ClipShifter (2.0)

LVC-Audio

Contents

License Agreement

This software is distributed in two forms: a free plugin with specific processing limitations, and a paid version. Both the free and the paid versions of the plugin can be used for personal or commercial purposes without limitations.

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Technologies Utilized

'**VST**' is a Technology and Trademark by Steinberg.

WDL-OL: Enhanced version of Cockos IPlug by Oli Larkin (https://github.com/olilarkin/wdl-ol)

Knobman and **Skinman**: from g200kg (http://www.g200kg.com/index_e.html)

Ubuntu Font Family: (http://font.ubuntu.com/)

Welcome, Thanks, and Contact Information

Thank you for installing and trying ClipShifter. I hope you find this useful, and I would like to hear your suggestions for future enhancements. Please feel free to contact me with any ideas, problems, preset suggestions, or comments at **support[*the email symbol*]lvcaudio.com**. Please visit lvcaudio.com for additional news about ClipShifter and other plugins.

Introduction

ClipShifter is a plugin designed to provide saturation and clipping effects to your audio. This can be useful for individual tracks, busses, and overall mixes. The sonic characteristics of the clipping distortion can be altered from hard, brickwall-style clipping, to softer saturation with compression. The unique threshold controls for ClipShifter's clipping can be set to dynamically change based on the transient characteristic of the audio.

ClipShifter is released as a free and paid plugin. Paid features are indicated in this manual.

Initial Setup

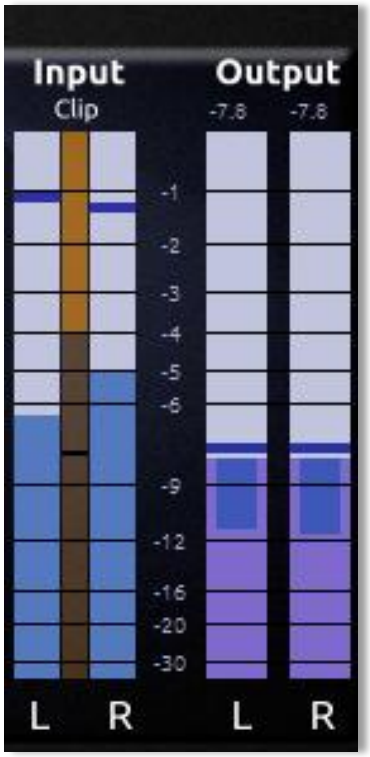
ClipShifter is provided as a VST plugin for computers running the Microsoft Windows operating system. To install the software, place the 'ClipShifter.dll' file within your VST plugin folder on your computer. Ensure that your audio software recognizes this directory as the location for VST plugins.

ClipShifter is provided as a 32-bit and 64-bit plugin for Windows Vista/7/8. The 64-bit plugin should be used with 64-based audio applications. There is an additional download for Windows XP 32-bit systems.

User Interface

ClipShifter is organized into three main sections: the waveform history view, the input and output VU meters, and the control settings. The main controls of ClipShifter are adjusted by using various knobs and buttons. For each of the knobs, the text value is displayed below. Manual entry of values can be entered by clicking on the text field, and typing in the desired value. Double-clicking on any knob or slider will return the control to the default value.

VU METERS



VU and Peak Program Meters: The audio meters on ClipShifter display information about the input, output, clipping amounts, and clipping threshold. The input meters display the signal level after gain is applied through the Input Gain. The meter bar responds similarly to traditional VU meters, with a quicker rise time (0.1ms integration time, and 300ms to decrease -20 dB). The dash meter responds more closely to a Peak Program Meter (PPM), with a slower rise and fall (3ms integration time, and 5 seconds to decrease -20 dB). All meters are calibrated as 0dBFS, meaning that any signal that maximizes the meters represents digital clipping.

Clipping and Threshold Indicators: The orange meter in the middle displays information about the amount of clipping reduction by applied to the signal. The dark orange meter shows the amount of clipping that is occurring. The responses of the clip reduction meter is a quick rise time, and a slower fall time (0.1ms integration time, and 1s to decrease -20dB). The black dash indicator displays the current threshold for clipping to occur. This is dependent on the settings within the Threshold section and the incoming audio.

The output meters display the signal level that is sent from the plugin to the host. This includes all gain settings and clipping.

Please note: there is no internal protection from sending signals that are too hot to the host program. Distortion due to inappropriate gain staging can occur, depending on the settings and the host application.

Loudness Output Meter: An additional feature of the Output meter is a loudness indicator. This is the smaller blue area within the purple-blue VU bar. This indicates the relative loudness of the output. The top of the area is determined by the peak output level. The bottom of the area is determined by the RMS value of the output (RMS window of 1000ms). The size of the bar relates to the dynamic range of the material. When heavy clipping occurs and the output levels approach 0dBFS, the loudness meter will be narrow (i.e. representing little difference between the peak of the audio and the RMS value). Audio with more dynamic characteristics will have a relatively broader loudness meter. Although this can be useful in determining the overall level of dynamic range, the meter is less accurate at lower audio levels.

Above the Output meters are a text readout of the current output levels in decibels. These numbers have an instantaneous integrations time, and a long decay time. The numbers will only display volume levels above -60dB. Signal levels below -60dB will not be displayed.

Any signal that is above 0dBFS will be displayed in **red text**, indicating digital clipping. To reset the values, click on the numbers. This will automatically reset the numbers to the default -60dB value.

CONTROLS

The main control interface of ClipShifter contains various controls for changing the gain of the audio, controlling the clipping settings, and adjusting the sonic characteristics of the clipping. The In Gain control applies gain before any other processing occurs. Conversely, the Out Gain applies gain after all processing is finished and before the signal is routed to the audio host.

Initial and End Threshold: The larger Threshold controls in the center of the plugin change the clipping amount. The section has two knobs used to set the clipping threshold (Initial and End). These settings are used in conjunction with the Attack Time and Fast Release controls to shift the threshold of clipping based on the level of the input signal. After the input signal reaches the level set by the Initial Threshold, the actual threshold of the plugin begins to change toward the amount set by the End Threshold. The quickness of the shift is determined by the Change Time control.

After the signal decreases, the actual threshold of the plugin begins to shift back towards the Initial Threshold setting. The release time works as a multiple of the Attack Time. If a faster release time is desired, the Fast Release button can be enabled. The indicator in the middle of the Threshold section shows where the actual threshold of the plugin is functioning in comparison to the Initial and End settings.

Clip Shape: The Clip Shape controls the type of clipping applied to signals above the threshold. Setting the Clip Shape to 1.00 equates to a brickwall-style clipping (this is similar to clipping that occurs in electronics using diodes across the output of a signal chain). Anything above this threshold is cut drastically. Decreasing the Clip Shape starts to soften the clipping (similar to diodes placed within the feedback loop of an opamp/transistor-based circuit). In addition, softer clipping (i.e., lowering the Clip Shape) will compress the signal. At the minimum setting, almost no audible clipping and approximates a compressor set at a low 1:1.5 ratio.

Harmonics: The Harmonics control adjusts the amount of even- and odd-order distortion while the signal is clipping. A setting of 1.0 will produce a symmetrical output signal that

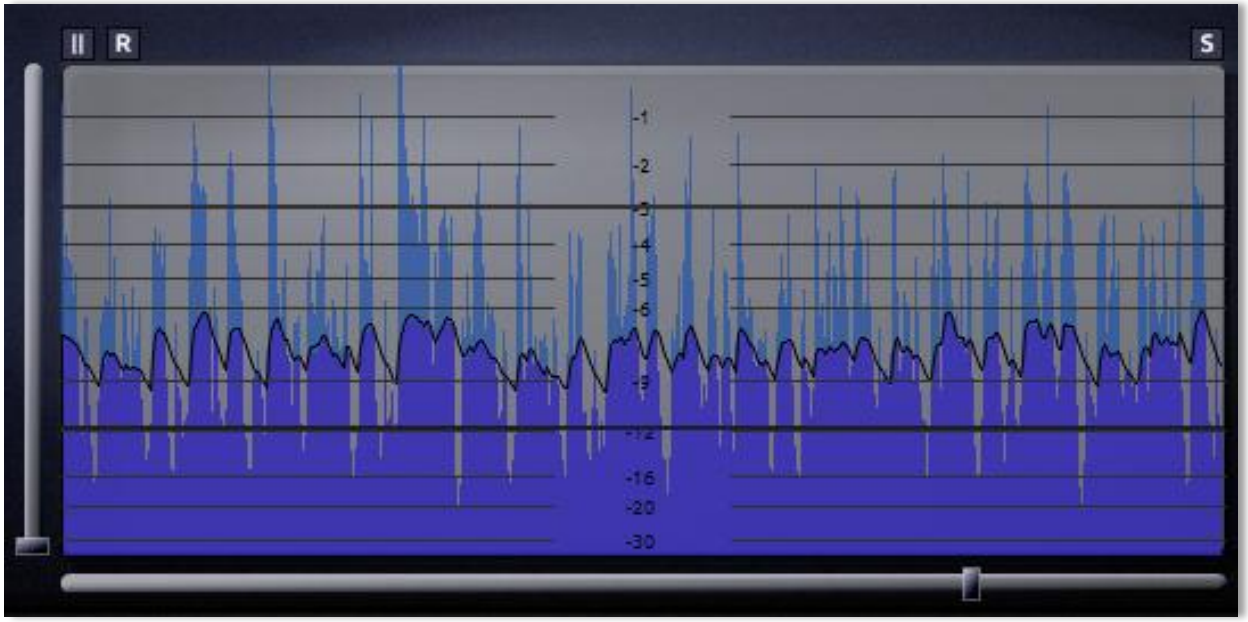
contains only odd-order harmonics. Settings less than 1.0 will increase the amount of even-order harmonics in the output.

Please note: settings less than one will also increase the overall gain of the output signal.

Link: The Link button locks the Initial and End thresholds to the same amount, and disables the Change Time. Linking the two controls holds the actual threshold of the plugin to the amount set by the controls. While linked, the Change Time setting has no impact and the meters and waveform history will not show any change.

Bypass: The Bypass button disables processing of the plugin. Although the meters continue to work, all controls within the Input, Output, and Threshold section do not have any effect on the audio signal.

WAVEFORM HISTORY



The Waveform History displays information about the peak input, peak output, and dynamic clipping threshold. By default, the output meters are displayed in a purple-blue color, and the input levels are displayed in a lighter blue color. These colors match the colors displayed in the VU meters. The dynamic clipping threshold is displayed as a solid black line.

Display Time: The default setting for the waveform history is to display 10 seconds worth of audio, and to show the entire volume level (i.e. from infinity to 0dBFS). The horizontal and vertical sliders can be used to adjust these settings.

Sliding the horizontal slider (located at the bottom of the waveform history display) adjusts the amount of time that is displayed. The minimum setting is 0.25 seconds, and the maximum time is 20 seconds.

Display Volume Level: The vertical slider adjusts the floor of the display with respect to volume. Increasing the slider will raise the floor from infinity to approximately -3dB. Increasing the slider zooms in on the loudest parts of the audio signal. The vertical slider also adjusts the scale displayed by the VU meters.

Please note: increasing the slider on quieter audio signals can have the effect of hiding the input, output, or clipping threshold.

Initial and End Threshold Adjustment: Both the Initial and End Threshold values are displayed on the waveform history view as horizontal black lines. The actual clipping threshold will vary between these two lines. Although the thresholds can be adjusted by the corresponding knobs, they can also be adjusted within the waveform history view. Clicking on either of the lines will enable the Initial or End threshold to be adjusted by the mouse.

Pause and Reset Buttons: The waveform history view can be temporarily paused to view the audio signals. The button and the top-left side (button) pauses and un-pauses the display. This does not affect the audio that is processed through the plugin. Additionally, the vertical volume adjustment slider can be used while the waveform history view is paused. The Reset button (button) next to the Pause button clears out the current waveform display.

Display Setting: Clicking the Settings button (button) displays an additional dialog box for adjusting the meters and waveform history view. Clicking the button a second time closes the dialog box, although the settings are still applied. The two color knobs at the top of the box adjust the hue of the Input and Output displays. Changing the Input hue will subsequently change the waveform view and the Input VU meter. Both the Input and Output colors are semitransparent, so the colors will interact with one another.

The Hide/Show buttons turn on or off the corresponding display within the waveform history view. The default setting is for the Input, Threshold, and Output to be displayed simultaneously; however, one or more settings can be turned off.

Features of the paid version of ClipShifter

The paid version of ClipShifter adds two additional features: oversampling and multiband frequency controls. Both controls enhance the versatility of ClipShifter. Specifically, these controls make ClipShifter more desirable for final audio processing or for more strategic saturation effects.

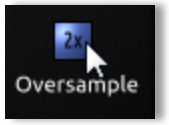
OVERSAMPLING

A problem with many audio plugins is aliasing artifacts. This occurs with certain algorithms that apply non-linear processing to the audio signal. It is especially prevalent at lower sampling rates (e.g., 44100 or 48000). The result is that additional audio content is present at unintended frequencies. This can sound like undesired distortion or harmonic ringing. In general, ClipShifter minimizes aliasing effects through the use of its internal clipping algorithm; however, certain situations can produce unintended audio artifacts.

Oversampling is a process that greatly minimizes alias effects. The general process is to increase the samplerate internally, process the audio, filter out all of the alias artifacts, and then down convert the samplerate. This occurs internally and the audio host application is not aware that any samplerate conversion is occurring.

ClipShifter has settings for 2x, 4x, and 8x oversampling. This means that audio is internally being processed at the samplerate of the track, times the multiple (e.g. 4x of a 44100 sample is up-converted to 176400 samples per second). The result is that most of the aliasing artifacts do not occur once the samples are filtered and returned to the original samplerate before passing to the host application.

Since oversampling involves processing 2, 4, or 8 times the amount of data as well as additional filtering, oversampling can tax the computer's CPU. Depending on the computer, this may appear as a CPU usage of 2 to 4 times the amount when compared with oversampling disabled.



Clicking the oversample button cycles through the various options (i.e. Off, 2x, 4x, 8x).

Please note: Oversampling involves filtering high frequency content of the audio. Depending on the frequency content of the source and the settings of the multiband frequency control, audible difference can exist between the different settings. This can sound like a slight decrease in the very top end of the audible spectrum (i.e., -3dB above 17kHz).

MULTIBAND FREQUENCY CONTROL

The Multiband Frequency Control is a 3-way variable crossover control that can be used to tailor the sound of ClipShifter. The saturation/clipping processing can be selectively applied to any of the frequency bands. The controls include frequency selectors, gain controls, solo buttons, and a frequency selector control.

The settings can be accessed by clicking the F button next to the Input Gain control. Clicking the button a second time closes the control, although the current configurations are still applied to the audio.

Frequency Controls: The top two knobs control the frequency points of the 3-way crossover. The crossover utilizes a LR 4th order filter to split the audio into Low, Mid, and High sections. The default settings for the controls are 20 Hz for the Low control, and 20,000 Hz for the High control. When the controls are set to their default settings, the filters are disabled. As the knobs are changed from their default settings, the frequencies splits are engaged. When the filters are disabled, the corresponding Gain and Solo buttons are hidden. For a 2-band processor, either the Low frequency or High frequency can be disabled. The result is that the audio is split into 2 distinct frequency bands (with the Mid either functioning and the High band when High is disabled, or functioning as the Low band with the Low is disabled).

Gain Controls: The gain controls change the gain setting of each of the Low, Mid, and High sections prior to processing through the clipping processor. In effect, these controls are input gains for each frequency section.

Solo Buttons: Clicking a solo button will mute the audio from the other frequency bands. This can be useful in adjusting the appropriate frequency split and/or volume. The waveform display will also change to show the soloed output only.

Frequency Selector: The Frequency Selector determines what audio is processed through the clipping algorithm, and what audio is passed through to the output without any clipping or saturation. The default setting of for all of the audio signal to be processed through the saturation algorithm. When all three frequency bands are enabled, the selector can be set to process Low, Low + Mid, Mid, Mid + High, High, or All frequency bands. Clicking the Active button cycles through all of the available options. This is dependent on the number of frequency bands currently enabled. When a frequency is not enabled, the audio is still passed through to the output with all of the input, output, and band-specific gain; however, the clipping algorithm will be bypassed for this frequency band.

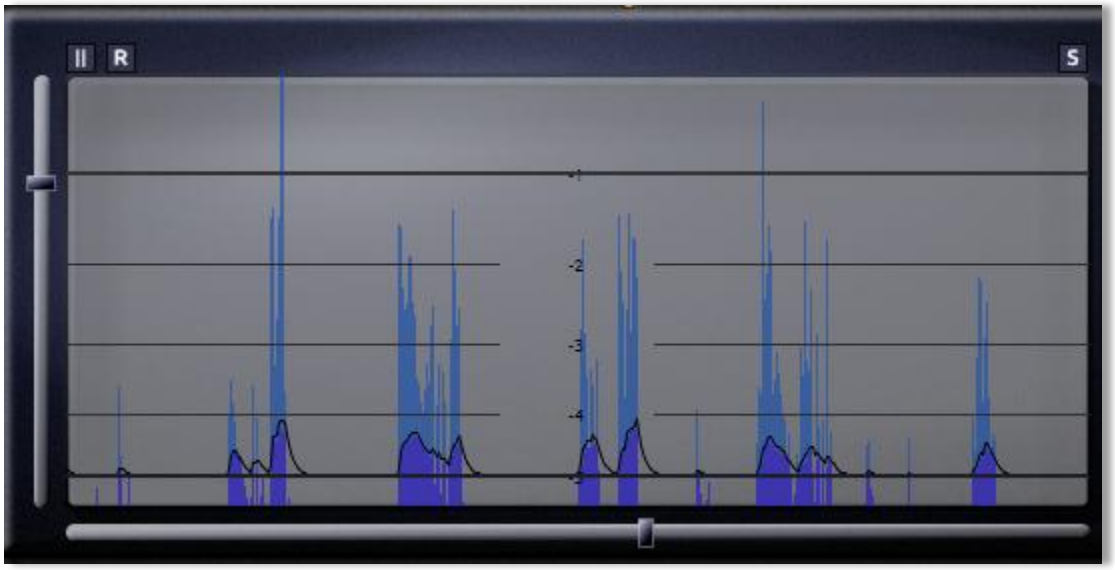
Please note: the input gain displayed within the waveform history view and input VU meters show the audio that is being processed by the clipping algorithm, even if no clipping is currently occurring. When the Frequency Selector is set to process one or two bands, the displayed input values will show the level of the Frequency Selector bands. This includes any input gain and frequency specific gain.

Presets and Uses

USAGE

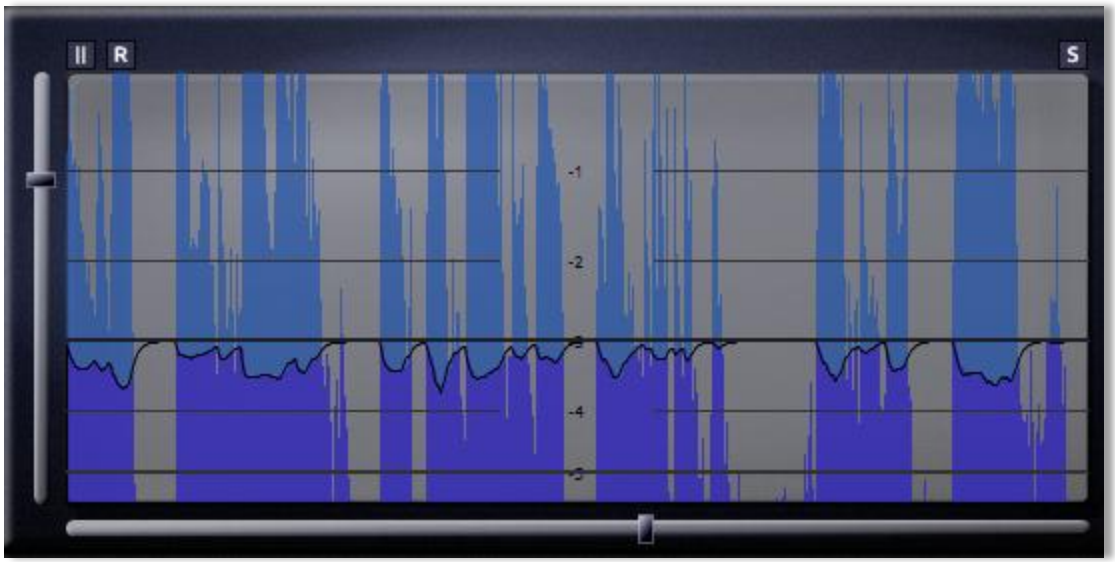
ClipShifter contains a few basic presets; however, experimentation is a necessity. Each audio signal will be different. Using the meters, waveform history view, and or course ears, is required to adjust the controls in order to produce the desired sound. Input Gain, in conjunction with the Initial and End Thresholds, is critical in obtaining a good sound.

Since ClipShifter can change the threshold of clipping dynamically, interesting results can be produced. Typical usage can involve setting the Initial Threshold to a lower setting than the Ending Threshold. The result is that the first transient signal that is above the Initial Threshold will be clipped, but additional audio that is above the threshold will contain fewer amounts of clipping. This is significantly impacted by both the attack and release time. By using the waveform history view, the dynamic nature of the clipping threshold can easily be visualized and subsequently adjusted.



Decreasing Distortion: Initial Threshold at -5dB, Ending Threshold at -1dB

Conversely, clipping the secondary transients more than the initial transient can occur by reversing the Initial and Ending Thresholds. Using a higher Initial Threshold (e.g., -3 dB), lower End Threshold (-5 dB) results in less saturation for the initial transient, but increasing levels of saturation for audio above the threshold.



Increasing Distortion: Initial Threshold at -3dB, Ending Threshold at -5dB

Preset Defaults & Format

ClipShifter can save and load presets using the standard fxp/fxb format. There are 20 slots for user presets. Saving and loading these presets is controlled by the audio host program.

Several default presets have been configured for ClipShifter. These include a variety of settings for drums, master buss, vocals, and bass. Please keep in mind that the overall sound of ClipShifter is highly dependent on the input level of the audio signal in relationship to the Initial and Ending Thresholds. Adjusting the input and output gain will be a necessity for any preset.

Certain presets are marked with "FREQ" (e.g. "Master1 – FREQ"). These presets use oversampling and/or the multiband frequency control. These presets will load within the free version of ClipShifter without any problem; however, the oversampling or frequency controls will remain bypassed.

Exporting and Importing Presets

ClipShifter can also export and import presets. Exporting presets produces a text file that can be loaded at a later time. The files are not specific to a particular user, so the files can be sent to other ClipShifter users to load.

To export a preset, set all of the controls and buttons to the desired configuration and left-click on the Preset Button at

the lower-left corner of the plugin. A file entitled “preset.txt” will be created within the same folder as the plugin dll file. If a preset.txt file already exists, the file will be overwritten each time the export button is pushed.

To import preset via the preset.txt file, open the file in a text editor like Microsoft notepad, or Notepad++. Select all of the text and copy the text into the clipboard. This can easily done with keyboard shortcuts (ctrl-A to select all text, ctrl-C to copy the text). When the preset text is in the clipboard, right-click on the Preset button to load the preset. All of the controls will be automatically set to the values within the preset.txt file.

Please note: It is important that the entire text is selected while copying the settings into the clipboard. Settings that are not copied will not be loaded correctly. Additionally, some programs such as Microsoft Word may introduce additional formatting characters to the text file when opening. Therefore, a “simpler” program might work more consistently.