scale. The Loudness meter shows the perceived level of a signal.



In the Level meter, the signal level for each channel is represented by a blue bar. Signals (above the target level) approaching the 0 dB level are represented by a yellow bar. When the level exceeds 0 dB, the portion of the bar above the 0 dB point turns red.

RMS and peak levels are shown simultaneously, with RMS levels appearing as dark blue bars and peak levels appearing as light blue bars. When the level exceeds 0 dB, the portion of the bar above the 0 dB mark turns red.

- *Peak and RMS fields:* Peak and RMS values are displayed numerically (in dB increments) above the Level meter. Click the display to reset values.

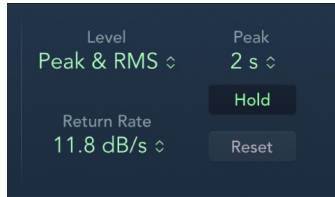
The Loudness meter shows the momentary loudness level. Loudness indicates the perceived level of a signal that is indicative of human hearing, making it a useful reference tool when mixing or mastering. The Loudness meter conforms to the AES 128 specification.

Loudness is measured in Loudness Units, relative to Full Scale (LUFS). Each LU is equivalent to one decibel. MultiMeter provides two indicators above the Loudness meter.

- *LU-I field:* Loudness Unit-Integrated, which indicates the perceived level from the start to the end of the program material.
- *LU-S field:* Loudness Unit-Short term, which indicates the perceived level of the most recent 3 seconds of program material.

For information about adding the MultiMeter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).

## Level meter controls



- *Level pop-up menu:* Choose how levels are displayed. The options are Peak, RMS, Peak & RMS, True Peak, and True Peak & RMS.
  - The two RMS options show the effective signal average and provide a representative overview of perceived volume levels.
  - The True Peak options display interpolated sample values.
  - The Peak option shows level peaks accurately.
- *Return Rate pop-up menu:* Choose how quickly analyzed signals return from peak (maximum) levels to zero or incoming signal levels. This is expressed in dB per second.
- *Peak pop-up menu:* Choose the hold time for the Level meter. Choose 2, 4, or 6 seconds—or an infinite hold time.

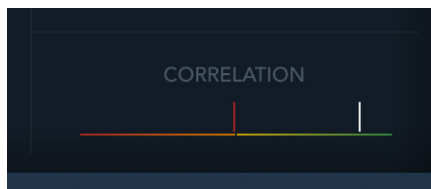
*Note:* The Hold button must be turned on for the selected time value to have an effect.

- *Hold button:* Turn on to show a small indicator for the most recent peak level. This is displayed as follows:
  - *Level meter:* A small yellow segment above each stereo level bar indicates the most recent peak level.
  - *Correlation meter:* The horizontal area around the correlation indicator denotes phase correlation deviations in real time, in both directions. A vertical line to the left of the correlation indicator shows the maximum negative phase deviation value. You can reset this line by clicking it during playback. For more information, see [MultiMeter Correlation meter in Final Cut Pro](#).
- *Reset button:* Reset all values in the Level meter.

## MultiMeter Correlation meter in Final Cut Pro

The MultiMeter effect's Correlation meter gauges the phase relationship of a stereo signal.

For information about adding the MultiMeter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



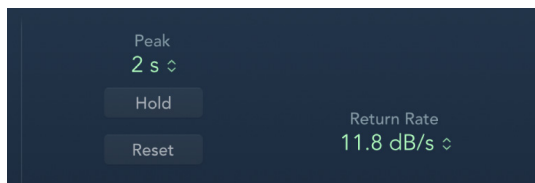
Correlation meter scale values indicate the following:

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

## MultiMeter Peak controls in Final Cut Pro

MultiMeter offers common Peak controls for the Analyzer and Goniometer metering tools. The Level and Correlation meters have separate Peak controls. See [MultiMeter Level and Loudness controls in Final Cut Pro](#).

For information about adding the MultiMeter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Peak pop-up menu:* Choose the hold time for all metering tools. Choose 2, 4, or 6 seconds—or an infinite hold time.  
*Note:* The Hold button must be turned on for the selected time value to have an effect.
- *Hold button:* Turn on peak hold for all metering tools in MultiMeter. This is displayed in the following ways:
  - *Analyzer:* A small segment above each 1/3-octave level bar indicates the most recent peak level.
  - *Goniometer:* All illuminated pixels are held during a peak hold.
- *Reset button:* Reset the peak hold segments of all metering tools.
- *Return Rate pop-up menu:* Shown only in mono instances. Choose how quickly analyzed signals return from peak/maximum levels to zero or incoming signal levels. This is expressed in dB per second.

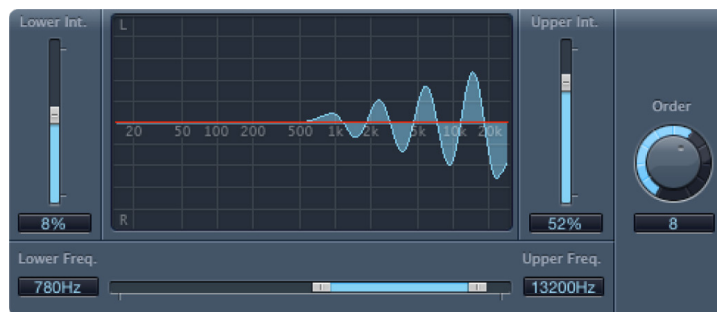
*Note:* This parameter is shown below the MultiMeter Level and Loudness meters when a stereo instance is active.

# Stereo Spread controls in Final Cut Pro

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

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

For information about adding the Stereo Spread effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



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

*Note:* When setting the Lower Int. and Upper Int. sliders, be aware that the stereo effect is most apparent in the middle and higher frequencies, so distributing low frequencies between the left and right speakers can significantly alter the energy of the overall mix. For this reason, use low values for the Lower Int. parameter, and avoid setting the Lower Freq. parameter below 300 Hz.

- *Graphic display:* Shows the number of bands the signal is divided into, and the intensity of the Stereo Spread effect in the upper and lower frequency bands. The upper section represents the left channel, and the lower section represents the right channel. The frequency scale displays frequencies in ascending order, from left to right.
- *Upper and Lower Freq. (Frequency) slider and fields:* Determine the highest and lowest frequencies that will be redistributed in the stereo image.
- *Order knob and field:* Determine the number of frequency bands that the signal is divided into. A value of 8 is usually sufficient for most tasks, but you can use up to 12 bands.



# SubBass

## Intro to SubBass in Final Cut Pro

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

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

SubBass creates two bass signals, derived from two separate portions of the incoming signal. These are defined with the High and Low parameters. See [SubBass controls in Final Cut Pro](#).

**WARNING:** Using SubBass can produce extremely loud output signals. Choose moderate monitoring levels, and only use loudspeakers that are actually capable of reproducing the very low frequencies produced. Never try to force a loudspeaker to output these frequency bands with an EQ.


SubBass is unlike a pitch shifter in that the waveform of the signal generated by SubBass is not based on the waveform of the input signal, but is sinusoidal—that is, it uses a sine wave. Given that pure sine waves rarely sit well in complex arrangements, you can control the amount of—and balance between—the generated and original signals with the Wet and Dry sliders.

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

The High Ratio and Low Ratio knobs define the transposition amount for the generated signal in each band. This is expressed as a ratio of the original signal. For example, a Ratio setting of 2 transposes the signal down one octave.

**Important:** Within each frequency band, the filtered signal should have a reasonably stable pitch in order to be analyzed correctly.

In general, narrow bandwidths produce the best results, because they avoid unwanted intermodulations. Set High Center a fifth higher than Low Center, which means a factor of 1.5 for the center frequency. Derive the sub-bass to be synthesized from the existing bass portion of the signal, and transpose by one octave in both bands (by setting Ratio to 2). Do not overdrive the process or you will introduce distortion. If you hear frequency gaps, move one or both Center frequency knobs, or widen the Bandwidth setting of one or both frequency ranges a little.

 **Tip:** Be prudent when using SubBass, and compare the extreme low-frequency content of your mixes with that of other productions. It is very easy to go overboard with it.

## SubBass controls in Final Cut Pro

SubBass offers the following controls.

For information about adding the SubBass effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



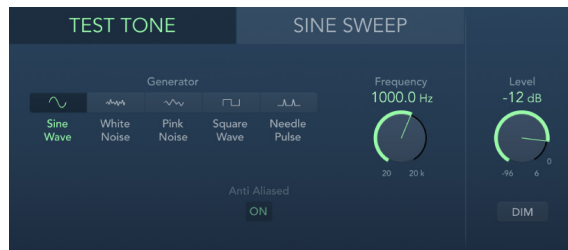
- *High Ratio knob and field:* Adjust the ratio between the generated signal and the original upper band signal.
- *High Center knob and field:* Set the center frequency of the upper band.
- *High Bandwidth knob and field:* Set the width of the upper band.
- *Graphic display:* Shows the selected upper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

# Test Oscillator controls in Final Cut Pro

Test Oscillator is useful for tuning studio equipment and instruments. It can be inserted as an instrument or effect plug-in. It operates in two modes, generating either a static frequency or a sine sweep. Click the Test Tone or Sine Sweep tab to access each mode.

- In Test Tone mode, a test signal is generated immediately when the plug-in is inserted. You can switch off the test tone by bypassing the plug-in.
- In Sine Sweep mode, a user-defined frequency spectrum tone sweep is generated when you click the Trigger button.

For information about adding the Test Oscillator effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



## Test Tone controls

- *Generator buttons:* Select the waveform to use for test tone generation. The Square Wave and Needle Pulse waveforms are available as either aliased or anti-aliased versions. The Sine Wave and Square Wave waveforms are bipolar. The Needle Pulse waveform is a unipolar impulse waveform that is always positive, which makes it a useful tool for analyzing polarity or phase issues.
- *Anti Aliased button:* Enable to use anti-aliased versions of the Square Wave or Needle Pulse waveforms.
- *Frequency knob and field:* Set the frequency of the oscillator (the default is 1 kHz). You can also double-click this field and enter a value ranging from 1 Hz to 22 kHz, exceeding the possible values that can be set with the knob. If you enter "1," a 1 Hz test tone is the result.
- *Level knob and field:* Set the overall output level. This parameter is common to both modes.
- *Dim button:* Reduce the output level by 50%. This parameter is common to both modes.

## Sine Sweep controls

- *Time knob and field:* Set the duration of the sine wave sweep.
- *Sweep Mode pop-up menu:* Choose a sweep curve—Linear or Logarithmic.
- *Start Frequency and End Frequency knobs and fields:* Set the oscillator frequency for the beginning and end of the sine sweep.

*Note:* The Frequency field shown below the Start Frequency and End Frequency controls is a real-time display of the frequency sweep.

- *Trigger pop-up menu:* Choose the sine sweep mode.
  - *Single:* Triggers the sweep once.
  - *Continuous:* Triggers the sweep indefinitely.
- *Trigger button:* Start the sine sweep of the spectrum set with the Start Frequency and End Frequency controls.

# Vocal effects

## Intro to vocal effects in Final Cut Pro

You can use the vocal effects of Final Cut Pro to correct the pitch of vocals or enhance audio signals. These effects can also be used for creating unison or slightly thickened parts, or even for creating harmony voices.

## DeEsser controls in Final Cut Pro

DeEsser is a frequency-specific compressor, designed to compress a particular frequency band within a complex audio signal. It is used to eliminate hiss (also called *sibilance*) from the signal.

The advantage of using DeEsser rather than an EQ to cut high frequencies is that it compresses the signal dynamically, rather than statically. This prevents the sound from becoming darker when no sibilance is present in the signal. DeEsser has extremely fast attack and release times.

When using DeEsser, you can set the frequency range being compressed (the Suppressor frequency) independently of the frequency range being analyzed (the Detector frequency). The two ranges can be easily compared in DeEsser's Detector and Suppressor frequency range displays.

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

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

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

For information about adding the DeEsser effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



## Detector controls

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

## Suppressor controls

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

## Center section controls

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

# Pitch Correction

## Intro to Pitch Correction in Final Cut Pro

You can use the Pitch Correction effect to correct the pitch of incoming audio signals. Improper intonation is a common problem with vocal clips, for example. The sonic artifacts that can be introduced by the process are minimal and can barely be heard, as long as your corrections are moderate.

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


*Note:* Polyphonic recordings, such as recordings of choirs, and highly percussive signals with prominent noisy portions can't be corrected to a specific pitch. Despite this, you may want to try the effect on drum signals.

### Defining the quantization grid

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

The Scale pop-up menu allows you to choose different pitch quantization grids. The scale that is set manually (with the keyboard graphic in the effect window) is called the *User Scale*. The default setting is the *Chromatic* scale. If you're unsure of the intervals used in any given scale, choose it in the Scale menu and look at the keyboard graphic. You can alter any note in the chosen scale by clicking the keyboard keys. Any such adjustments overwrite the existing user scale settings.

There is only one user scale per project. You can, however, create multiple user scales and save them as Pitch Correction effect settings files.

 **Tip:** The *drone* scale uses a fifth as a quantization grid, and the *single* scale defines a single note. Neither of these scales is meant to result in realistic singing voices, so if you're after interesting effects, you should give them both a try.

Use the Root pop-up menu to choose the root note of the scale. (If you chose User Scale or Chromatic in the Scale pop-up menu, the Root pop-up menu is nonfunctional.) You may freely transpose the major and minor scales, and scales named after chords.


## Excluding notes from Pitch Correction

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

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

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

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

 **Tip:** It's often best to correct only the notes with the most harmonic gravity. For example, choose "sus4" from the Scale pop-up menu, and set the Root note to match the project key. This limits correction to the root note, the fourth, and the fifth of the key scale. Activate the bypass buttons for all other notes and only the most important and sensitive notes are corrected, while all other singing remains untouched.

## Automating Pitch Correction

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

For more information on automating Pitch Correction, open Final Cut Pro and choose Help > Final Cut Pro Help.



## Pitch Correction controls in Final Cut Pro

The Pitch Correction effect offers the following controls.

For information about adding the Pitch Correction effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Use Global Tuning button:* Turn on to use the project's Tuning settings for the pitch correction process. If Use Global Tuning is turned off, you can use the Ref. Pitch field to freely set the desired reference tuning.
- *Normal and Low buttons:* Determine the pitch range that is scanned (for notes that need correction).
- *Ref. Pitch field:* Set the desired reference tuning, in cents (relative to the root).
- *Root pop-up menu and field:* Click to choose the root note of the scale from the Root pop-up menu.
- *Scale pop-up menu and field:* Click to choose different pitch quantization grids from the Scale pop-up menu.
- *Keyboard:* Click a key to exclude the corresponding note from pitch quantization grids. This effectively removes this key from the scale, resulting in note corrections that are forced to the nearest available pitch (key).
- *Byp (Bypass) buttons:* Click to exclude the corresponding note from pitch correction. In other words, all notes that match this pitch will not be corrected. This applies to both user and built-in scale quantization grids.
- *Bypass All button:* Compare the corrected and original signal, or use for automation changes.
- *Show Input and Show Output buttons:* Click to display the pitch of the input or output signal, respectively, on the notes of the keyboard.
- *Correction Amount display:* Indicates the amount of pitch change. The red marker indicates the average correction amount over a longer time period. You can use the display when discussing (and optimizing) the vocal intonation with a singer during a recording session.

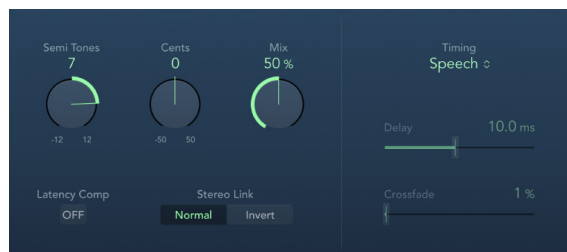
- *Response slider and field:* Set how quickly the voice reaches the corrected destination pitch. Singers use portamenti and other gliding techniques. If you choose a Response value that's too high, seamless portamenti turn into semitone-stepped glissandi, but the intonation is perfect. If the Response value is too low, the pitch of the output signal doesn't change quickly enough. The optimum setting for this parameter depends on the singing style, tempo, vibrato, and accuracy of the original performance.
- *Detune slider and field:* Detune the output signal by the set value.
- *Input Detune slider and field (Extended controls area):* Detune the input signal by the set value, thus affecting it before any pitch correction takes place. This parameter is of particular benefit when automated.

## Pitch Shifter

### Pitch Shifter controls in Final Cut Pro

Pitch Shifter provides a simple way to combine a pitch-shifted version of the signal with the original signal. Use pitch shifting to achieve the best results.

For information about adding the Pitch Shifter effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Semi Tones knob and field:* Set the pitch shift value in semitones.
- *Cents knob and field:* Control detuning of the pitch shift value in cents (1/100th of a semitone).
- *Mix knob and field:* Set the balance between the effect and original signals.
- *Latency Comp button:* Turn on to compensate for delays that may be introduced by some algorithms with particular types of source material.
- *Stereo Link buttons:* Click Normal to retain the source stereo signals. Invert swaps (inverts) stereo channel signals, with right channel processing occurring on the left, and vice versa.

- *Timing pop-up menu:* Choose an algorithm to determine how timing is derived.
  - *Drums:* Maintains the groove (rhythmic feel) of the source signal.
  - *Speech:* Provides a balance between both the rhythmic and harmonic aspects of the signal. This is suitable for complex signals such as spoken-word recordings, rap, and hybrid signals such as rhythm guitar.
  - *Vocals:* Retains the intonation of the source, making it well-suited for signals that are inherently harmonic or melodious, such as string pads.
  - *Manual:* Uses the settings of the Delay, Crossfade, and Stereo Link parameters.
  - *Pitch Tracking:* Follows the pitch of incoming audio material.

*Note:* The following controls are active only when Manual is chosen from the Timing pop-up menu.

- *Delay slider and field:* Set the amount of delay applied to the input signal. The lower the frequencies of the input signal, the higher (longer) a delay time is required—to effectively pitch shift the signal.
- *Crossfade slider and field:* Set the range (shown as a percentage of the original signal) used to analyze the input signal.

## Use Pitch Shifter in Final Cut Pro


Pitch Shifter is used most effectively when you take a structured approach.

1. In the Final Cut Pro timeline, select a clip with the Pitch Shifter effect applied, then open the effect's settings in the Audio inspector.

For information about adding the effect and showing its controls, see [Add Logic effects to clips in Final Cut Pro](#).

2. To set the amount of transposition, or pitch shift, drag the Semi Tones slider.
3. To set the amount of detuning, drag the Cents slider.
4. Click the Timing pop-up menu and choose the algorithm that best matches the material you're working with.

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

 **Tip:** While auditioning and comparing different settings, temporarily set the Mix parameter to 100% because this makes Pitch Shifter artifacts easier to hear.

# Vocal Transformer

## Intro to Vocal Transformer in Final Cut Pro

Vocal Transformer can be used to transpose the pitch of a vocal line, to augment or diminish the range of the melody, or even to reduce it to a single note that mirrors the pitches of a melody. No matter how you change the pitches of the melody, the constituent parts of the signal (formants) remain the same.

You can shift the formants independently, which means that you can turn a vocal clip into a Mickey Mouse voice while maintaining the original pitch. Formants are characteristic emphases of certain frequency ranges. They are static and do not change with pitch. Formants are responsible for the specific timbre of a given human voice.

Vocal Transformer is well suited to extreme vocal effects. The best results are achieved with monophonic signals, including monophonic instrument clips. It is not designed for polyphonic voices—such as a choir on a single clip—or other choral clips.

### Setting pitch and formant controls

Use the Vocal Transformer's Pitch parameter to transpose the pitch of the signal upward or downward. Adjustments are made in semitone steps. Incoming pitches are indicated by a vertical line below the Pitch Base field. Transpositions of a fifth upward (Pitch set to +7), a fourth downward (Pitch set to -5), or by an octave (Pitch set to  $\pm 12$ ) are the most useful, harmonically.



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

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

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

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

## Using Robotize mode

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



The Tracking slider and field work in conjunction with four buttons that immediately set the slider to the most useful values, as follows:

- *-1 (sets the slider to -100%):* All intervals are mirrored.
- *0 (sets the slider to 0%):* Delivers interesting results, with every syllable of the vocal clip being sung at the same pitch. Low values turn sung lines into spoken language.
- *1 (sets the slider to 100%):* The range of the melody is maintained. Higher values augment, and lower values diminish, the melody.
- *2 (sets the slider to 200%):* The intervals are doubled.

The Pitch Base parameter is used to transpose the note that the Tracking parameter is following. For example, with Tracking set to 0%, the pitch of the (spoken) note is transposed to the Pitch Base value.

## Vocal Transformer controls in Final Cut Pro

Vocal Transformer offers the following controls.

For information about adding the Vocal Transformer effect to a clip and showing the effect's controls, see [Add Logic effects to clips in Final Cut Pro](#).



- *Pitch knob and field:* Determine the amount of transposition applied to the input signal.
- *Robotize button:* Turn on Robotize mode, which is used to augment, diminish, or mirror the melody.
- *Pitch Base slider and field (available only in Robotize mode):* Use to transpose the note that the Tracking parameter is following.
- *Tracking slider, field, and buttons (available only in Robotize mode):* Control how the melody is changed in Robotize mode.

- *Mix slider and field*: Define the level ratio between the original (dry) and effect signals.
- *Formant knob and field*: Shift the formants of the input signal.
- *Glide slider and field (Extended controls area)*: Determine the amount of time the vocal transformation takes, allowing sliding transitions to the set Pitch value.
- *Grain Size slider and field (Extended controls area)*: The Vocal Transformer effect algorithm is based on granular synthesis. Use the Grain Size parameter to set the size of the grains, thus affecting the precision of the process. Experiment to find the best setting (try Auto first).
- *Formants pop-up menu (Extended controls area)*: Determine whether Vocal Transformer processes all formants ("Process always" setting), or only the voiced ones (Keep Unvoiced Formants setting). The Keep Unvoiced Formants option leaves sibilant sounds in a vocal performance untouched. This setting produces a more natural-sounding transformation effect with some signals.
- *Detune slider and field (Extended controls area)*: Detune the input signal by the set value. This parameter is of particular benefit when automated.

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