

Advanced Clipper

Version 1.1.0

Welcome

Thank you for downloading this fine plug-in. **Advanced Clipper** let's you clip your without generating hidden inter sample peaks.

In order to get the most out of the **Advanced Clipper**, please spend a few moments reading this brief manual.

License

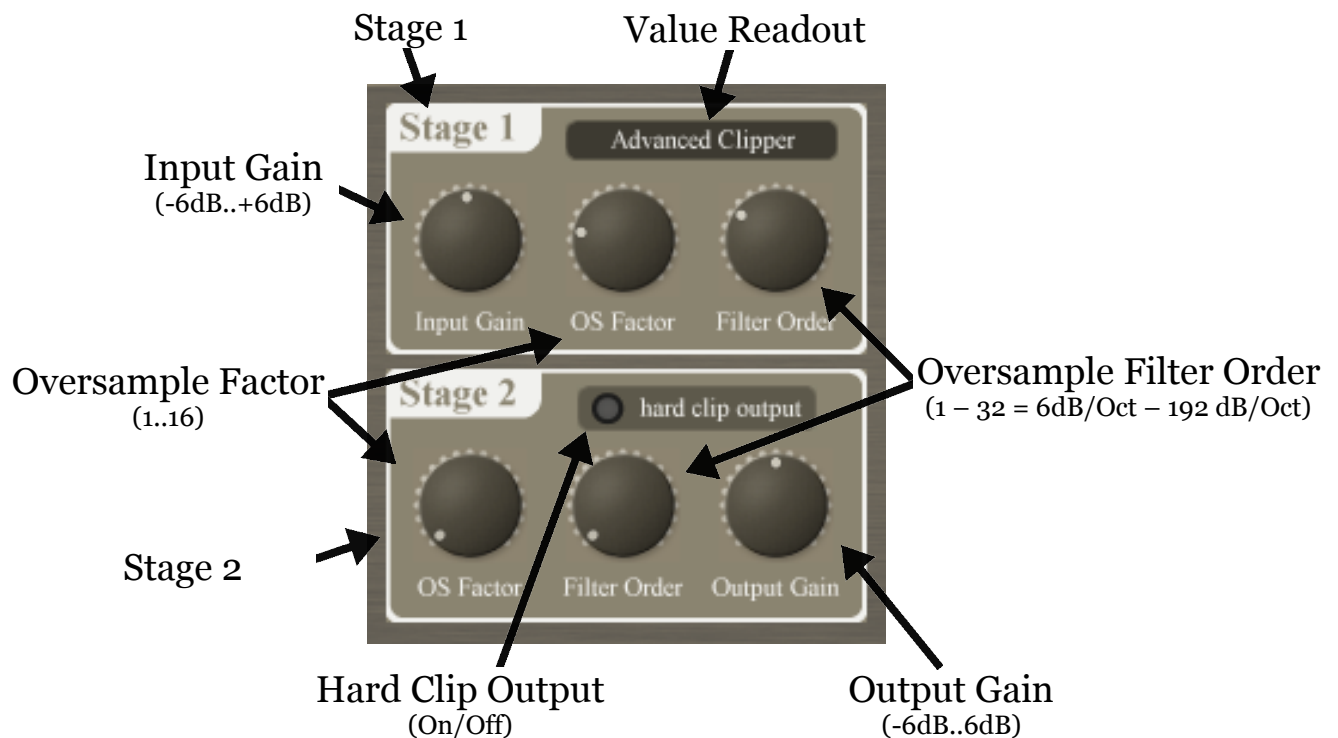
The pre-compiled **Advanced Clipper** has a very simple license:

1. **Advanced Clipper** is freeware. This means that you are free to distribute it, give it to friends, or otherwise share it around. However, only the entire unaltered archive, including this document, may be re-distributed.
2. Copyright of the code and the finished plug-in remain the property of the *Delphi ASIO & VST Project* and namely *Christian-W. Budde*.
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4. Magazine editors are welcome to include the plug-in on cover mount discs or similar media; however, I request that am informed about it via [e-mail](#). A few copies of the publication are always appreciated, but not expected.

User Interface

The user interface shows all adjustable parameters including a readout for every value. There are no meters available to maintain the lowest possible CPU usage without wasting too much CPU cycles. Either a dedicated analyse plugin or the build in meters can be used for this task.

Here is a commented screenshot:



The plugin features 2 independent clipping stages and an additional hard clipper on the output. Each stage contains an adjustable oversampling filter. Furthermore in the first stage the input gain can be controlled and in the second stage the output gain can be controlled.

In the first stage a field can be found that displays a readout of the parameter currently adjusted. After opening the plugin it reads 'Advanced Clipper' as shown above in the commented screenshot.

The dials can be adjusted by clicking and dragging up and down on a dial. To reset the dials to their defaults hold the [Ctrl] key while clicking on the dial. Holding the [Shift] key enters the fine tune mode.

The parameters

This plugin features 7 adjustable parameters in two categories. The categories represent the two stages '**Stage 1**' and '**Stage 2**'. Both stages are equal in terms of the controls, so only the parameters of a single stage are described in detail here.

Input Gain

To push the incoming signal a bit more to the limit or to experiment the effects of clipping, the input can be gained by up to 6 dB. Or, if the input is already too loud it can be attenuated likewise of up to 6 dB.

OS Factor

This parameter controls the oversampling factor. A factor of '1' means no oversampling takes place. For any larger factor, an internal signal at a higher samplerate is interpolated, which is clipped instead. This method can cover inter sample peaks effectively

Filter Order

The interpolation filter order can be adjusted to attenuate unwanted higher order harmonics above the original samplerate, that would otherwise mirror back into the audible range non-linearly (also known as aliasing). A higher attenuation of harmonics unfortunately comes to the expense of a group delay that can generate new clippings during down sampling. Adjust this parameter with care. The effect might be subtle though.

Hard Clip Output

Even after two stages of clipping there still might be some clippings in the signal due to the oversampling filter. Using the 'Hard Clip Output' switch they can be clipped to the ceiling.

Output Gain

In case the signal is needed at a different level than full scale, it can be adjusted between a range of -6dB to +6dB.

Usage

In case of a digital signal there exists not only typical peaks, that can be visualized by simple peak meters, but also inter sample peaks. They appear between the samples and get audible in the DA converters low pass filter that can be found in every CD player for example. They are quite hard to detect, since every DA converter might have slightly different lowpass filters. There exists some useful tools to visualize them. However, this is only the half rent. The next step is to eliminate them without losing too much of the dynamic resolution.

The **Advanced Clipper** uses several stages to get rid of inter sample peaks. Therefore, the signal can be oversampled several times, revealing inter sample peaks. They can be removed completely at this higher samplerate. Though, new inter sample peaks can be introduced during downsampling. For typical settings (not too steep oversampling filter order) they should occur less than present in the input signal. Repeating the steps two times and applying a hard clipper at the output should reduce the number of intersample peaks dramatically if everything is done correctly.

A higher oversampling rate catches the inter sample peaks more accurate and gives the oversampling filter more space to filter out the harmonics, but also increases the CPU usage. Higher filter orders can remove unwanted higher order harmonics better, but lead to new inter sample clippings. To reduce these effectively the second stage should be driven less aggressive than the first stage (lower filter order, high oversampling rate).

Also dial in a very small headroom (like -0.02 dB) before you apply the hard clipper.

There might be several strategies to clip the audio signal right, but here's a recommendation: Use the first stage to roughly clip the audio signal without alias. Dial in a decent oversampling (like 4x) and a rather high filter order (e.g. 48 dB/oct). Due to the high filter order, the output can contain inter sample peaks and even normal peaks. Use the second stage to smooth these out. A high oversampling factor (8x) and a low filter order (6 dB/oct) should work satisfying. Reduce the output level to have some minimal headroom (-0.02 dB as mentioned above). Enable the hard clip output switch to kill the last remaining clippings.

Feedback / Bug Reports

I am always eager to hear feedback or have bugs reported. The easiest way is to send me a mail to: Christian@aixcoustic.com

Furthermore feel free to download the source code, that can be found in the [Delphi ASIO & VST Project](#) at sourceforge.net.

Version History

1.0.0	First release!
1.1.0	Manual added

Credits

- Programming: Christian W. Budde
- Additional Framework Programming: Tobias Fleischer, Maik Menz
- Special Thanks: Swen Müller, Duncan Parsons, Laurent de Soras
- Documentation based on a template by Greg Pettit

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