Wave Arts FinalPlug 7

User Manual

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1.0 Introduction

FinalPlug combines a lookahead peak limiter with ITU BS.1770 loudness metering and bit depth truncation and noise shaped dithering. FinalPlug can be used as an insert effect, providing limiting for gain staging or volume maximization, or can be placed at the end of your mastering chain to guarantee your audio meets loudness and true peak delivery requirements. Here are some of FinalPlug's key features:

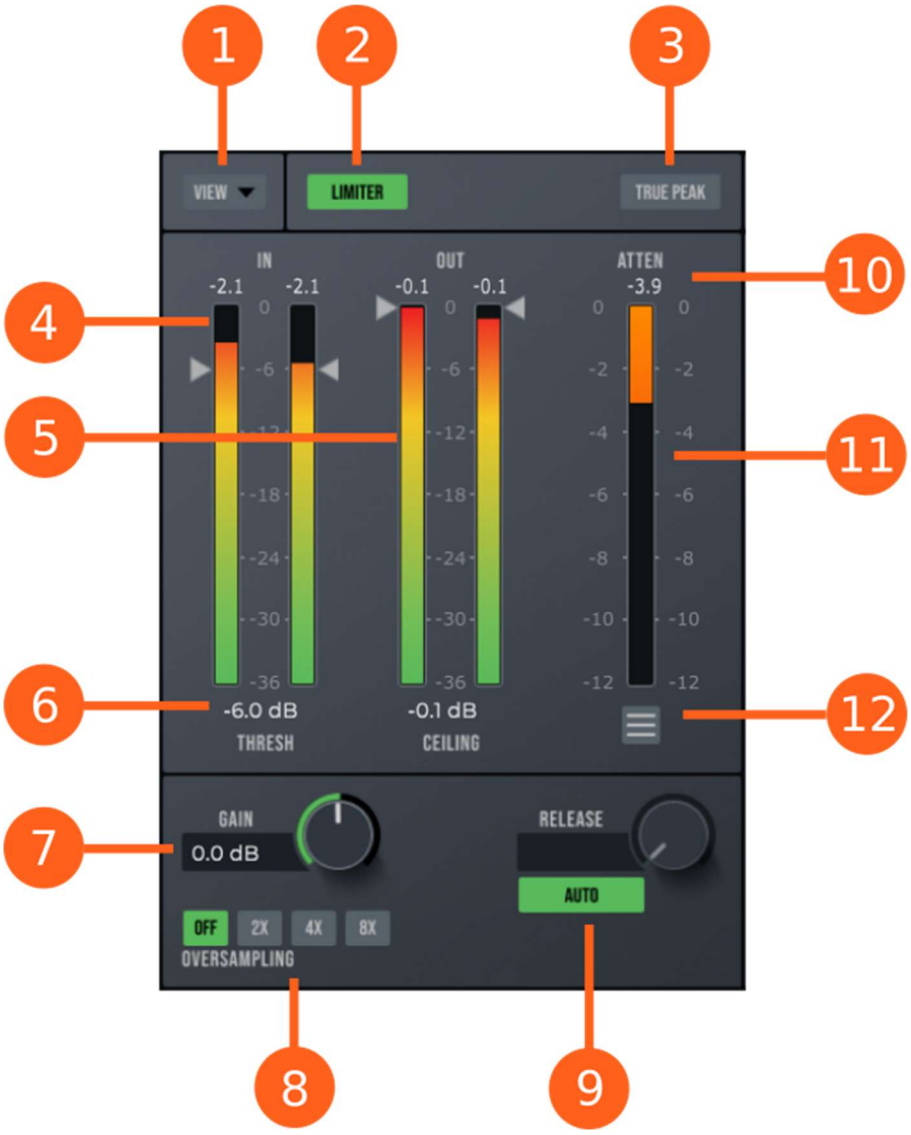
- Sonically transparent peak limiting, for use as peak limiter or volume maximizer.
- True peak limiting mode.
- Auto-release mode.
- Limiter oversampling up to 8x.
- ITU BS.1770 loudness metering.
- Writes loudness and overflow state as automation parameters.
- MATCH button automatically configures the limiter to meet loudness and true peak delivery targets.
- Preset menu of common industry delivery targets, or make your own.
- Bit depth truncation from 4 to 24 bits.
- TPDF dithering.
- Comprehensive noise shaping options.
- Customize the UI to show only the features you need.

2.0 User Interface

2.1 Overview

Label	Description
1	Menu bar: Bypass, Undo/Redo, Presets, A/B buffers, and Settings.
2	Peak limiter processor.
3	Loudness metering and delivery targets.
4	Bit quantization and dithering.
5	View popup menu.
6	Information bar showing help tips.

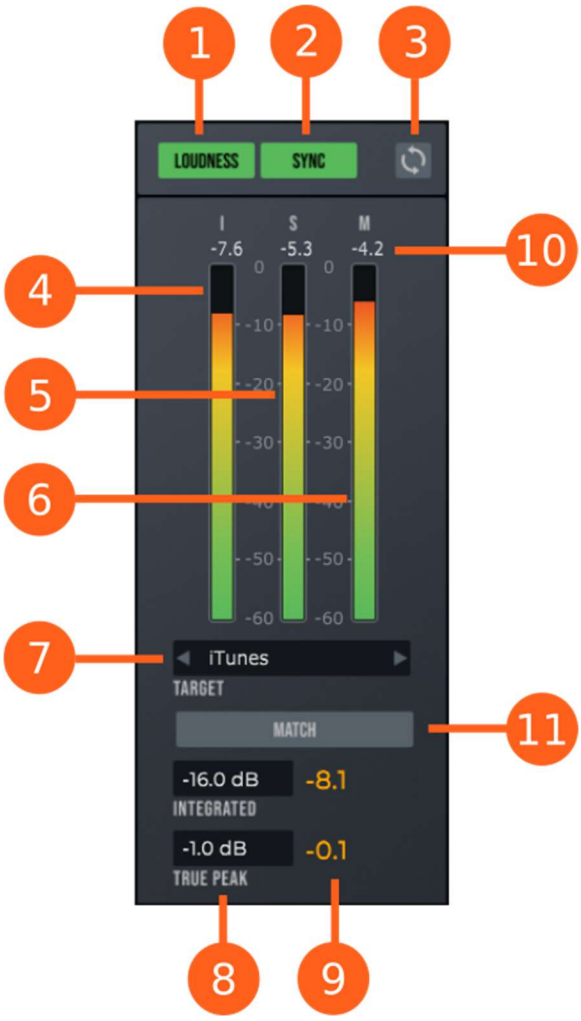
2.2 Peak Limiter



Label	Description
1	Opens View popup menu where you can show/hide the loudness meters, dithering, and info bar. The limiter is always displayed.
2	Enable/bypass the peak limiter.
3	Enable True Peak limiting. When enabled the limiter responds to true peak levels and input and output meters show true peak levels.

4	Input meters and threshold slider. Input levels above threshold are limited. Lower the threshold to increase gain and compression and loudness.
5	Output meters and ceiling slider. Ceiling sets the maximum peak limited output level.
6	Threshold and ceiling levels. You can click on these and type in values. Press TAB to advance, type ESC or click on unused UI area to exit text entry mode.
7	Input gain control. The gain is applied before the input meter.
8	Oversampling selection, affects only the limiter section.
9	Release control sets the release time of the limiter. AUTO mode sets the release time dynamically based on the crest factor of the signal.
10	Peak hold displays for input, output, and attenuation. Click to reset, SHIFT-click to reset all.
11	Attenuation meter.
12	Popup menu of input/output, attenuation, and loudness meter ranges.

2.3 Loudness Metering



Label	Description
1	Enable/bypass the loudness meters.
2	Synchronize to DAW transport. When enabled the loudness meters will automatically reset when the transport rewinds or fast-forwards.
3	Resets the loudness meters to initial state.
4	Integrated Loudness meter.
5	Short-term Loudness meter.

6	Momentary Loudness meter.
7	Target preset menu with common industry delivery targets, and options to save your own presets.
8	Integrated Loudness and True Peak target levels, set by preset menu, or manually entered by clicking on fields and typing.
9	Current Integrated Loudness and True Peak levels. These are colored orange when they exceed the target levels.
10	Integrated, Short-Term, and Momentary Loudness peak levels. Click to reset, SHIFT-click to reset all.
11	Match button sets the limiter parameters to match the current target levels, see below.

To meet an audio delivery standard, do the following:

1. Disable the limiter section, or select the "Analyze" master preset.
2. Select a delivery target preset from the target preset menu. This sets the Integrated Loudness and True Peak target levels.
3. Play your session entirely to capture the current loudness and true peak levels. If the loudness or true peak levels exceed the targets they are displayed in orange.
4. Click MATCH. This sets the limiter parameters to meet the targets.
5. Play your session again and the loudness and true peak levels should not be exceeded.

*Tips: the algorithm matches the **final** IL level at the end of playback; during playback the IL may briefly exceed the target level.*

Tips: if your audio does not require peak limiting, the final true peak level may end up below the target; this is correct and expected behavior.

Tips: if the limiter is very active, the final IL level may not exactly hit the target. Applying MATCH again and doing another pass should fix.

Tips: you can set error margins for the target levels in the preferences, by default these are set to 0.05 dB.

2.4 Dither and Bit Quantization

Label	Description
1	Enable/bypass the dither processor.
2	Mute the dither noise, applying only bit quantization.
3	Plot of the noise shape response, showing how quantization noise energy is redistributed.
4	Sets the number of bits to quantize to.
5	Sets the noise shape filter.

3. Detailed Description of FinalPlug

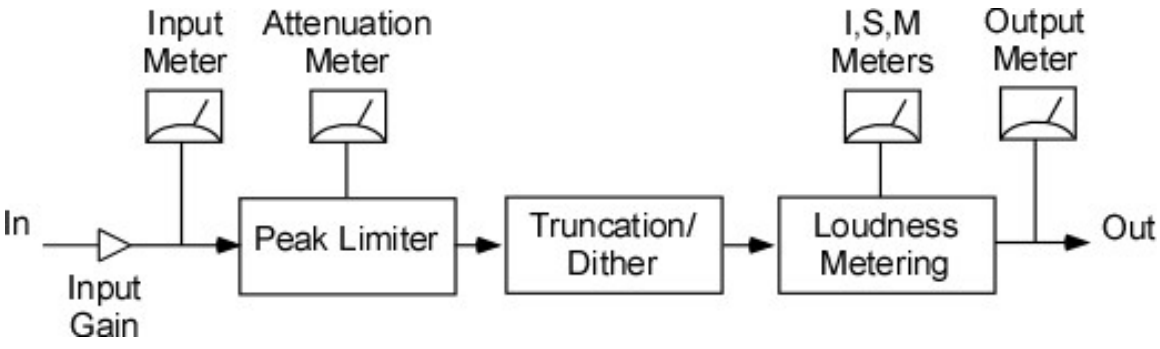
3.1 Overview

FinalPlug is intended as the final plug-in to process a mix on the master output. It allows you to meet audio delivery standards based upon target Integrated Loudness and True Peak levels, as specified by the ITU BS.1770 loudness standard. As described in the section 2.3, this requires two passes: a first playback to capture the loudness and true peak levels of the mix, and a second pass to apply the proper gain and peak limiter parameters to meet the targets.

FinalPlug can also be used as an insert effect for simple peak limiting. One use is for loudness maximization, where the signal gain is increased, thus increasing the loudness of the signal, while keeping peaks below maximum. The use of a peak limiter to set the loudness is more flexible than normalizing, as normalization applies the same gain to the entire signal, hence the gain applied is limited by the peak value. The FinalPlug limiter also applies gain, but ducks the gain around peaks in the signal, so you can push up the signal level while keeping peaks below maximum. With careful use, you can increase the perceived loudness of a typical mix by 6 to 12 dB without any audible artifacts.

FinalPlug also includes a comprehensive bit truncation and dithering processor. These functions were principally developed for production of 16-bit CD audio, and as CD production has fallen precipitously since peak production in year 2000, the functions are provided for legacy compatibility and are hidden by default.

A diagram of FinalPlug's audio routing and meter placement is shown below.



Note that the dither processor runs before loudness metering, although in the UI it is shown on the right side.

3.2 Peak Limiting

FinalPlug's peak limiter can be used to do tasks ranging from light peak limiting to volume maximization to very heavy compression. The FinalPlug limiter detects peaks in the signal and very quickly reduces the input gain prior to the peak, preventing clipping. After the peak has passed, FinalPlug restores the input gain to the nominal level.

The peak limiter will limit any peak that exceeds the threshold level set by the user. The peak limiter uses a built-in lookahead delay to respond to peaks just before they occur. Both the lookahead delay and attack time are fixed at 1.5 milliseconds. This is designed according to the pre-masking phenomenon of the human auditory system whereby loud sounds effectively mask sounds that occurred just prior. Hence, any distortion caused by the rapid gain ducking will be effectively masked by the onset of the peak. The shape of the peak itself is unaffected by the gain ducking; this is in contrast to "soft clip" peak limiters that use wave shaping techniques. After the peak has passed, the gain is restored to nominal level according to the release time set by the user.

The release time is an important parameter. Short release times cause the gain to be increased rapidly after peaks. This is important when you want to aggressively compress a signal. However, short release times also cause the gain to modulate more deeply during steady audio signals, causing distortion. This is because with aggressive settings the peak limiter will be compressing each oscillation of the input signal; ducking the gain to allow the peak oscillation to pass, then quickly restoring the gain as the signal passes through 0.

In contrast, long release times prevent distortion during steady audio signals. This is because the gain will be ducked on the onset of the signal and will stay ducked because the release happens so slowly. Hence the amount of gain modulation with each oscillation of the input is very small, and hence the distortion is greatly reduced compared to using a fast release time. However, the disadvantage of using a long release time is that it takes longer for the gain to increase after loud sounds in the input. This can cause an audible pumping of the background level of the audio; each peak in the audio causes the background to be reduced in volume and then the background level increases slowly until the next peak.

FinalPlug features an auto-release mode which dynamically chooses the optimum release time based upon the program content. FinalPlug chooses the release time based on the crest factor of the input signal. The crest factor is an indicator of the peakiness of a signal; it is obtained by comparing the peak level of a signal to the average level. FinalPlug uses a short release time for input signals that are peaky, and it uses a long release time for signals that are steady. The implementation of the auto-release mode is a

balance between low distortion operation, aggressive compression, and reduced pumping effects. The auto-release mode in FinalPlug achieves a nice balance, preventing distortion during steady tones even when compressing heavily, while reducing pumping effects and using fast release times when possible.

FinalPlug limiter provides an internal oversampling feature which allows the internal peak limiter to run at 1x, 2x, 4x, or 8x the original sampling rate. The sampling rate converter is very high quality, using a 64-point interpolation filter, and is computationally expensive. The reason to run the limiter at a higher sampling rate is as follows. The limiter can rapidly modulate the signal gain, which creates modulation harmonics. If these harmonics extend to high frequencies above the Nyquist frequency (half the sampling rate) they wrap around into the audible range but at frequencies no longer harmonically related to the original signal, known as "alias distortion." If instead the limiter is run at a higher internal sampling rate, the high frequency overtones will be filtered out when converting back to the base sampling rate, with significantly less alias distortion. It should be noted that the FinalPlug limiter modulation generates less harmonic distortion than a "soft-clip" style limiter, so the benefits of oversampling are reduced with respect to the soft-clip style limiters.

Lookahead peak limiters like FinalPlug can be used to increase the loudness of a mix. This technique has often been used to excess, creating ultra-compressed audio that can be quite fatiguing to listen to, and resulting in the so-called "loudness wars." One need only look at the digital content of a typical pop CD to see how hyper-compressed typical CD music was, prior to the onset of streaming delivery and loudness standards. Readers are encouraged to read the following: "Current Trends in Mastering: The Loudness War," by Mark Donahue, June 2003 Performer Magazine, and "What happens to my recording when it's played on the radio?" By Frank Foti, Omnia Audio & Robert Orban, CRL/Orban.

3.3 Loudness Metering

In 2010, the International Telecommunication Union (ITU) developed a new standard BS.1770 for measuring the loudness of audio signals. There was a real need for an improved standard, as standards based simply on peak or RMS level led to a wide range of loudness of different audio programs. For example, listeners could experience wildly differing volumes of TV advertisements with respect to TV programs, or between songs played on the radio, or between film previews and the film soundtrack. The goal of the loudness standard was to determine a single numerical value to quantify the subjective loudness of an entire audio program. This would allow programs to be normalized to the same loudness with respect to this metric.

Most loudness measurements are some variation of RMS (root mean squared) levels. Essentially this is an average of the squared sample values

of the digital signal. This is completely analogous to the energy in an electrical signal. The instantaneous power of an electrical signal is proportional to the voltage squared, with units of Watts. To determine the energy in the signal requires summing this over time, resulting in units of Watt-hours. Similarly, the energy of a digital signal is the sum of the squared sample values.

Related standards such as A-weighted and C-weighted Sound Pressure Level apply a filter to weight the measurement as a function of frequency, in an effort to model the sensitivity of the human ear. For example, the A-weighted frequency response rolls off the low frequencies and some highs, so it is most responsive to midrange frequencies between 1-8 kHz.

The ITU BS.1770 standard applies a frequency weighting, called K-weighting, which combines a low frequency roll-off with a high frequency boost above 2 kHz. Hence K-weighted measurements are less sensitive to low frequencies and more sensitive to high frequencies.

The ITU BS.1770 standard also specifies a gating algorithm so that the final loudness measurement ignores long periods of silence in the signal, which would decrease the loudness value if included in the average. The so-called "Integrated Loudness" (IL) measurement is the final loudness metric for the entire audio program. The units are LUFS (loudness units, relative to full scale) also referred to as LKFS (loudness units, K-weighted, full-scale). They are in fact dB units entirely analogous to dB RMS units but with different weighting and gating logic. Hence if you increase the signal by 1 dB you would expect the IL to also increase by 1 LUFS.

Also computed is the "Short-Term" loudness which is computed over a period of 3 seconds without gating, and the "Momentary" loudness which is computed over a period of 0.4 seconds without gating.

Finally, the ITU BS.1770 standard establishes the notion of True Peak value. The True Peak value is computed by oversampling the signal and then measuring the peak value, and hence it measures peaks that can occur between the samples, known as "inter-sample" peaks. Why is this important? Limiting with respect to the true peak value is important to avoid clipping or overflow conditions in subsequent audio processing. For example, the audio might be played through a consumer D/A converter, which might internally oversample the audio and which might be implemented using fixed-point arithmetic. If an inter-sample value overflows, this can result in audible distortion due to the fixed-point arithmetic clipping the result or worse, wrapping around to the opposite polarity. Similar concerns exist for audio encoder/decoder algorithms or hardware implemented using fixed point arithmetic. By limiting to True Peak levels, subsequent audio processing is given more computational headroom. In time, as floating-point arithmetic becomes ubiquitous in all audio processing algorithms and hardware, there will be less need to worry about true peak levels.

The FinalPlug loudness meters display Integrated, Short-Term, and Momentary loudness, and True Peak level. Additionally, when the Limiter is in True Peak mode, it responds to True Peak input levels and displays True Peak levels.

FinalPlug includes two dummy parameters “Integrated Loudness” and “Overflow” which can be recorded as automation parameters in the host DAW. FinalPlug updates the Integrated Loudness (IL) parameter with the current measured loudness, simulating a user gesture on the dummy control. By recording the Integrated Loudness parameter as an automation track, you can see a plot of the measured IL along with your audio. The range is limited to -72 to 0 dB so the automation display will have good resolution in the region of interest. Updates happen at the metering rate, nominally 20 Hz, settable in preferences. The Overflow parameter toggles on when the IL exceeds the target IL, and toggles off when the IL is below the target IL. The “Overflow includes True Peak” preference setting allows overflow to also toggle on when the measured True Peak level exceeds the target. By default this option is disabled. Automation of these dummy parameters can be disabled in the preferences via the “Write Automation” preference.

3.4 Truncation and Dithering

FinalPlug also features bit depth truncation with dithering and noise shaping. Bit depth truncation typically occurs as the final step in mastering when the sample resolution must be reduced to fit on the recording medium. For example, compact discs (CDs) are recorded with 16-bit samples, whereas digital audio editors typically use 32-bit floating point sample format containing at least 24-bit sample resolution. Hence, the 24-bit resolution samples must be reduced to 16-bit prior to burning the CD. This is done by discarding the least significant bits, which is the definition of bit depth truncation.

Truncation will cause the addition of "quantization noise" and "quantization distortion" to the resulting signal. For low level signals, i.e., signals that are only a few bits in amplitude, the quantization distortion can sound very objectionable. Dithering is a technique used to eliminate quantization distortion by adding a small amount of random noise to the signal prior to truncation. The added noise "dithers" the signal between the various quantized output levels in such a way that the quantization distortion is converted to pure broadband noise. Hence, dithering adds noise to the signal so as to convert the objectionable quantization distortion to less objectionable noise. Dithering is thus converting distortion to hiss. FinalPlug uses TPDF (triangular probability distribution function) noise for dithering. Mathematically, this is known to be the optimal type of noise to convert quantization distortion to noise that is uncorrelated with the original signal.

Noise shaping is a technique used to shape the spectrum of the quantization noise to make it less perceptible to the human ear. Typically, noise shapers push the quantization noise to higher frequencies to which the ear is less sensitive. However, there are many variations of noise shaping techniques in use. The user must decide which shape is best for the task at hand. In this spirit, FinalPlug provides a wide variety of noise shapes to choose from. Many of the shapes are based on other noise shapers in common use.

It should be noted that dithering and noise shaping may offer few practical benefits. This is because acoustic recordings typically contain much higher levels of noise than required to properly dither a signal. Hence, adding dithering will have no effect except to slightly increase the existing noise level. A study by D. W. Fostle (Audio Magazine, March, 1996) showed that very few recorded CDs have low enough noise floors to benefit from proper dithering and noise shaping. The advantages of dithering are even more tenuous when considering 24-bit media, like DVD-audio. Dithering and noise shaping will have benefits only when the original material has a noise floor below the quantization level of the final medium. This is unlikely when dealing with acoustically recorded material. However, purely synthetic material (say from virtual instrument) may indeed have very low noise floors and may benefit from proper dithering and noise shaping.

As shown in the above schematic, truncation and dithering occurs after peak limiting. Hence, the output level as shown by the output meters may slightly exceed the limiter ceiling level when dithering is active. This is because the dithering and noise shaping occurs after peak limiting. The effect is negligible for typical settings. We recommend that a ceiling of -0.1 dB be used for CD and DVD-audio mastering to allow a tiny bit of headroom for dither noise. Additional headroom may be needed when dithering signals at small bit depths, such as 8 bits.

FinalPlug allows bit depths down to 4 bits. Why? First, you might want to use FinalPlug to apply bit truncation as a digital distortion effect, to do this you can select a bit depth and mute the dither noise. Second, it’s instructive to play with low bit depths in order to hear the effects of quantization, dithering, and noise shaping. Try setting the bit depth to 4 bits and mute the dither noise, the sound will be very distorted by the truncation. Now unmute the dither and hear how the noise reduces the distortion. Now select different noise shapes to hear how the noise is moved to different frequency ranges.

3.5 Parameters

This section describes all the internal parameters of FinalPlug as would be displayed by a generic parameter-value plug-in interface. Most of these have a one-to-one correspondence with controls on the user interface.

Parameter name	Values
Bypass	0 = Off, 1 = On (Bypassed)
Limiter Enable	0 = Off, 1 = On
Threshold	-36 dB to 0 dB
Release	1 to 1000 msec
Ceiling	-36 dB to 0 dB
Dither Enable	0 = Off, 1 = On
Bits	4 to 24
Noise Shape	0 = Flat, 1 = 44.1 kHz Shape1, 2 = 44.1 kHz Shape1 Ultra, 3 = 44.1 kHz Shape2, 4 = 44.1 kHz Shape2 Ultra, 5 = 44.1 kHz Shape3, 6 = 44.1 kHz Shape3 Ultra, 7 = 44.1 kHz Shape4, 8 = 44.1 kHz Shape4 Ultra, 9 = 44.1 kHz Shape5, 10 = 96 kHz, 11 = 96 kHz Ultra
Mute Dither	0 = Off, 1 = On
Auto Release	0 = Off, 1 = On
TruePeak Enable	0 = Off, 1 = On
Loud Enable	0 = Off, 1 = On
Loud Sync	0 = Off, 1 = On
Loud Target	-200 to 0 dB †
TruePeak Target	-200 to 0 dB †
Input Gain	-30 to +30 dB
Upsample Factor	0 = 1x, 1 = 2x, 2 = 4x, 3 = 8x

Integrated Loudness	-72 to 0 dB ‡
Overflow	0 = Off, 1 = On ‡

† When target values are set to -200 dB, they display as an empty string. The “None” target preset has both targets set to -200 dB to clear the target fields in the UI.

‡ The Integrated Loudness and Overflow parameters are dummy parameters which have no effect on audio processing. These parameters are optionally written as automation parameters so the user can plot automation tracks of IL and overflow condition along with the audio in the DAW session. These parameters are written when the “Write automation” preference is enabled, and the DAW has been set up to record them.

3.6 Latency

The limiter, when enabled, imposes a latency of 3 msec. The limiter oversampler, when enabled, imposes a latency of 63 samples. These values are provided to the host DAW for latency compensation. When FinalPlug is bypassed, the signal runs through a delay line equal to the current latency.

4. Installation and Registration

4.1 Installation

Installers for FinalPlug are found on the downloads page of the Wave Arts website. There are separate installers for Mac and Windows. Mac installers are ".dmg" files which after downloading will expand into a ".pkg" installer file; double-click on the ".pkg" file to launch the installer. Windows installers are ".exe" files; double-click on the ".exe" file to launch the installer. The installers provide various options for selecting which plug-in formats to install and whether to use PACE iLok or Wave Arts licensing.

4.2 Registration

We support two licensing methods – Wave Arts licensing and PACE iLok. When installing the plug-ins you must select which version of the plug-ins you wish to use. When you purchase a plug-in, you will be e-mailed a serial number (looks like WA-PPP-XXXX-XXXX where PPP is a product code and X is a hex digit). Use the serial number to unlock the plug-in as described below.

4.3 Wave Arts licensing

Prior to registration, the plug-ins operate in demonstration mode; they are fully functional but stop operating after 30 days. To unlock the plug-in after purchasing, open the plug-in, select the Tools->Register option, and enter your name, email address, and serial number. The plug-in will contact our registration server and download a license file which will unlock the plug-in. You should see a message saying your registration was successful.

If your computer is not connected to the internet, use the Tools->Offline Register option and follow the instructions to generate a keyfile at our website registration page and import the keyfile.

If you have purchased a plug-in suite, when you unlock any one of the plug-ins within the suite, the entire suite will be unlocked.

4.4 PACE iLok licensing

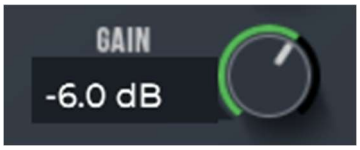
All our plug-ins support PACE iLok. Prior to activation, the plug-ins will allow you to start a 30-day trial by creating an iLok account. To unlock the plug-in after purchasing, go to our Product Registration page, select "iLok", and enter your serial number. A PACE redeem code (looks like XXXX-XXXX-XXXX-

XXXX-XXXX-XXXX-XXXX-XX) will be displayed and also emailed to you. There are two ways to redeem the code and generate a license. When opening the plug-in a dialog window will appear and give you the option to Activate the plug-in, you can paste the PACE redeem code there, and proceed to create or login to an iLok account and then transfer the license to an iLok or your machine. Otherwise, go to http://www.ilok.com, create an iLok account, and download and install the iLok License Manager. Within the manager, under the Licenses menu, select "Redeem Activation Code" and paste your redeem code. Then transfer the license to either an iLok dongle or your machine. The plug-in will run only if it can find a license on an iLok or the machine.

When purchasing a plug-in suite, the redeem code will generate multiple licenses, one per plug-in in the suite, but the licenses are grouped together.

5. Plug-in Control Operation

5.1 Knobs



Please refer to the following guide for information about the various ways you can use knobs:

Function	Mac	Windows
Increase/Decrease a parameter value (rotate clockwise/counterclockwise)	Click on the knob + drag up/down -or- Mousewheel	Click on the knob + drag up/down -or- Mousewheel
Fine adjustment — increase/decrease	Shift + click + drag up/down -or- Command + click + drag up/down	Right click + drag up/down -or- Shift + click + drag up/down -or- Control + click + drag up/down
Reset knob to default value	Double-click	Double-click

By default knobs follow up/down mouse motion, but this can be changed in the preferences settings.

5.2 Text Entry

Many value displays are editable text. A text field is editable if your mouse cursor changes to an I-beam when moved over the text. Following is a table that fully describes how to use the text editing features:

Function	Mac	Windows
Enter text entry mode	Click in the display	Click in the display
Select text	Click + drag	Click + drag

Select entire text	Double-click	Double-click
Delete character to left of cursor	Delete	Backspace
Delete character to right of cursor	Fn+Delete	Delete
Move the cursor left/right	Left/Right arrow keys	Left/Right arrow keys
Extend the current selection	Shift + click + drag -or- Shift + left/right arrow keys	Shift + click + drag -or- Shift + left/right arrow keys
Exit text entry mode	ESC -or- Return/Enter -or- Click on panel	ESC -or- Return/Enter -or- Click on panel
Select next parameter to edit	Tab	Tab
Select previous parameter to edit	Shift + Tab	Shift + Tab

You'll find that many parameters, such as frequency, will recognize units typed into the text field. The following values, when typed into a frequency value box, are equivalent:

2k = 2 kHz = 2000 = 2000 Hz

5.3 Sliders

Function	Mac	Windows
Increase/Decrease a parameter value	Click + drag	Click + drag
Fine adjustment — increase/decrease	Shift + click + drag	Right click + drag -or- Shift + click + drag -or- Ctrl + click + drag
Reset slider to default value	Double-click	Double-click

6. Menu Bar and Preset Manager

This section describes the operation of the menu bar, preset manager, and the other functions available in the menus.

6.1 Enable

The enable button looks like a power button. When lit the plug-in is enabled, and when unlit the plug-in is bypassed – it passes audio but without modification.

6.2 Undo/Redo

Clicking the Undo button causes the parameters and view options to revert to their settings prior to the last edit. The undo stack is unlimited, so you can keep clicking and revert back to the settings when the plug was initially opened. Clicking Redo restores the last undo. You can continue to redo back to the settings before undo was first used. However, if you make any edits the redo stack is discarded.

The A and B buffers (described below) have independent undo/redo stacks. The undo/redo stacks store all parameters and many of the UI view options. Plug-in preferences are not stored in the undo/redo stacks.

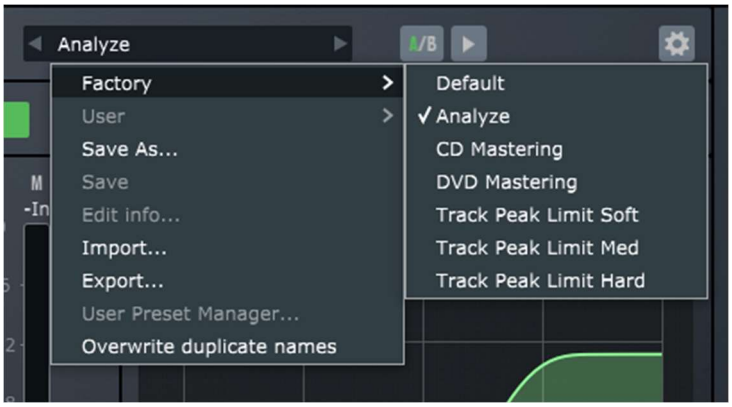
6.3 Preset name and arrow controls

The currently selected preset name is displayed in the text field in the menu bar. Changing any parameters causes an asterisk (*) to be displayed at the end of the name. This indicates that changes have been made to the preset. In order to save the changes to a user preset you must select the "Save…" item in the Preset menu, described below.

The arrow controls to the left and right of the preset name cycle through the set of factory and user presets. Clicking the right arrow goes to the next preset, clicking the left arrow goes to the previous preset.

6.4 Preset menu

The Preset menu contains lists of factory and user presets for easy selection, and options for managing presets. The functions are described in the following sections.



6.4.1 Factory Presets

Factory presets are selected from a rolloff menu at the top of the Preset menu. Factory presets cannot be modified or deleted.

6.4.2 User Presets

User presets are selected from a rolloff menu just below the Factory presets in the Preset menu. When you first run a Wave Arts plug-in, there will not be any user presets and the menu will be empty. When you save a preset using the "Save" option the preset is added to the User menu. All instances of a plug-in share the same set of user presets. So, after you save a preset with one instance of a plug-in, you can go to another instance and find that the preset can be found in its User preset menu too.

User presets are stored in an XML format file called "<plugin> Presets.xml", where <plugin> is the name of the plug-in you are using. If the file is deleted, an empty preset file will be created automatically the next time the plug-in runs. User presets files are stored in the following directory, depending on the operating system, where <username> is your login name:

Mac OS-X:
/home/<username>/Library/Application Support/Wave Arts/<plugin>/

Windows:
C:/Users/<username>/AppData/Roaming/Wave Arts/<plugin>/

6.4.3 Save As…

When you have created an effect you want to save as a preset, select the "Save As…" option. You will be asked to name the preset and the preset will be saved in the set of User presets. If you supply the same name as an existing user preset, the existing preset will be either be overwritten with the

new preset or the new preset will be given a unique name, based on the setting of the "Overwrite duplicate names" option.

6.4.4 Save

The Save option is used to save changes to a user preset. The Save option is enabled when the current preset is a user preset and you have made changes, in which case an asterisk (*) will be appended to the preset name. Select the Save option to save the changes to the preset.

6.4.5 Import Legacy

Import legacy will find any user preset files from earlier versions of the same plug-in and will display them in a rolloff menu. Selecting a preset file will import the presets into the current user presets, converting the legacy presets to the current version.

6.4.5 Import…

User presets can be written to files using the "Export" function, and read from files using the "Import" function. Selecting the "Import…" option will open a file chooser to select the preset file for importing. After selecting the file the presets are read and will appear in the User Presets menu.

Duplicate preset names are handled according to the "Overwrite duplicate presets" setting.

Import can be used to import presets from an earlier version of the plug-in; the preset parameters will be converted to the current version of the plug-in.

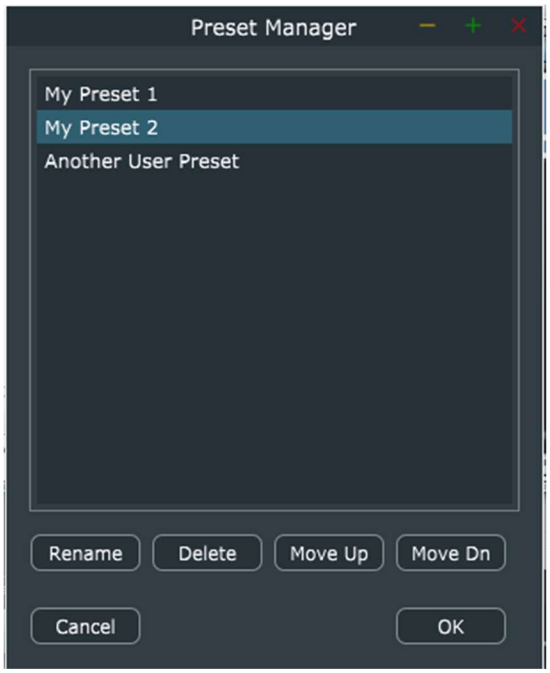
6.4.6 Export…

Selecting the "Export…" option will open a file chooser to specify the output file. Then your user presets are written to the file.

Preset Export is also useful for making backup copies of your user presets. If you have a large set of user presets, be sure to export them to a backup file.

6.4.7 User Preset Manager

The User Preset Manager provides a simple interface to organize user presets. You can rename presets, delete presets, and organize them by moving them up/down in the list.



6.4.7 Overwrite duplicate names

If this option is selected, then when you save a preset using "Save As…" using the same name as an existing preset, or when you import a preset with the same name, the existing preset is deleted and replaced with the new preset. If the overwrite option is not select, the duplicate preset name is handled by renaming the new preset, appending "(Copy)", "(Copy 1)", "(Copy 2)", etc. as needed to avoid a duplicate name.

6.5 A/B buffers

The A/B edit buffers allow you to compare two different sets of parameters or presets. One of the A or B indicators is always lit; the one that is lit shows the current buffer. Clicking the A/B button will switch to using the other buffer, thus changing the effect settings (assuming different settings are stored in A and B). This is quite literally an A/B compare function.

Once you have settings you like in buffer A, switch to buffer B and setup different settings, then click A/B to switch between the two.

6.6 Copy buffer

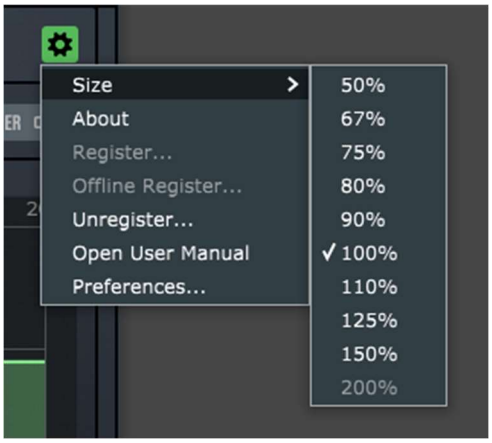
Clicking the Copy (arrow) button switches to the other buffer *while also copying the buffer*, hence the two buffers will be equal after the copy. The arrow points to the right when A is selected, and points left when B is selected.

Typical buffer use is:

1) Get settings you like in buffer A.
2) Copy to B, switching to buffer B.
3) Further edit the settings in buffer B.
4) Click A/B to compare the two buffers.

6.7 Tools menu

The Tools menu contains various important options, described below.



6.7.1 Size

The Size menu rolloff lets you change the interface size with one mouse gesture. After changing the size, clicking the plug-in title in the title bar will toggle between the last two selected sizes. Sizes that would be too large for the display are disabled.

6.7.2 About...

The About option displays important information about your plug-in. An example is shown below:

On the top line, the plug-in name and version are displayed, along with the current plug-in format (AAX, VST3, AU), bit depth, and CPU architecture. This is useful to determine which format of the plug you are running. The build date of the plug-in is displayed on the next line. If the plug-in is using Wave Arts licensing, the registration status is displayed on the next line. If the plug-in is operating in demo mode, the time remaining (if any) is displayed. If the plug-in has been successfully registered (unlocked), the serial number is displayed. If the plug-in is using iLok licensing, it will display "Pace/iLok licensing".

6.7.3 Register...

Select this option to register (unlock) your plug-in. Enter your name, email address, and serial number, and click OK. This option is not available if using iLok licensing.

6.7.4 Offline Register...

Select this option to register the plug-in when your computer is not connected to the internet. This option is not available if using iLok licensing.

6.7.5 Unregister…

Select this option if you no longer need to use the plug-in on this machine. The license file will be deleted and the serial number usage count will be decremented, allowing you to register on another machine. After confirming, the plug-in will revert to an unlicensed state, either bypassed or demo mode. The unregister option requires an internet connection. This option is not available if using Pace/iLok licensing.

6.7.6 Open User Manual…

Select this option to open this user manual in a browser.

6.7.7 Preferences…

This option opens the Preferences dialog to customize the plug-in operation.

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Support

For assistance, please send email to:

support@wavearts.com

If you are having problems with a plug-in, please include the following information: plug-in name, operating system, and host software you are using. The version numbers are also helpful.

For software updates, revision history, frequently asked questions (FAQ), and more, please visit our website at:

www.wavearts.com

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