

SLICE EQ

OPERATOR'S MANUAL

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Glossary

DAW

Slice EQ is not a standalone program, it needs to be used in conjunction with a host application where it is loaded as a plugin. These programs are usually called "DAWs" (Digital Audio Workstations) or simply "hosts". There are many different DAWs available on the market, and Slice EQ works with those that support either the VST 2, AAX or the Audio Unit plugin architecture.

Patch

A patch is a configuration of Slice EQ that gives it a certain sound. Whenever you pull a slider or turn a knob you modify the current patch.

Preset

Presets are ready-made patches which ship with Slice EQ. Presets also make good starting points for your own patches.

Snapin

Some of Kilohearts' products can host other plugins called snapins. While Slice EQ works as a standalone VST, AAX or Audio Unit plugin it can also be loaded as a snapin in for example Multipass.

System requirements

These are the minimum recommended system requirements for running Slice EQ.

CPU

2 GHz or faster

Memory

1 GB or more

Operating System

Windows (7 or newer) or Mac OS X (10.7 or newer)

Software

A DAW supporting VST 2, AAX, or Audio Unit plugin standards.

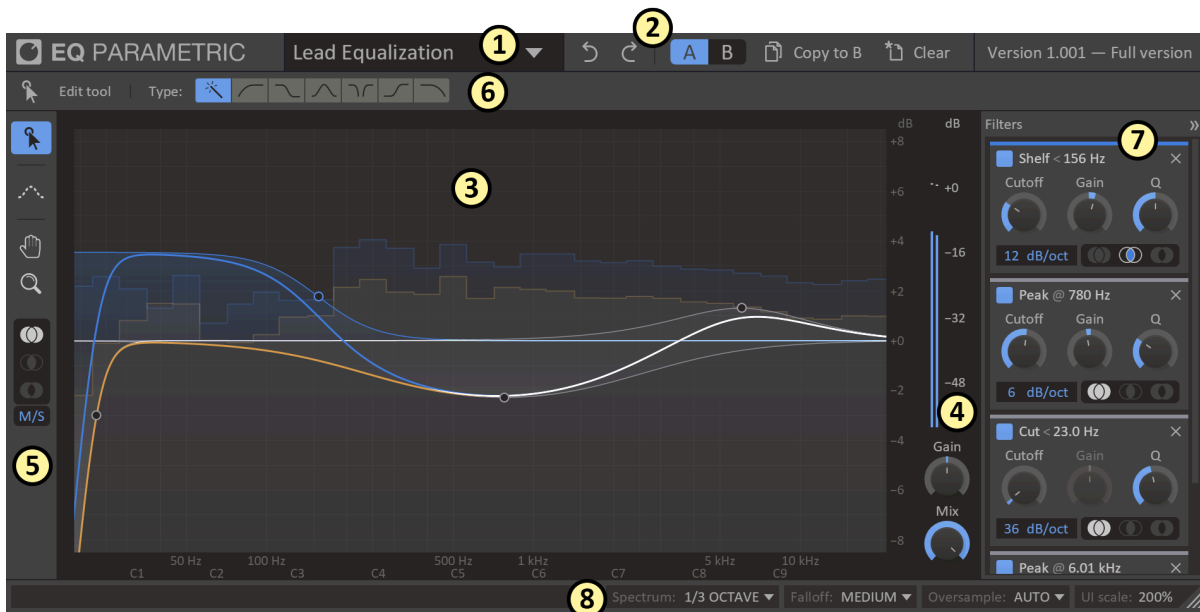
If you have many instances of Slice EQ running, or use many filters at the same time in your patch the CPU usage will increase accordingly. Thus, we cannot guarantee that Slice EQ will work flawlessly in all use cases even if your system does meet the minimum recommended system requirements.

Overview

Kilohearts' Slice EQ is a powerful parametric equalizer, supporting six different filter types with full control over filter roll-off and Q. Stereo processing is possible with both **Left/Right** and **Mid/Side** modes.

Slice EQ has a workspace centric user interface which gives the user the option between either editing the filters visually in a spectrum view, or controlling each filter parameter individually by turning knobs.

User interface



This is an overview of the plugin UI. It may look a bit overwhelming at first, but let's try to break it down a bit.

1. In the top bar the name of the **current patch** is shown. By clicking the patch name you will open up the **preset browser** from where you can save and load presets.
2. The top bar also holds controls for **Undo** and **Redo** as well as the **A/B** configuration selector, which allows you to quickly switch between two different equalization setups to compare them.
3. In the middle of the UI you will find the main work area. It shows all the filters curves and their combined equalization curve, as well as a real time spectrum analyzer of the processed audio.
4. In the rightmost part of the work area you will also find a large VU meter measuring the overall volume of the processed audio, together with controls for overall gain and dry/wet mix.

5. To the left of the main work area you will find a toolbar with small selection of tools. It also holds controls for which stereo channel you are editing, and whether the equalizer works in L/R or M/S mode.
6. Just above the work area is a small bar containing options for the currently selected tool.
7. To the right of the work area is a list of all the filters in the current patch. This list is hidden by default and can be expanded by clicking on the panel labelled **Filters**.
8. The very bottom of the window holds the status bar, which gives you tips about whatever your mouse is over at the moment. It also has controls for changing the appearance of the spectrum, changing oversampling, and resizing or scaling the UI.

Operating the controls

Most parameters of Slice EQ are controlled by the knobs and the sliders seen in the UI. To move a knob or slider simply click on it, and while holding the mouse button down move the mouse up or down.

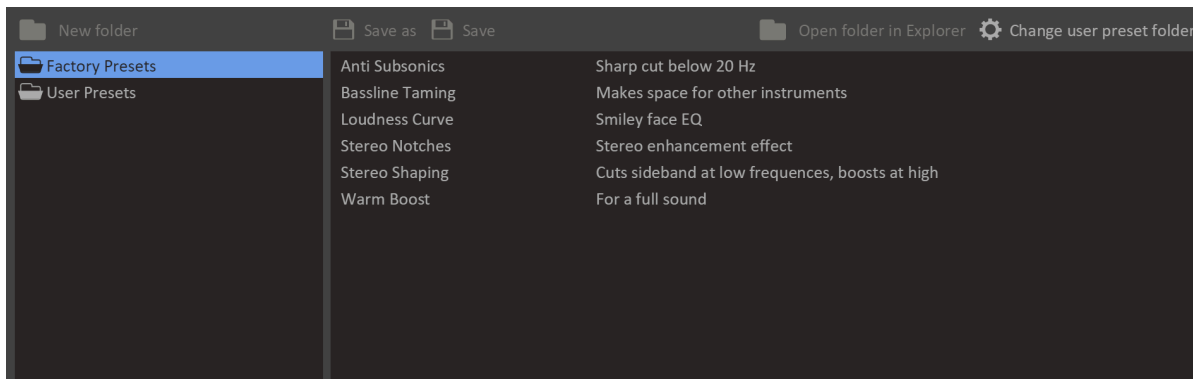
Sometimes you might want more precise control when tuning a parameter. Hold the **shift** key while moving a knob or slider to enter **fine tuning** mode, where the knob or slider will move more slowly.

You can reset a knob to its default position by double clicking it.

Finally, most controls support entering the value using your keyboard by right-clicking on them.

Working with patches

The first thing you want to do after installing Slice EQ is probably to try out some of the presets it comes with. To reveal the preset browser, click on the name of the current patch in the top most part of the UI.



Loading a preset in the preset browser is as easy as clicking on it. The preset browser also allows saving presets, and organizing them into folders. Presets are divided into two categories, **factory presets** and **user preset**.

Factory presets are included with Slice EQ or downloaded using the Kilohearts installer. Factory presets are write protected to protect them from accidental corruption.

The user presets folder is where you as the end user should store your own presets. By default this folder is located in your documents folder, but you are free to relocate it elsewhere.

Building your sound

Once you have checked out some of the presets you are probably eager to get your hands dirty and create your own patches. We designed Slice EQ to be easy and fast to work with, and we hope you will enjoy experimenting and discovering the possibilities. Still, knowledge is power, so in this section we will go through all the features of Slice EQ in detail.

A parametric equalizer, like Slice EQ, works by combining the effect of several simple filters into a complex equalization curve across the whole spectrum. Slice EQ supports the following 6 different filter types:

Peak

Boosts or attenuates sound around a certain frequency.

Low & High Shelf

Boosts or attenuates sound below or above a certain frequency.

Notch

Completely cuts out sound around a certain frequency.

Low & High Cut

Completely cuts out sound below or above a certain frequency.

For each filter, the following parameters can be controlled independently:

Cutoff

The center frequency for the filter. For example, the peak filter boosts or attenuates sound close to this frequency.

Gain

How much the filter boosts or attenuates the sound. Not available for **Notch** and **Cut** filter types.

Q

Adjusts the bandwidth of the filter. Increasing Q will create a sharper peak around the cutoff frequency.

Roll-off

The roll-off, measured in decibels per octave, dictates how sharp the transition from stopband to passband is for a filter.

Edit tool



The first tool in the toolbar is the edit tool, and this is what you will use most of the time to add new filters and edit them.

To add a new filter, first select the filter type from the palette in the tool options, then click and drag the zero decibel line to add a new filter of that type. When the magic wand icon in the filter type palette is selected the filter type will be decided automatically depending on where on the frequency axis you click. A small icon above the mouse cursor will show you what filter type will be added.

Filters are edited by dragging their handles. When dragging a handle left and right you will change the cutoff frequency for that filter. When dragging up and down you will change either the gain (for peak and shelf filters) or Q (for cut and notch filters). To change Q for any filter type, hold the **Alt** key while dragging, or scroll the mouse wheel while hovering a handle with the mouse cursor. Adjust the roll-off by holding the **Ctrl** key and dragging up and down. To increase precision while dragging any parameter hold the **Shift** key.

The Q of High & Low Cut filters will snap to a value of 0.707, or about -3 dB, when dragging their handles. This is the Q value where the filter cuts as steeply as possible without overshooting and boosting any frequency. Additionally, High & Low Cut filters with a Roll-off of 6 dB/octave will have their Q locked to this value for technical reasons.

Several filter handles can be selected at once by clicking and dragging a selection rectangle around them. They can then be dragged together.

To remove one or more filters you must first select them and then delete them by pressing the **Delete** key on your keyboard.

Pan view tool

The pan view tool is used to pan the viewport along both the frequency and the gain axis. Simply click and drag to pan the view.

It is also possible to pan the view with any tool selected by clicking and dragging on the axis labels below and to the right of the viewport.

Zoom view tool

The zoom view tool is used to zoom the viewport along both the frequency and the gain axis. Simply click and drag to zoom the view.

It is also possible to zoom the view with any tool selected by scrolling the mouse wheel when hovering the axis labels below and to the right of the viewport.

Reference tool

The reference tool allows you to display an overlaid spectrum of another sound in the main work area. This can be useful in many ways, for example when trying to match the equalization curve of one song to another one, or during mixing when you want to avoid having two tracks clash in the same frequency range. Reference spectrums can be recorded and later loaded from **profiles** which can be easily shared between different sessions.

The reference data can come from many different sources:

Off

Turns off the reference spectrum.

Sidechain

Displays the spectrum of the sidechain input in real time.

Factory profile

Load one of the reference profiles bundled with the plugin.

User profile

Load one of the reference profiles you saved earlier.

Create profile from file

Create a reference profile from audio in a **wav** file.

Record

Record the current (main or sidechain) input to a profile.

Edit mode & Stereo mode

Below the tools in the toolbar you will find controls for handling equalization in stereo.

At the bottom is a dropdown that lets you choose if the equalizer works in **Left/Right** or **Mid/Side** mode. The Left/Right mode will let you apply different equalization to the left and the right channels. The Mid/Side mode, on the other hand, will let you equalize the mono content (mid) separately from the stereo content (side).

On top is the edit mode selector. It lets you choose what channel to edit with the edit tool. By default it is set to "both channels", but you can change it to left, right, mid, or side to only edit that channel. Instead of clicking, you can also toggle the current edit mode with the **Tab** key.



Gain & Mix

In the bottom right corner you will find the **Gain** and **Mix** knobs.

The Gain knob adjusts the overall gain of the equalizer. You can use this to compensate for your equalization inadvertently making the overall volume louder or softer.

The Mix knob scales the Gain parameter of all filters, scaling it down to 0 dB at 0% mix. Low cut, high cut and notch filters are not affected.

Filter list

The filter list is found at the far right part of the user interface. The list is hidden by default and can be expanded by clicking on the panel labelled **Filters**.

The filter list gives you access to each parameter of every filter via a knob. You can also bypass filters in this list by clicking on the toggle button in the top left corner of each filter. Filters can be removed by clicking the X button in the top right corner.



Spectrum view settings

In the status bar you will find two dropdowns which can be used to tune the look of the spectrum.

Spectrum mode

Adjusts the frequency resolution of spectrum analyzer. Select between **octave** or **1/3 octave** bars, and **exact** which shows a continuous curve.

Falloff

Adjusts how quickly the spectrum falls off to zero when the sound goes silent.

Oversampling

The equalizer can optionally oversample the signal in order to provide filter responses shaped closer to their analog prototypes. By default the oversampling is enabled automatically when the sample rate is low enough. You can override this behaviour to force oversampling on or off using the dropdown in the status bar.

Acknowledgements

This development of this product was helped by the following pieces of excellent open source software:

Boost C++ Libraries

Skia Graphics Library
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Symbiosis AU/VST
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