Sonoris Linear Phase Equalizer 2.1 VST

User Manual

Introduction

© 2008 Pieter Stenekes
info@sonoris.nl

Thank you for choosing Sonoris Equalizer!

What is it?

What is it? The Sonoris Linear Phase Equalizer (SLPQ) is a parametric linear phase equalizer in VST format. The plugin is suitable for mixing and especially mastering and features 7 bands, including lowpass, highpass, peaking and shelving filters. The SLPQ can be used to enhance or correct difficult material like vocal or instrumental soloists and groups, orchestral recordings and complex mixes, without introducing any unwanted coloring. The linear phase implementation of the SLPQ ensures a transparent character and just boosts or cuts a frequency range without adding a "sound". It doesn't smear transients or create mud, nor does it alter the imaging and depth information of the original sound. This way it is possible to boost or cut more than with a conventional equalizer without any of these negative side effects.
The linear phase algorithm of the SLPQ is based on a technique called "backward-forward filtering", until now only implemented in some expensive high end equalizers. The main advantage of this technique is that IIR filters can be used instead of FIR filters, the latter is commonly found in linear phase implementations. IIR filters are known for their more analog kind of filtering and are also more efficient than FIR filters. The filters used in the SLPQ are actually the same as in the Sonoris Equalizer and has also correct gain up to Nyquist.
Every band can be set up to process stereo, L , R or M(id) and (S)ide channels. Processing the mid or side information can be very useful in certain situations. In mastering for example, it allows you to enhance a centered vocal while leaving the other instruments untouched. Or to center a bass without losing the stereo imaging of the rest. Adjustment of is made easy because the SLPQ allows for monitoring the LR or MS channels.
The plugin has a large graphical display that shows exactly what you get. The Sonoris Linear Phase Equalizer also has an automatic upsampling mode. In this mode, the SLPQ has an even more accurate response, especially at the higher frequencies.

About the plugin

Much effort has been put in maintaining the sound quality. All calculations are performed with 64 bit resolution, integer math, except for the filter code.
The result is converted to the 32 bit floats VST format with the use of TPDF dither to preserve as much information as possible, OR to 64 bit floats undithered for VST 2.4 compatible hosts.

Basic Operation

All knobs can be controlled with the mouse by dragging up or down. To increase the resolution press the shift key while dragging. Pressing control while clicking on a knob resets the knob to the default value.
An added feature is the possibility to adjust the knobs with the mousewheel. The shift key also controls the resolution of the mousewheel movement.
Clicking on a value field enables the direct entry of a value.

About this helpfile

This helpfile explains all settings and options to get started. Basic knowledge of parametric equalizers is needed.

Have fun!

Features

- 7 band parametric linear phase equalizer
- Lowpass and highpass filters up to 48 dB/octave
- Peaking and shelving filters
- Stereo, L, R or MS processing and monitoring
- No pre-warping effects, that is, correct response up to Nyquist
- Auto upsampling if needed for an even more accurate response
- Soft Engage technology for the peaking and shelving filters to prevent pops and crackles
- A/B comparison
- Large graphical display
- Three zoom levels
- High resolution level meter
- Full automation possible
- Mousewheel support
- Settings can be saved and loaded
- 64 bit integer math in audiochain, except for the filter code
- TPDF dither is used to convert the 64 bit signal back to 32 bit floats
- 64 bit undithered output in VST 2.4 compatible hosts
- Installer / uninstaller

A/B

This button switches between settings A and B for comparison. All adjustments are automatically saved to the current set. Copying from one set to another is possible by pressing the control key and the A/B key at the same time.

This setting is automatable.

B1 – 7 buttons

These buttons switch the filters on and off. For the bands 2 to 6 Soft Engage technology is used, that guarantee no pops and crackles. This technology is also used when adjusting the other controls of this bands. Soft Engage is a Sonoris proprietary technology that prevent sudden changes in the audio data that could cause unwanted noise.
Right clicking the button switches the band processing between LR(stereo), (L)eft, (R)ight, (M)id and (S)ide.

These settings are automatable.

B2, 3, 5 and 6 filter type buttons

These buttons selects between the peaking and shelving filter types for band 2, 3, 5 and 6.

This setting is automatable.

Bypass

This button switches bypass on or off. The bypass is a soft bypass, disabling the filters, the volume stage and the monitor setting.

This setting is automatable.

Gain

This setting sets the gain of the filter. The gain ranges from –18 to +18 dB.

This setting is automatable.

Freq

The freq value sets the resonance or cut-off frequency for the chosen filter. This setting ranges from 16Hz to 20KHz.

This setting is automatable.

Bandwidth / Slope

This sets the bandwidth for the peaking filters and the slope for the shelving and LP/HP filters. The bandwidth is scaled in octaves ranging from 0.1 to 4.0. For the LP/HP filters the slope is measured in dB/octave. For the shelving filters the slope is a Q-factor. The slope ranges from 0.1 and 4.0 too.

This setting is automatable.

Graph display

The graph display shows the magnitude response of the current setting. This display updates whenever a setting is changed.
All filter bands have a little square in the same color as the band name. You can drag these squares up and down with the mouse to change the gain of the filter. When you hold the shift key while dragging, you can change the frequency setting. Dragging with the control key changes the bandwidth or slope.
Clicking on the graph display switches between the three zoom levels.

These setting are automatable.

Monitor

Chooses between the following monitoring modes:

- All: displays all curves and monitoring is stereo output (all colors)
- LR: displays LR curves and monitoring is stereo output (red)
- L: displays L curves and monitoring is left channel output (blue)
- R: displays R curves and monitoring is right channel output (cyan)
- M: displays M curves and monitoring is mid channel output (yellow)
- S: displays S curves and monitoring is side channel output (green)

Toggle between modes by clicking on the small arrows or by clicking in the middle of the textbox. When you click at the left of the textbox, a menu appears. You can also use the mousewheel.

This setting is automatable.

Trim fader

With this fader you can adjust the output gain of the equalizer. The gain ranges from –18 dB to +18 dB. You can also use the mousewheel.

Holding control when left clicking on the fader resets to the default setting.

This setting is automatable.

Metering

The level meters show the real-time level of the monitored signal.

When a level reaches 0 dB the clip indicator lights up. You can reset this indicator by clicking on it.

Options

In the option menu are four choices:

- Help: Shows this helpfile (Windows version only)
- Quality: the quality setting controls the length of the buffer that is used to create the linear phase. The higher the quality, the longer the buffer and the higher the latency. Auto mode sets the lowest quality for playback and the best quality for rendering, in compatible hosts only
- Load presets. This function allows you to recall previous settings stored in a file
- Save settings. This function allows you to save the current settings to a file

