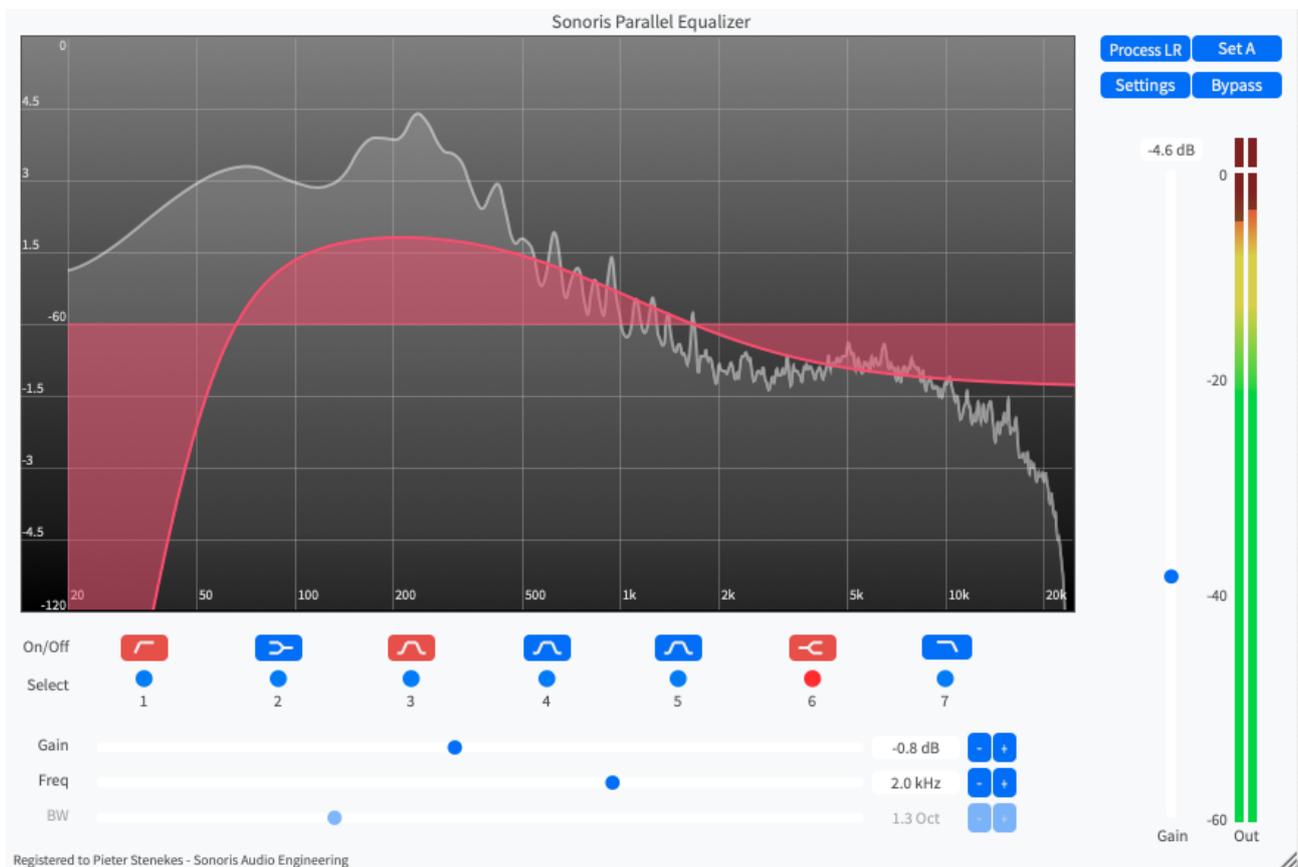


# Sonoris Parallel Equalizer 1.2

## User Manual



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# Introduction

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Thank you for choosing Sonoris Parallel Equalizer!

## What is it?

The Sonoris Parallel Equalizer (SPEQ) is a minimum phase parametric equalizer in VST, VST3, AU and AAX format for Windows and Mac. The plugin is suitable for mixing and especially mastering. It has fixed filter types on each band and band 2 to 6 are connected in parallel. This may be familiar to you, some analog eq's have the same topology. The filters are built in a feedback/feedforward manner to provide cutting and boosting. Band 3 to 5 are constant-Q bells and 2 and 6 are shelves. Band 1 and 7 are low/high cuts and connected in series with the rest.

The main feature is that this eq is a lot more subtle and it needs more work to set up right because there is some interaction between the bands. In normal series connected eq's the gains of two bands just add up but in a parallel eq it doesn't. The filters are parallel to each other as well as the unprocessed signal and this makes that if you have for example a low shelf boost and switch on a low frequency bell boost too, it just doesn't do much more. You can't easily overdo things, but have to work harder to get it right. Also, the signal path is cleaner because the filters all work on the same input. The filters have correct gain up to Nyquist. The plugin has a large graphical display that shows exactly what you get. The size of the plugin can be changed to your liking by dragging the lower right-hand corner.

## About the plugin

Much effort has been put in maintaining the sound quality. All calculations are performed with 64 bit resolution internally.

## Basic Operation

All controls can be controlled with the mouse by dragging or clicking. To increase the resolution press the shift key while dragging. The sliders have a snap function that allows for a quick reset to the default value. An added feature is the possibility to adjust the knobs with the mousewheel. Clicking on a value field enables the direct entry of a value.

## Activation

Details on how to activate the plugin can be found here:  
<https://www.sonorissoftware.com/activation/>

## About this helpfile

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

Windows is a registered trademark of Microsoft Corporation in the United States and other countries. Mac and the Mac logo are trademarks of Apple Computer, Inc. registered in the U.S. and other countries. VST/VST3 is a trademark of Steinberg Media Technologies GmbH. AAX is a registered trademark of Avid Technology, Inc.

## Features

- ✓ 7 band parametric minimum phase equalizer with parallel connected bands
- ✓ Feedforward / feedback topology
- ✓ Lowpass and highpass filters up to 24 dB/octave
- ✓ Constant Q bells, shelving, low and high cut filters
- ✓ Stereo, L, R or MS processing
- ✓ No pre-warping effects, that is, correct response up to Nyquist
- ✓ Soft Engage technology to prevent switching noise
- ✓ A/B comparison
- ✓ Large graphical display
- ✓ Spectrum analyzer
- ✓ Three zoom levels
- ✓ High resolution level meter
- ✓ Full automation possible
- ✓ Mousewheel support
- ✓ Settings can be saved and loaded
- ✓ Double precision internal resolution and interfacing with compatible DAWs
- ✓ Windows 32/64 bit VST, VST3 and AAX versions
- ✓ Mac OS 64 bit Intel / Apple Silicon VST, VST3, AU and AAX versions
- ✓ 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. Soft Engage technology is used, that guarantee no pops and crackles during switching. Soft Engage is a Sonoris proprietary technology that prevent sudden changes in the audio data that could cause unwanted noise.

These settings are 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  $-12$  to  $+12$  dB. Double clicking the thumb will reset the slider to 0dB.

This setting is automatable.

## Frequency

The frequency 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 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.

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 dot with the band number written on it. You can drag these dots up and down with the mouse to change the gain and frequency of the filter. As soon as you click a dot, the sliders will load the corresponding settings.

To affect the gain only, hold the ctrl key during dragging. To affect the frequency only, hold the shift key during dragging. Holding the ctrl AND shift key affects the BW or slope.

Clicking on the graph display switches between the three zoom levels.

These settings are automatable.

## Process

Chooses between the following processing modes:

- LR: processing the stereo input
- L: processing the left channel input
- R: processing the right channel input
- M: processing the mid channel input
- S: processing the side channel input

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. Double clicking the thumb will reset the slider to 0dB.

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.

# Settings

In the settings menu are these choices:

- Load. This function allows you to recall previous settings stored in a file
- Save. This function allows you to save the current settings to a file
- Analyzer: Toggles spectrum analyzer
- Help: Shows this helpfile
- About: Shows the about box with registration info.
- Show tooltips: toggles tooltips on or off