

Introduction

This manual contains complete documentation for FabFilter Pro-L in printer-friendly format. All information in this manual is also accessible via the Help button in FabFilter Pro-L's user interface.

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About FabFilter Pro-L

The limiter is an indispensable tool in modern mastering and mixing, and there are a lot of different flavors available on the market today. Some limiters try to be as transparent and 'safe' as possible, while others are designed to just go loud. One limiter may work best on rock music, while another performs best on electronic dance music.

FabFilter Pro-L is a professional brickwall limiter, that combines all these flavors and qualities in one plug-in, making it suitable for any type of music/audio. It can be as transparent as needed, and can go as loud as you want it to go. Equipped with excellent metering, oversampling and dithering, Pro-L is everything you need in a limiter.



Key features include four different limiting styles, all with their own characteristics, adjustable look-ahead, attack and release settings, separate channel link settings for transients and release, dithering with three different noise shape settings, up to four times oversampling, accurate and clear metering with K-System support and a unique real-time limiting display.

FabFilter Pro-L is available in VST, VST3, AU, RTAS, AS and AAX formats for Mac OS X and Windows.

Windows requirements:

32-bit: Windows 7, Vista or XP
64-bit: Windows 7 or Vista (x64)
VST 2/3 host or Pro Tools
SSE-capable processor

Mac OS X requirements:

32-bit: OS X 10.4 or higher
64-bit: OS X 10.6 or higher
AU or VST 2/3 host or Pro Tools
Intel or G4/G5 processor

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[Using FabFilter Pro-L](#)

[Acknowledgements](#)

Quick start

The installer will copy the FabFilter Pro-L plug-in into the common VST, VST 3, AU (OS X only) and Pro Tools plug-in folders on your computer. On Mac OS X, the global plug-in folders in /Library/Audio/Plug-Ins are used.

In most cases, your host will then recognize the plugin automatically. However, if the instructions below do not work, see [Manual installation](#) instead.

- **Cubase**
Choose an empty insert slot, for example in the Mixer, and select FabFilter Pro-L from the menu that appears. To use the MIDI features in Pro-L, create a new MIDI track and set its output to the Pro-L instance you have just created. (The VST 3 version of Pro-L can be found in the Dynamics section.)
- **Logic Pro**
Choose an empty insert slot on one of your audio tracks, instrument tracks or buses and select FabFilter Pro-L from the pop-up menu. You will find FabFilter Pro-L in the *Audio Units* -> *FabFilter* section (named FF Pro-L).
- **Ableton Live**
In Session view, select the track you would like to place FabFilter Pro-L on, for example by clicking the track name. At the left top of Ableton Live's interface, click on the Plug-in Device Browser icon (third icon from the top). From the plug-ins list, double-click FabFilter/FabFilter Pro-L, or drag it onto the track.
- **Pro Tools**
Choose an empty insert slot on one of your audio tracks, instrument tracks or buses and select FabFilter Pro-L from the pop-up menu in the Dynamics section.

To begin, just open the plug-in and move the Gain slider. Enjoy!

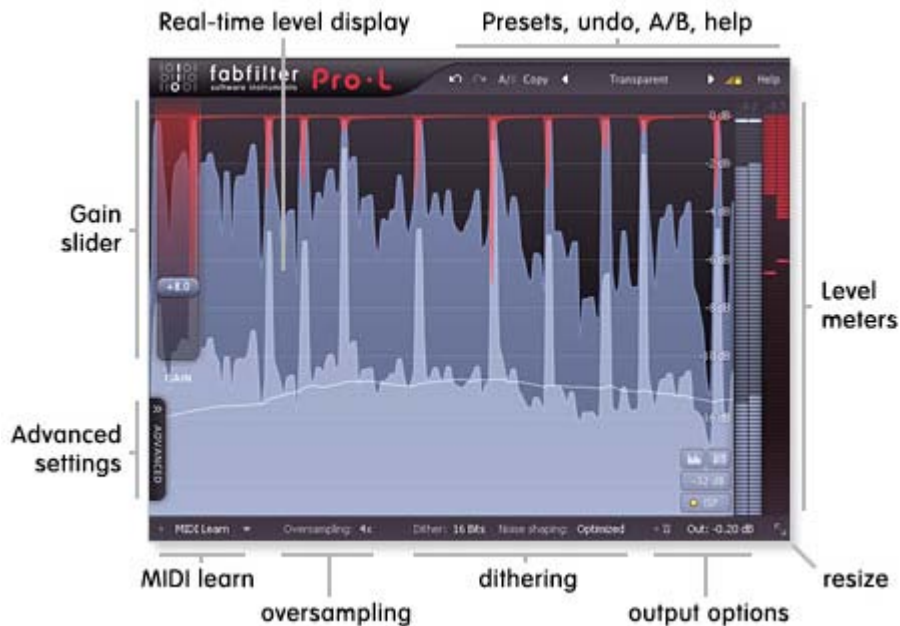
Next: [FabFilter Pro-L overview](#)

See Also

[About FabFilter Pro-L](#)

Overview

The interface of FabFilter Pro-L is designed to eliminate clutter while providing as much information as possible. It consists of the following elements:



- **Real-time level display**
The real-time moving level display shows you at a glance what's happening to your audio. It shows input (gray), output (blue) and gain reduction (red) levels together with the RMS output level (white line). See [Metering](#).
- **Gain slider**
The Gain slider is the most important control: it adds gain to the incoming signal to make it louder, while limiting to the maximum output level specified by the bottom bar. See [Knobs and gain slider](#) and [Recommended workflow](#).
- **Level meters**
The output level meter and the gain reduction meter on the right provide an immediate overview of the current output and gain reduction levels with K-system metering support, together with accumulating level readings above the meters. See [Metering](#).
- **Advanced settings**
The advanced settings panel lets you customize FabFilter Pro-L's behavior by selecting different limiting styles and tweaking their settings. See [Advanced settings](#).
- **Oversampling**
The Oversampling setting sets the amount of internal oversampling, which reduces aliasing and generation of inter-sample peaks at the cost of additional CPU usage. See [Oversampling](#).
- **Dithering and noise shaping**
The Dithering and Noise Shaping settings specify whether and how the output signal is dithered. Dithering helps to preserve (perceived) detail and resolution by reducing quantize distortion. See [Dithering and noise shaping](#).
- **Output options**
On the far right of the bottom bar, you can bypass the entire plug-in and adjust the final output level. See [Output options](#).
- **Resize**
The resize button in the lower-right corner lets you choose between normal and compact interface layouts. See [Compact view](#).
- **MIDI learn**
MIDI Learn lets you easily associate any MIDI controller with any plug-in parameter. See [MIDI Learn](#).
- **Presets, undo, A/B, help**
With the preset buttons, you can easily browse through the factory presets or save your own settings so you can re-use them in other songs. The Undo, Redo, A/B and Copy buttons at the top of the plug-in interface enable you to undo your changes and switch between different states of the plug-in. Finally, the Help menu provides access to help and version information. See [Loading presets](#) and [Undo, redo, A/B switch](#).

Next: [Knobs and gain slider](#)

See Also
[Quick start](#)

Knobs and gain slider

It is easy to control FabFilter Pro-L's parameters with the large round knobs. They will light up when you move the mouse cursor around to indicate that you can adjust them. The moment you move the mouse cursor over a knob, a parameter value display will pop up, which shows the name and the current value of the parameter.



All knobs support four ways of control:

1. Vertical mode
Click on the center area of a knob and drag up or down to rotate it. The knob reacts to the speed with which you are dragging, so if you move the mouse slowly, you make precise adjustments.
2. Rotate mode
Grab the arrow of the knob and drag it around. By moving the mouse cursor further away from the knob while dragging it, you can make precise adjustments.
3. Mouse wheel mode
Perhaps the easiest way to make adjustments is by using the mouse wheel when you hover over a knob. This mode works for all the knobs and possible panning rings. (On Windows, you might need to click in the plug-in interface first to make sure it is the active window.)
4. Text entry mode
Double-click on a knob to enter the value exactly using the keyboard.

Tips

- To reset a knob to its default position, hold down the *Ctrl* key (Windows) or *Command* key (Mac OS X) and click the knob once.
- To fine-tune a value when using vertical drag mode or the mouse wheel, hold down the *Shift* key while dragging or moving the mouse wheel.
- In text entry mode, there are several handy shortcuts. With frequency values, you can type e.g. '1k' to set the value to 1000 Hz. With dB values, you can type e.g. '2x' to get +6 dB (the value that corresponds to two times louder). With all values, you can also type a percentage (e.g. '50%' will put a knob exactly in the middle position).
- Pro Tools note: the RTAS/AS and AAX versions of our plug-ins implement the default Pro Tools keyboard shortcuts for fine tune (*Ctrl*+drag on Windows, *Command*+drag on Mac OS X) and reset to default (*Alt*+click).
- Sometimes, knobs in our plug-in interfaces are linked: these be adjusted simultaneously by holding down the *Alt* key while dragging on one of them. For example, an output level and input level setting of a plug-in could be adjusted simultaneously (in the opposite direction) this way.

Gain slider



The gain slider adjusts the threshold of limiting and the increase in output gain at the same time. Simply put, it makes your audio louder, while respecting the current maximum [output level](#). You can control the gain slider in the following ways:

- Mouse: simply click anywhere on the slider and drag the mouse up and down, or use the mouse wheel while hovering above the slider. If you click on the slider background and move the mouse slowly, you make more precise adjustments.

The Gain slider is reverse-linked to the Output Level parameter, so holding down the *Alt* key while increasing the gain, will decrease the output level with the same amount. This way, you can easily test how the limiting affects the audio without changing the apparent volume.

- Text entry: Double-click on the slider to enter the value exactly using the keyboard.

Next: [Recommended workflow](#)

See Also
[Overview](#)

Recommended workflow

To get the best results with FabFilter Pro-L, we recommend the following steps:

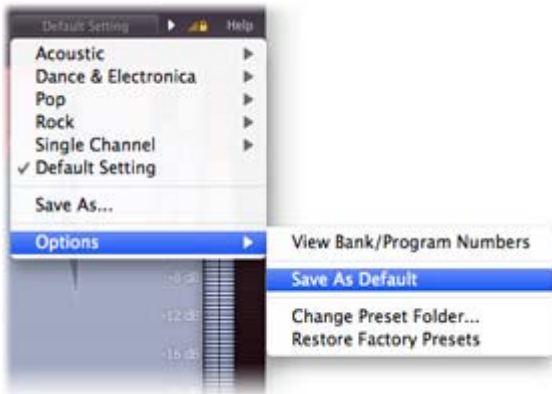
Step 1: Choose a good starting point preset

Of course, the simplest way to start working with Pro-L, is to just open it with its Default Setting preset and move the Gain slider up until you reach the desired level. We have chosen the default preset carefully, to work well on almost any audio.

Instead of just using the default settings however, you can also try one of the excellent factory presets that come with FabFilter Pro-L. They are divided per musical genre and have descriptive names, so you can easily choose the preset that works best for your purpose and audio material.



To make it easy to try and compare different presets on your audio, while leaving the current amount of limiting, the specified output level and other output settings unaffected, make sure the Lock Output button is enabled. It's placed directly right to the preset controls, at the top of the interface. When enabled, the current values of the Gain, Output Level, Oversampling, Noise Shaping and Dithering parameters will be preserved while loading presets.



After a while, you'll probably have a favorite preset and favorite settings that you use most of the time. In that case, it's a good idea to override the Default Preset so that the next time you open FabFilter Pro-L, you're ready to go right away. To save all current settings as default, simply choose *Options* > *Save As Default* from the preset menu.

Step 2: Refine settings if needed

If you like to, you can of course adjust or refine the settings that influence the sound and flavour of limiting. Just open up the Advanced panel to get access to settings like Style, Lookahead, Attack, Release and Channel Link. To learn how to interpret these settings and adjust them with sense, see [Advanced settings](#).

Step 3: Set up Oversampling, Dithering and Output level with care

The final step in setting up the limiter is choosing the correct output settings. Whether you use dithering and noise shaping depends on your preferences and requirements. See [Dithering and noise shaping](#) for more information.

"So what's an appropriate Output Level setting? Should I choose -0.1 dB, -0.2 dB or -0.3 dB?"

Actually, there is not a single correct value for the [Output Level](#) parameter. It mainly depends on the presence and strength of *inter-sample peaks* in the outgoing audio signal.

In short, our advice is simple. Turn on [ISP metering](#), which exposes these inter-sample peaks. Then adjust the level so that the level meter doesn't exceed the maximum of 0.0 dBFS. This makes it very likely that a subsequent D/A conversion (or any other conversion, like to MP3 etc.) will handle your audio without introducing distortion.

Notes

- Not everyone cares about inter-sample clipping: see [Output options](#) for the full discussion.
- Inter-sample peaks are introduced when the limiter needs to react very quickly to the incoming signal. By increasing the [Lookahead setting](#), you allow the limiter more time to respond, and therefore reduce distortion and inter-sample peaks.
- Another way to reduce inter-sample peaks is by using [oversampling](#). In most cases, choosing 4x oversampling in combination with a minimum lookahead time of 0.1 ms, keeps inter-sample peaks within the range of 0.1 dB.

Next: [Metering](#)

See Also

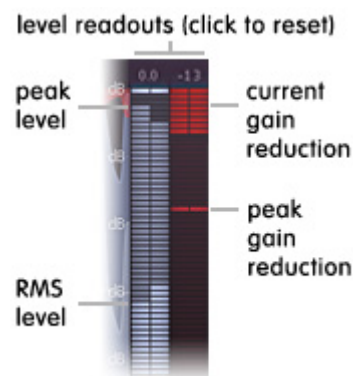
[Overview](#)

[Knobs and gain slider](#)

Metering

Accurate metering is extremely important in a limiter plug-in. To give you a perfect view of what's happening to your audio, FabFilter Pro-L offers very accurate output and gain change meters, including a textual representation of maximum peak levels, as well as a large real-time level display, showing levels and limiting over time. Using the Meter Scale button, you can choose a scale that fits your need. You can choose between three normal scales and three K-Metering scales.

Normal metering scales



FabFilter Pro-L has three normal metering scales. All three general scales have linear precision in the upper part of the metering, offering the best precision where limiting mostly happens:

- 16 dB: Showing the top 5 dB of input, output and gain reduction meters in the linear upper part of the metering, this scale offers a precise view of limiting in the top ranges.
- 32 dB: With a bit less, but still enough detail, combined with a fairly large overall range, this scale offers the best of both worlds; a good insight in the applied limiting and a proper impression of overall levels.
- 48 dB: Covering a wide range of 48 dB, using this scales gives you the best a general overview input and output levels.

Like the K-System meters described below, the normal output meter shows the RMS level and the peak level at the same time, but with a longer RMS integration time of 2000 ms. Above the meters, the maximum peak output level and gain reduction is displayed. Click on the level text to reset it.

K-System metering scales

The K-System, introduced by mastering engineer Bob Katz in 1999, is a protocol for setting mix and monitor calibrations in a studio environment. It is an attempt to standardize leveling practices throughout the audio industry. It uses three separate standards known as K-20, K-14, and K-12. With each step (from K-20 to K-12), the available dynamic range decreases as the average level increases. The top label of the meter scale indicates the maximum head-room (either 20dB, 14dB or 12dB), and just as with normal metering this matches the the full-scale digital. Your monitor gain should be calibrated carefully, so that the level at the 0 dB label of the meter matches 83 dBC.

The K-System meters show both peak and RMS level at the same time. The top red zone of the meters is the *loud* or *fortissimo* zone. In music recording, the RMS level should only reach the red zone in the loudest passages, climaxes or occasional peak moments. If you find yourself using the red zone all the time, you might want to check whether your monitor gain is properly calibrated.

- K-12: This scale is intended to be used exclusively for broadcast material, be it radio or television. With this system, -12dBFS = 0VU = 85dB SPL. The limited headroom of 12dB explains its exclusiveness to heavily compressed broadcast material.
- K-14: This should be the standard for the majority of commercial recordings created for home listening. Pop music and home theatre mixes are examples of material that would fall under K-14, where -14dBFS = 0VU = 85dB SPL. The available headroom is 14dB. The K-14 scale is probably the most widely used of the three standards.
- K-20: Offering the widest available dynamic range of the three systems, this scale should be used primarily for large theatrical mixes, dynamic music mixes, and Classical style mixes. Any material with a wide dynamic range should be reserved for the K-20 standard. In this system, -20dBFS = 0VU = 85db SPL. As you might guess, 20dB of headroom is available.


To read more about the K-System and how to use it properly (including monitor gain calibration), read Bob Katz' article: [An Integrated Approach to Metering, Monitoring, and Levelling Practices](#).

The real-time level display

The large real-time level display shows input level (grey), output level (light blue), gain reduction (red) and RMS level (white line) at the same time. It gives you a very good insight in the amount of limiting going on, and the overall peak and average levels.

In case you don't want to be distracted by the display, you can simply disable it using the Show Display button, right under the meter scale select button, left of the level meters. The Show Meters button next to it disables the level meters so you can turn off all visual feedback.

ISP (Inter-Sample Peak detection)



Digital audio processing, and especially ultra-fast limiting or hard clipping in the digital domain, can introduce harmonic frequencies that can't be expressed properly with the sample rate you're using. Still, a D/A converter needs to interpret that signal and translate it to an analog wave form. At some points, especially at sharp transients as a result of limiting/clipping, the resulting wave form that is constructed out of the samples, can have peaks that are higher than the peaks in your original digital signal. The quality of your D/A converter will determine how these peaks are handled, and how they affect the sound.

Of course, it's always best to minimize inter-sample peaks, and at the same time ensure that any peaks in your audio, both normal or inter-sample, stay within the 0 dBFS range. Then you can be sure that D/A conversion (and also other operations like MP3 conversions), do not introduce unwanted distortion.

Using the ISP button, you can enable inter-sample peak detection. Pro-L's output level meter will then show the inter-sample peaks in the resulting audio, so you can correct the [output level](#) for them. The level readout above the output meter will show exactly how much more headroom is needed. After adjusting the output level, click on the level text to reset it.

Normally, using Pro-L's [oversampling](#) (preferably 4x) in combination with a minimum [lookahead time](#) of 0.1 ms (which is still *very* fast), brings down inter-sample peaks to a range of only about 0.1 dB. However, the ISP option will show you exactly what's going on, so you don't have to guess.

For more information on inter-sample peaks, we recommend this article: [Digital Distortion in CDs and DVDs](#) .

Tips:

- After working with Pro-L for a while, you probably have your own favorite metering settings. You can simply save all current parameter and interface settings of the plug-in as the default startup settings, by choosing *Options > Save As Default* from the plug-in's preset menu.
- You can also switch to the Compact interface layout to hide the real-time display and see larger output meters. See [Compact view](#).

Next: [Advanced settings](#)

See Also

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Advanced settings

Most of the time, choosing the right preset from FabFilter Pro-L's preset menu works very well. But in some cases, you might want to adjust and fine-tune the limiting behavior. To access Pro-L's advanced settings, click the Advanced button right under the [Gain slider](#) at the left of the interface, which makes the advanced panel slide out. Clicking the button again will hide the panel.



Style

FabFilter Pro-L comes with four advanced and highly program-dependent limiting algorithms, which all have their own distinct character. One is not better than the other; they are all great, and they all go very loud if you want them to.

It is best to select the appropriate algorithm depending on what kind of effect you want to achieve: some are designed to be as transparent as possible, while others may add a nice punch or flavor. Here's a short description of all algorithms:

- **Transparent:** As the name says, this algorithm is designed to stay true to the original sound and feel as much as possible, avoiding pumping effects and coloring. It works great on most program material, but especially on rock and pop music. It performs best with a short Lookahead (< 0.5 ms), an Attack between 50 and 200 ms, and a Release between 200 and 1000 ms.
- **Punchy:** Of all algorithms, this one is the most apparent. When pushed, it becomes quite punchy and will introduce a bit of pumping, which can add some nice flavor. Since it's fairly 'safe', minimizing distortion most of the time, it works miracles on single tracks, like vocals, bass or guitar and can give a nice edge to beat-oriented music. It performs best with a normal Lookahead (> 1.0 ms), an Attack around 250 ms and Release around 500 ms.
- **Dynamic:** By enhancing transients before actually applying limiting, this algorithm excels in preserving the original punch and clarity of your audio. It's probably the best algorithm for rock music, but it can work surprisingly well on other types of audio as well. The best starting point is to choose a very short Lookahead time (< 0.5 ms) and set the Attack knob and Release knob half way, and adjust them from there. Note however that this algorithm uses more CPU power than the other algorithms, and also introduces more latency.
- **Allround:** This algorithm is designed to be very 'safe', minimizing distortion while still going as loud as possible. It can work well on almost any program material! Try starting with a fast Lookahead (around 0.5 ms), an Attack around 250 ms and Release around 500 ms.

Lookahead

The Lookahead knob sets the look-ahead time for the initial 'transient' stage. This allows the limiter to examine the incoming audio in advance and predict the amount of gain reduction needed to meet the requested [output level](#). If the look-ahead time is very short, the limiter doesn't have much time to move to the desired level: this will generally have the effect of preserving transients better and increasing the apparent loudness, but at the expense of possible distortion. Longer look-ahead times are safer, but less loud.

Very short look-ahead times (less than 0.1 ms) will approximate 'hard clipping', introducing distortion and aliasing. This causes inter-sample peaks which can cause further distortion later on. To reduce aliasing and inter-sample peaks, we advice to use oversampling. Also, turn on [ISP metering](#) to visualize the inter-sample peaks generated. This enables you to adjust the output level accordingly.

Attack and Release

Apart from the fast 'transient' stage, the limiter has a slower 'release' stage that responds to the overall dynamics of the incoming audio. The Attack and Release knobs control how quickly and heavily the release stage sets in. Shorter attack times will allow the release stage to set in sooner; longer release times will cause it to have more effect.

In general, short attack times and long release times are safer and cleaner, but they can also cause pumping and reduce clarity. On the other hand, long attack times and short release times can increase apparent loudness and presence, but at the expense of possible distortion.

Channel linking

When limiting a stereo signal, it is generally desirable to process both channels in the same way to avoid changing the stereo image inadvertently. However, when a short peak occurs in one channel, removing it is often almost inaudible. In this case, it is better to remove it only in the channel where it occurs.

You can control this behavior completely with the two channel linking knobs. The Transient knob controls the amount of channel linking for the 'transient' stage that mainly operates on short peaks. It often works well to

choose less than 100% here. The Release knob controls the channel linking for the 'release' stage, where it is best to start with 100% which will completely link the channels. However, you can of course experiment with different settings depending on the level of limiting and the character of the incoming audio signal.

Tips:

- When you first start to work with FabFilter Pro-L, try to use the factory presets. These are great starting points, smartly divided in music categories, and with descriptive names. These might just do the trick already, without even needing to open the Advanced Panel.
 - If you have found a preset that you really like, which works well on most music you usually process, you can easily save it as the default preset! Just choose Options > Save As Default from the plug-ins preset menu, and the next time, Pro-L will start-up with your favorite settings.
-

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Oversampling

The limiting algorithm often needs to make very quick changes to the audio in order to remove peaks while preserving transparency and apparent volume. These sudden changes can introduce aliasing, which causes distortion and generally reduces the quality of the audio signal. Oversampling is a way to reduce that aliasing by running the internal limiting process at a sample rate that is two or four times higher than the host's sample rate.



Oversampling: 4x

"When do I need to turn on oversampling?"

You need it more when the limiting process operates faster (using short [lookahead](#) times), and when limiting more heavily, both leading to a higher level of aliasing. The aliasing will cause inter-sample peaks and these can cause distortion later on, for example during D/A conversion or conversion to mp3. There are only two small drawbacks to oversampling: it increases CPU usage, and it can introduce a very slight pre-ring due to the phase-linear filtering that is needed. Generally this effect is so small that it's inaudible, but it's good to be aware of this and not blindly assume that oversampling is always better.

"Why can my output level exceed the specified Output Level setting when oversampling is enabled?"

When using oversampling, limiting is applied to the upsampled audio (two or four times the normal sample rate), ensuring that no sample value in the upsampled result will exceed the specified Output Level. However, even though most aliasing is filtered out during the final downsampling stage, still some inter-sample peaks may exist. Because of these peaks, the downsampling process which reconstructs the audio in the original sample rate, can generate waveforms with a slightly higher level than the specified Output Level. The amount of that overshoot highly depends on the speed and amount of limiting. In most cases, using a minimum lookahead time of 0.1 ms keeps the overshoot within the range of 0.1 dB.

Using ISP metering and oversampling

As we just explained, oversampling might already reveal the presence of inter-sample peaks in the resulting audio. But to fully expose them, you should use [ISP metering](#). This also clearly visualizes the beneficial effects of oversampling. Using 4x oversampling will dramatically reduce the inter-sample peaks, which in turn allows you to increase the desired [output level](#) without inter-sample clipping. You can also clearly see that if you use a slightly longer look-ahead time, there will be less inter-sample peaks, both with and without oversampling.

Notes

- When you use very short look-ahead times of less than 0.1 ms (clipping), it may become difficult to keep inter-sample peaks within a workable range. Again, in most cases, choosing 4x oversampling in combination with a minimum lookahead time of 0.1 ms, keeps inter-sample peaks within a small range of 0.1 dB.

Next: [Dithering and noise shaping](#)

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Dithering and noise shaping

In modern music production, most people are used to working with 24-bit audio to preserve as much resolution and precision as possible. Plug-ins usually work with 32-bit or 64-bit floating-point sample values. However, most audio finally ends up on a normal CD that only uses 16 bits of resolution. This means that at some point, the bit depth has to be reduced to suit the final medium.

The simplest way to reduce the bit depth of an audio signal, is to just truncate the least significant bits of every sample. However, this causes quantization distortion (in the form of unwanted harmonics) in the resulting audio. The best way to avoid this distortion, is by adding a tiny bit of white noise to the audio signal before truncating any bits. This eliminates the nasty quantization distortion at the cost of a slightly higher noise level in the final audio. This is called dithering.

To make the effect of applying dithering noise less audible in the final audio (in other words: to improve the signal-to-noise ratio), we can filter the noise introduced by the dithering process. That way, we don't end up with plain white noise (having a flat spectrum), but with noise that is less audible at frequencies where the human ear is most sensitive.

Dithering in FabFilter Pro-L

In the bottom bar of FabFilter Pro-L's interface, you'll find the dithering settings. With two simple controls, you can specify your preferred dithering and noise shaping settings:

- The Dithering parameter specifies the desired bit depth of the resulting audio. You can choose to dither/quantize to 24, 22, 20, 18 and 16 bits. Of course, if you don't want any dithering, quantization (or noise shaping) to occur, just choose 'off'.
- The Noise Shaping setting lets you choose between various noise shaping algorithms:
 - The Basic setting lowers the overall noise floor a few dB, at the cost of increasing noise levels for frequencies above 6 kHz.
 - With the Optimized setting, the effect is more extreme; you'll get an even lower overall noise level, but noise frequencies above 10 kHz are boosted more extremely.
 - Weighted noise shaping will transform the noise spectrum according to the ear's sensitivity to certain frequencies at low listening levels. Theoretically, this results in the lowest audible noise. This noise shaping setting is designed to be used at 44.1 kHz. It still works at other sample rates, but the frequency spectrum of the resulting noise isn't optimal anymore.

Myths and facts

Theoretically, dithering the best way to retain as much resolution as possible when quantizing your audio. However, in the real world, dithering often has little to no audible effect. Here are a few things to keep in mind:

- Most of today's music is mastered at quite loud (if not ridiculously loud) average levels, leaving very little dynamics in the final result. This already masks the small level of distortion due to quantization, so dithering probably won't have any audible effect.
- A lot of audio recordings already have a relatively high noise floor, due to the use of microphones, amplifiers, analog outboard, mixing consoles etc. In that case, dithering will have no beneficial effect at all; it will just increase the existing noise floor!
- Dithering should *only* be used as the final stage of audio processing/mastering. With any further processing, like gain changes, applying effects, or converting to yet another bit depth, the effects of dithering will be lost. If your host offers a post-gain effect insert slot on the master channel, use this slot for FabFilter Pro-L when dithering is enabled.
- Dithering more than once doesn't make any sense. It will just increase the overall noise level in your audio.

So when should you use Pro-L's dithering? The rule of thumb would be: when you use FabFilter Pro-L in the final stage of mastering, handling audio with a very low noise floor of itself, and the end result is still fairly dynamic. But the most important advice of all is.... use your ears!

Notes

- The white noise used for dithering in FabFilter Pro-L is the industry-standard TPDF noise, 2-bit peak-to-peak.
- To read more on dithering, see [Dithering](#) and [Noise shaping](#) on Wikipedia.

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[Recommended workflow](#)

Output options

At the righthand side of the bottom bar in the interface, FabFilter Pro-L contains the output level and bypass parameters.



The Output Level knob plays an important role: it sets the desired maximum output level for the limiter. It seems logical to set it to 0.00 dBFS: you want the output to be as loud as possible, right? Not so fast.

Due to its ultra-fast behavior, the limiting process can generate inter-sample peaks: while none of the outgoing sample values are higher than 0 dBFS, the analog wave form that will be constructed out of the samples by the D/A converter can actually exceed this by several dB. Actually, any conversion that reinterprets the wave form, especially MP3 conversion, can expose inter-sample peaks. This will lead to unpredictable clipping and therefore possibly audible distortion.

"How bad is clipping caused by inter-sample peaks?"

This is very hard to tell: it depends on many things, such as the quality of the D/A converter and the character of the music. Many professionally mastered albums contain inter-sample clipping and this doesn't have to be a problem. The main effect is that the music may suffer from slight distortion when played by low-quality D/A converters, or when converted to MP3 format.

"So what do you recommend?"

We recommend to turn on [ISP metering](#) to visualize the inter-sample peaks so you are at least aware of them. If you want to ensure that there won't be any inter-sample clipping during subsequent conversions, you need to adjust the output level to keep inter-sample peaks under 0 dBFS.

You may find that the inter-sample peaks are very high and that you need to reduce the output level too much to avoid clipping. In this case, there are two ways to reduce the generated inter-sample peaks: turning on [oversampling](#), and slightly increasing the [lookahead setting](#) in the Advanced panel. In most cases, choosing 4x oversampling in combination with a minimum lookahead time of 0.1 ms, keeps inter-sample peaks within the range of 0.1 dB, so the corresponding output level setting would be -0.10 dB.

You can bypass the entire plug-in with the Global Bypass toggle button to the left of the output level button. While most hosts already provide the ability to bypass plug-ins, our internal global bypass feature is guaranteed to correctly compensate the latency of the plug-in and it also applies soft bypassing to avoid clicks. While the plug-in is bypassed, the display dims and a red light glows in the bypass button itself.

Notes

- When loading presets, enable the [Lock Output](#) option next to the presets button to preserve the current gain and output settings. All factory presets were saved with a default output level of 0.0 dB.
- You can directly adjust the output gain by clicking and dragging the output button vertically, so there is no need to click it first to view the output knobs. You can also double-click the output button to directly enter a value using the keyboard.
- The resize button next to the Out setting lets you switch the interface to [Compact view](#).

Next: [Compact view](#)

See Also

[Overview](#)

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Compact view

Some people like to judge their limiting by ear only. For that purpose, FabFilter Pro-L offers a 'Compact' view mode. You can switch between different views with the resize button in the lower-right corner. Most hosts support dynamic resizing of the interface; otherwise just close and re-open the interface window.

The Compact view doesn't have the animated level display, but its level meters have more detail than the Normal view and it offers a large Output knob.



In Compact view, you can adjust the gain and the output level also by clicking and dragging on the level readouts below the Gain and Output knobs. Like with any knob, double-click here to type a value directly.

Next: [MIDI learn](#)

See Also

[Overview](#)

[Recommended workflow](#)

MIDI Learn

Controlling FabFilter Pro-L's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.



Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

1. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
2. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- **Disable/Enable MIDI**
This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- **Clear**
This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- **Revert**
Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- **Save**
Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

Next: [Undo, redo, A/B switch](#)

See Also
[Overview](#)

Undo, redo, A/B switch

The Undo and Redo buttons at the top of the FabFilter Pro-L interface enable you to easily undo changes you made to the plug-in. With the A/B feature, you can quickly switch between two different states of the plug-in.



- The Undo button at the left will undo the last change. Every change to the plug-in (such as dragging a knob or selecting a new preset) creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last undo command. It steps forward through the history until you are back at the most recent plug-in state.
- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B). In the example above, the A state is active.
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Notes

- If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.
- The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

Next: [Loading presets](#)

See Also
[Overview](#)

Loading presets

FabFilter Pro-L comes with a selection of excellent factory presets that provide ready-to-go limiting setups for almost all scenarios and types of audio.

- To load a preset, click the preset button. The presets menu will appear with all available presets. Click a menu item to load that preset. The currently selected preset is highlighted with check marks.
- To explore the presets one by one, click on the little arrow buttons to the left and right of the main preset button. This will load the previous or next preset in the menu.

The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

Tips

- The Default Setting preset is loaded automatically when FabFilter Pro-L is started. To change the default settings, simply overwrite this preset by clicking Options > Save As Default in the presets menu.
- If somehow the factory presets are lost or not installed properly, click Options > Restore Factory Presets in the preset menu to restore them.

MIDI Program Change and Bank Select

Loading a presets can also be done via MIDI, using Bank Select and Program Change messages. Click Options > View Bank/Program Numbers in the preset menu to enable or disable the display of bank/program numbers (for example: *(2/65) My Preset*). This means that you can load that preset by first sending a Bank Select message to select bank 2 and then sending a Program Change message to select program 65.

Important: All the presets in your preset folder are numbered automatically, starting with bank 0 and program 0. This way, you are able to access any of the presets via MIDI. However, this also means that when you add new presets to the menu, bank/program numbers of other presets might change. Be aware of this when recording program changes in a song!

Next: [Saving presets](#)

See Also

[Overview](#)

[Recommended workflow](#)

Saving presets

You can easily extend the included presets with new settings to build your own library of presets for FabFilter Pro-L that you can reuse in various projects. This is also a good way to copy settings across multiple instances of FabFilter Pro-L in a session.

To save the current setting as a preset, click the preset button, and then click *Save As*. A standard Save dialog will appear. Type a name for the new preset and click *Save* to finish.

In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu. (On Mac OS X, this should be done with the Finder.)

See [How presets are stored](#) to determine the preset folder location and learn more about factory presets.

Next: [How presets are stored](#)

See Also

[Loading presets](#)

[Overview](#)

How presets are stored

Presets for FabFilter Pro-L are stored in separate files with the .ffp extension (for FabFilter Preset). All presets reside in subfolders in the main preset folder. The subfolders will show up as separate categories in the preset menu. You can also further divide the subfolders into categories.

You can manage the preset files just like other files on your computer. The easiest way to do this is in the Save dialog that appears if you are saving a preset. The preset menu will automatically reload itself with the changes when the dialog is closed.

Furthermore it is very easy to share your newly created presets with other users since FabFilter presets use the same file format on both Windows and Mac OS X.

The default location of the main preset folder is *My Documents\FabFilter\Pro-L* for Windows, and *~/Library/Audio/Presets/FabFilter/FabFilter Pro-L* for Mac OS X. To change this location, first copy all presets to the desired new location, and then click Options > Change Preset Folder in the preset menu and select the new folder.

Restoring factory presets

If you have accidentally lost the factory presets, you can easily restore them by clicking Options > Restore Factory Presets in the preset menu. This will install all factory presets again.

Next: [Purchasing FabFilter Pro-L](#)

See Also

[Saving presets](#)

[Overview](#)

Purchasing FabFilter Pro-L

Once you have downloaded and installed the evaluation copy of the FabFilter Pro-L, you may evaluate it during 30 days. Every time you start the plug-in, you will see the following dialog:




While there are still days left, you can click Evaluate to start working with the plugin. If you want to keep using FabFilter Pro-L after the evaluation period, you must buy a copy in the online FabFilter Shop by clicking the Buy Now button in the evaluation dialog.

- [Go to the FabFilter Shop and purchase FabFilter Pro-L now](#) 

We accept credit cards from all major companies, check payments, wire payments, and PayPal. The FabFilter Shop uses secure connections and encryption: therefore your personal information is completely safe.

Within a few minutes after you have purchased your copy, you will receive an e-mail containing your personal license key. You use this license key to turn the evaluation copy into a fully registered version without the evaluation dialog and the 30-day trial restriction.

Note: If the evaluation period has expired but you didn't have the chance to properly evaluate the plug-in, you can request a new evaluation period by contacting us at info@fabfilter.com .

Next: [Entering your license key](#)

See Also

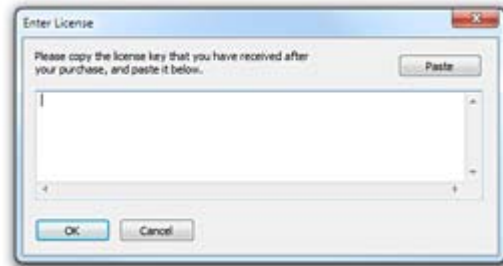
[Support](#)

[License agreement](#)

Entering your license key

After you have purchased FabFilter Pro-L in the online [FabFilter Shop](#), you will immediately receive an e-mail containing your personal license key. This license key will turn the evaluation version into a fully registered plug-in.

- Start FabFilter Pro-L and click Enter License in the evaluation dialog, or click Enter License on the Help menu if the plug-in is already running.
- Copy the license information from the e-mail you have received and paste it into the text field. Make sure that you are copying the entire license key including the *Product* and *Licensee* lines! If you are not sure what text to copy, just copy and paste the whole e-mail.



After you have entered your license information, you will need to restart the plug-in host, so make sure you save your settings if needed. From now on, you will be able to use FabFilter Pro-L for an unlimited period of time with full support via e-mail.

Tips

- After your purchase, you can always retrieve your license key again by logging into your [personal FabFilter account](#). Here you can also keep track of all your orders and take advantage of great discounts when buying other FabFilter plug-ins!
- To deauthorize your license key and remove it from the computer, click Deauthorize on the Help menu. If you temporarily needed to install your license on another computer, or if you have transferred your license to someone else, this is the way to deinstall your personal license key safely.

Next: [Support](#)

See Also

[Purchasing FabFilter Pro-L](#)
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[License agreement](#)

Support


If you need help with problems or questions, and the help file does not provide an answer, please visit the support pages on our web site.

- [Go to FabFilter Support](#) 

From here, you have direct access to the customer support forum, very useful tutorial videos for all FabFilter plug-ins, online and PDF versions of all our help files, and a section with extra downloads (such as presets, controller templates, older plug-in versions).

For sales questions and technical support, you can also contact FabFilter directly at info@fabfilter.com.

Reporting a bug

If you have encountered a bug in FabFilter Pro-L, first of all make sure that you are using the latest version of the plug-in, which you can find at www.fabfilter.com/download . You can check the version of your plug-in by opening an instance in your host, open the plug-ins help menu via the Help button at the right top of the interface, and choose About. If the bug is still present in the latest version, please send us an e-mail at info@fabfilter.com and include as much technical information as possible: operation system and version, host software and version, steps to reproduce the bug, etc. Thanks in advance!

Next: [Manual installation](#)

See Also

[Overview](#)

[About FabFilter](#)

Manual installation

When installing FabFilter Pro-L, the installation program will try to copy the plug-in into the appropriate plug-ins folders, and in most cases you host will recognize FabFilter Pro-L automatically. Otherwise, please follow these instructions:

Windows


On Windows, most hosts have their own VST plug-ins folder. So if you are using Windows and your host does not recognize FabFilter Pro-L, you need to locate the proper plug-ins folder for your host first (it is usually shown in the Preferences or similar dialog). Then, copy the file *FabFilter Pro-L.dll* from *C:\Program Files\FabFilter\Pro-L* (or *C:\Program Files (x86)\FabFilter\Pro-L* if you are using a 32-bit host on 64-bit Windows) to the plug-ins folder that you have found and restart the host so it can reload all its plug-ins. If you are using the RTAS or AAX version of Pro-L, copy the *FabFilter Pro-L.dpm* or *FabFilter Pro-L.aaxplugin* -and- *FabFilter Pro-L.dpm.rsr* files to the Pro Tools plug-in directory on your computer.

OS X

On OS X, plug-ins are installed in the standard plug-in folders in the system Library folder. These are the *only* possible correct locations:

- Audio Units: */Library/Audio/Plug-Ins/Components*
- VST/VST3: */Library/Audio/Plug-Ins/VST* and */Library/Audio/Plug-Ins/VST3*
- RTAS: */Library/Application Support/Digidesign/Plug-Ins*
- AAX: */Library/Application Support/Avid/Audio/Plug-Ins*

Note: AU and VST/VST3 plug-in may also be placed in the user's Library folders under *~/Library/Audio/Plug-Ins* as well.

If you still have problems, contact [FabFilter Support](#) .

Next: [VST plug-in versions](#)

See Also

[Quick start](#)
[Support](#)

VST plug-in versions

FabFilter Pro-L is available in both VST 2 and VST 3 formats. They can be installed and used both at the same time. The VST 3 format can only be used by hosts that support it, for example Steinberg's Cubase 5. At this moment, the VST 2 format is compatible with a larger variety of hosts.

VST 2

For the VST 2 format, there are two versions of the FabFilter Pro-L plug-in, a mono and a stereo one:

- *FabFilter Pro-L*: Stereo version of the plug-in
- *FabFilter Pro-L (Mono)*: Mono version of the plug-in

Note: The VST 3 version of the plug-in automatically adapts itself to both mono and stereo tracks.

Next: [License agreement](#)

See Also

[Quick start](#)

[Manual installation](#)

FabFilter Software License Agreement

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3. VST plugin technology

VST is a trademark of Steinberg Soft- und Hardware GmbH.

Next: [Acknowledgements](#)

See Also

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[Support](#)

Acknowledgements

For the development of FabFilter Pro-L's internal limiting algorithms, we worked closely together with Niklas Silen (AKA bManic). Niklas developed the Transparent and Dynamic limiting styles and was very influential in the general design of Pro-L. His never fading enthusiasm and persistence is inspiring. Thanks!

Furthermore, of course we want to thank all our beta testers, helping us to fine-tune the limiter.

FabFilter Pro-L comes with a selection of ready-made presets, most of which were created by external engineers:

- bManic (bM)
- Mark den Hartog / Zownd (MdH)
- Motoko (MTK)

Visit the [FabFilter sound designers page](#)  to read more about each sound designer.

Next: [About FabFilter](#)

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About FabFilter

FabFilter Software Instruments was founded in 2002 by Frederik Slijkerman MSc and Floris Klinkert MSc, and is based in the center of Amsterdam in the Netherlands. In their roles as professional musicians and studio engineers, Floris Klinkert and Frederik Slijkerman have always loved the sound of old analog gear but also kept up with new software synthesis developments. However, none of the available software effects and instruments met their high quality standards.

Many plug-in developers try to emulate old analog gear, or stuff their plug-ins with as many features and (often mediocre) effects as possible, diverting you from the real issue: the sound. Both also being professional software engineers, Frederik Slijkerman and Floris Klinkert decided to do things differently and create the audio effects and instruments they dreamed of, resulting in the release of FabFilter One in early 2004, followed by FabFilter Volcano, Twin, Timeless, Simplon, Pro-C, Volcano 2, Twin 2, Timeless 2, Pro-Q, Micro, Pro-L and Pro-G.

To accomplish their goals, a lot of effort has been put into developing highly advanced filtering, EQ'ing and dynamics algorithms, superb digital oscillators, and highly innovative user interface solutions. The highly optimized FabFilter plug-in design results in very little CPU requirements. FabFilter strongly believes that musicians don't just need more knobs, faders or options on plug-ins: they need better sound quality and usability. That is why FabFilter creates easy-to-use yet powerful plug-ins, with innovative interfaces and superb sound quality, for a reasonable price.

FabFilter will expand their range of plug-ins in the future, supporting both Mac OS X and Windows, providing the best possible sound quality and filter designs. We hope that you will enjoy our products. Please feel free to contact us if you have any suggestions!

The FabFilter team

See Also

[Support](#)

[FabFilter web site](#) 