apulSoft apQualizr v1.4 Manual

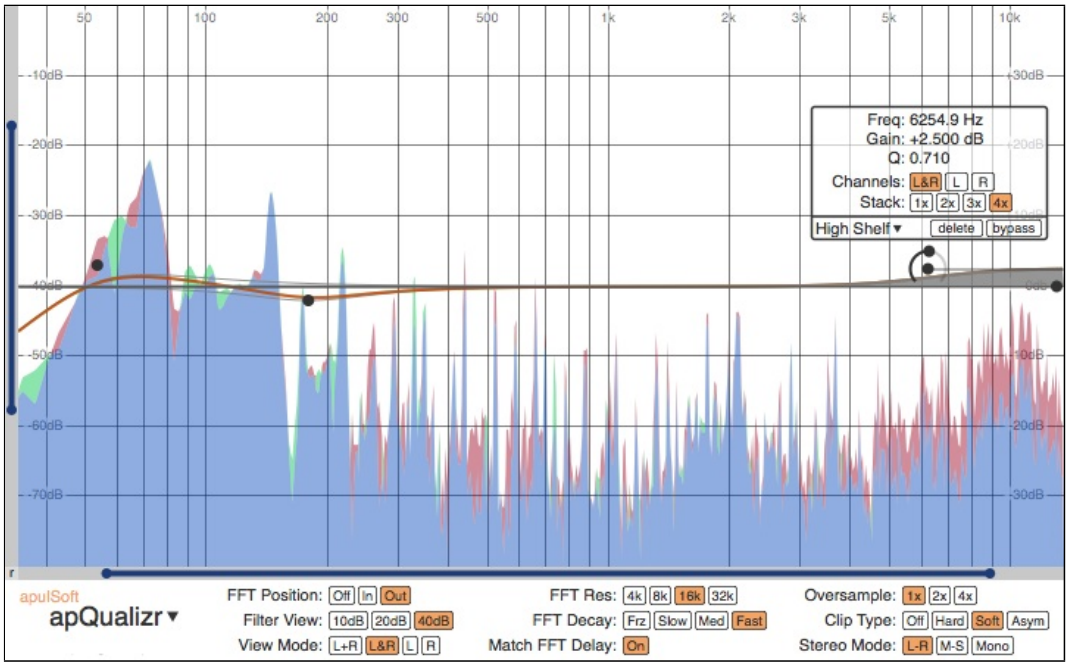
High-Precision Equalizer

(VST/AU Mac OSX , VST Windows)

http://www.apulsoft.ch

VST plugin technology by Steinberg.
AU plugin technology by Apple.
Manual written with LaTeX on January 5, 2010

1 Introduction



apulSoft apQualizr is a stereo equalizer/analyzer intended to operate as precisely as possible while letting the user see exactly what is going on. This goal is achieved by these main features:

- 64 bit internal precision
- 2x and 4x oversampling
- Unique filter stack feature to improve filter results
- Output hard- and softclipping built in
- Variable number of filter bands (up to 63)
- Up to 40 dB of gain per filter band
- Independent stereo filtering
- High resolution FFT analyzer with zooming
- Large analyzer display
- Graphical editing of filters on top of the analyzer display

Contents

2 End User License Agreement

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3 System Requirements

- Mac OS X
  - A computer running Mac OS X 10.4 or better (G5 or intel mac recommended).
  - A VST or Audio Units host.
- Windows
  - A computer running Windows XP or better ($>$1.5 GHz CPU recommended).
  - A VST host application.
  - An application to view pdf files to read the manual (Acrobat Reader).

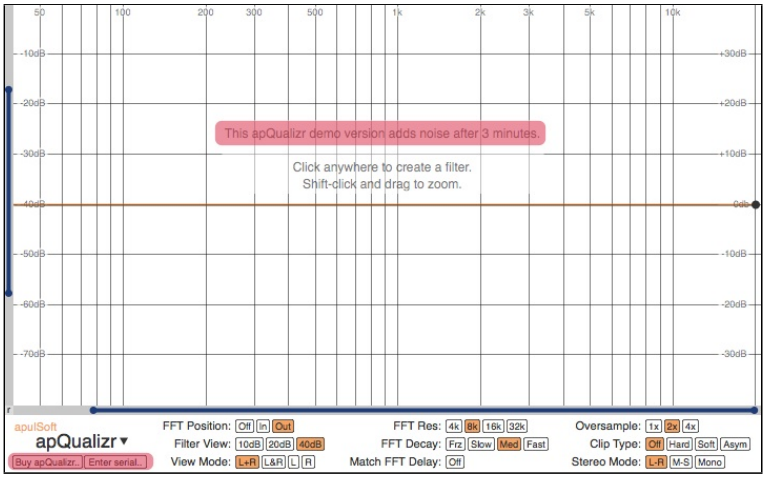
4 Installation

- Mac OS X
  - Close all VST and Audio-Units hosts.
  - Double-click the downloaded apQualizr_v###.dmg to mount the disc image.
  - Double-click ”Install apQualizr” to launch the installer.
  - Open a host of your choice.
  - apQualizr will run in demo mode until you buy a serial and enter it via ”Enter Serial...” (see section 6)
- Windows
  - Close all VST hosts.
  - Double-click the downloaded apQualizr_v###_setup.exe to launch the installer.
  - Click the **Install** button and let the installer complete its work.
  - Open a VST host.
  - apQualizr will run in demo mode until you buy a serial and enter it via ”Enter Serial...” (see section 6)

5 Limitations of the demo mode

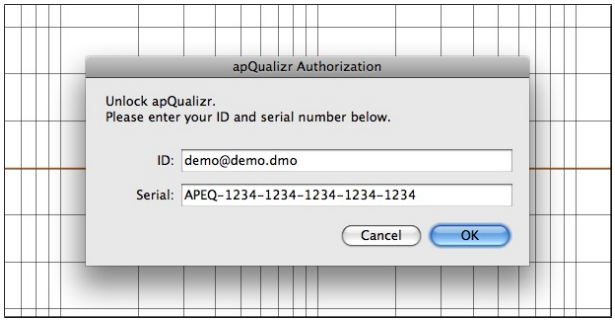
- After three minutes, apQualizr in demo mode will start to insert faded bursts of white noise. Every instance of apQualizr has an independent noise timer.

6 Buying apQualizr and unlocking the full version



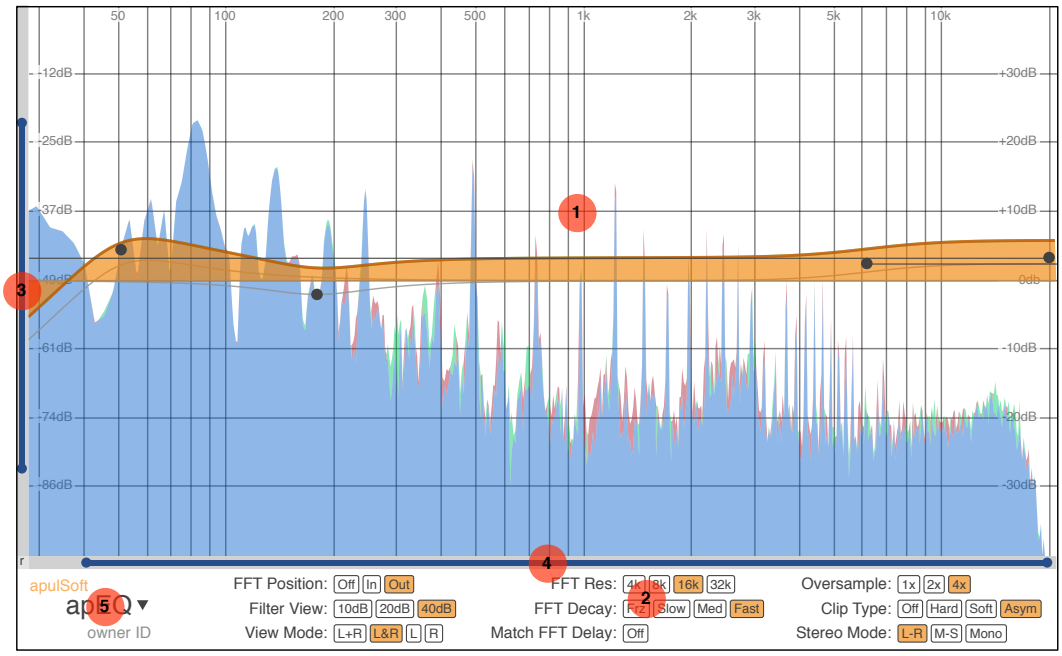
On first launch, apQualizr runs in demo mode. The demo limitations text will be visible at the center of the display. In order to buy apQualizr, click the **Buy apQualizr..** button (bottom left corner). This will open a Webbrowser and open the apulSoft apQualizr website. If you want to use apQualizr on a computer without internet connection, you can buy a license on another computer by browsing to `http://www.apulsoft.ch/apQualizr`.

Click the **Buy apQualizr** button on the website to buy an apQualizr license.
After successfully completing the buying process, you will receive an Email from Share-It containing your serial info. In the apQualizr interface, click the **Enter serial..** button.



Enter the ID and serial number exactly as received in the Email. In most cases, copy/paste can be used - however there are some plugin hosts that block plugins from receiving pasted strings. In that case, use the keyboard. Once the input is complete, click **Ok**. If the ID/serial info was correctly entered, the unlock box will disappear and apQualizr will run in full mode.

7 apQualizr Controls



The apQualizr graphical user interface consists of five main sections:

1. apQualizr display
   The main working area of apQualizr where filters are created and edited. At the same time it is the apQualizr frequency analyzer.

2. Global settings
   Radio buttons for all global settings.

3. Analyzer dB zoom slider
   Controls the decibel range visible in the analyzer.

4. Frequency zoom slider
   Controls the frequency range visible in apQualizr for both the analyzer and the filters.

5. apQualizr title with menu, buy/enter serial buttons (demo version)

7.1 apQualizr display

apQualizr's main control is the apQualizr display which consists of multiple layers. From back to front:

frequency/gain grid There are logarithmically spaced vertical guidelines for frequency with hertz labels at the top. The frequency range displayed depends on the host sampling rate. The graph starts at 30 Hertz and extends to almost half the sampling rate or 30 kHz for high sampling rates.
The horizontal gain guide lines are labeled on both sides. The left side labels are analyzer gain values. They depend on the analyzer gain slider which is located to the left of the labels. The right side labels are filter gain labels with +0 dB at the center of the view. The range of the gain guidelines is determined by the **View Range** global setting.

demo/filter creation text overlay If apQualizr is running in demo mode, a text box stating that fact is displayed on top of the guidelines.
The same box displays a filter creation hint as long as no filters have been created.
If apQualizr runs in full mode and filters are present, no text overlay is displayed at all.

frequency analyzer graph The analyzer frequency curve is drawn transparently onto the gain/frequency grid. Its main color is blue. If the **View Mode** option is not **L+R**, green is used for the left channel of the signal and red for the right channel. The range of decibels and frequencies displayed by the graph can be adjusted with the db and frequency sliders to the left and bottom of the display.

filter curves and handles All currently not edited filters are shown by a black circular handle and a curve showing their frequency response. Clicking a handle selects its filter for editing. In stereo mode, the filter curves are drawn with different colors depending on channel settings. **L&R** draws gray, **L** green and **R** red.

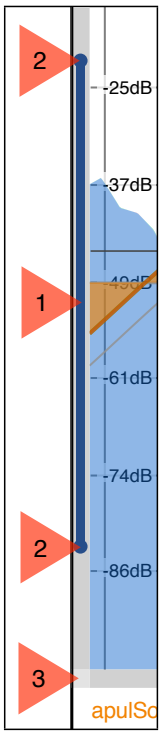
filter sum curve and gain handle The filter sum curve is drawn on top of the filter curves using a filled orange transparent area which shows the overall gain per frequency. The sum uses the same color coding for channels as the filter curves. On the right side of the filter sum curve, there is the master gain handle to change the overall gain of apQualizr.

selected filter curve and filter editor The selected filter draws its curve filled on top of everything else. The filter editor is drawn as a rounded white transparent box with controls inside. The placement of the box is determined by avoiding overlaps with the selected filter curve and the filter sum curve(s) to keep those curve visible while editing. If apQualizr can't fit the filter editor nicely, it gets drawn centered. More information about the filter editor can be found in subsection 7.4.

7.2 Zooming the display

apQualizr's main display can be zoomed horizontally and vertically. Vertical (gain) zoom only affects the analyzer graph, not the filter response graphs! Horizontal (frequency) zoom affects both the filter graphs and the analyzer display. So zoom the filter gain, use the **Filter View** global setting on the bottom of apQualizr.

7.2.1 Zooming with the sliders



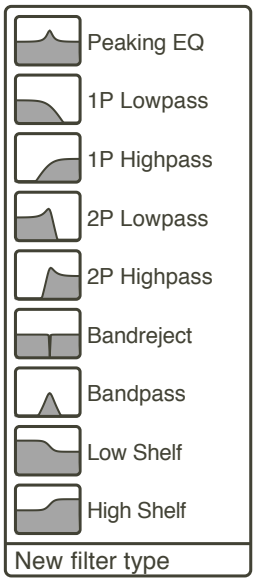
1. Clicking and dragging the middle of the slider handle moves the view without changing the range.
2. Dragging the circular end-knobs of the slider handle only adjusts the minimum or maximum border of the viewed range thus allows to change the zoom factor.
3. Clicking the **'r'** button in the corner of the two view sliders resets the zoom to apQualizr default values. With default values the analyzers covers 80dB of gain range.

7.2.2 Zooming with Shift

A very easy and quick way to zoom to specific regions of the analyzer graph is pressing Shift and mouse-dragging a zoom rectangle in the apQualizr display. This will zoom in to the specified rectangle unless apQualizr is already at maximum zoom. Just clicking the mouse without dragging zooms out 50%.

7.3 Filter Creation

To create a filter, click the apQualizr display anywhere. The x coordinate determines the frequency and the y coordinate the gain of the new filter. The filter type chooser pops up:



Once a filter type is clicked, a new filter is added to apQualizr. More information about the available filter types can be found in section 8.

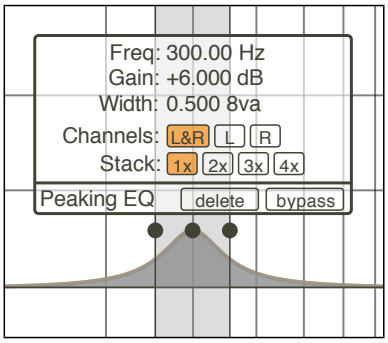
7.4 Editing Filters

All filters are represented by small black circular handles in the apQualizr display. Clicking a handle selects its filter and opens the filter editor. Only one filter can be selected and edited at a time.
The filter handle can be dragged to adjust filter frequency and gain. If a filter handle is dragged outside the visible range, it appears as a hollow circle. Dragging still works on these!
Keyboard modifier keys enable additional options:

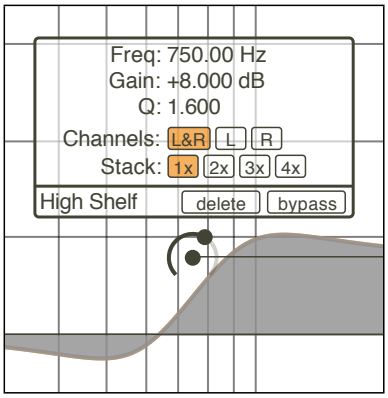
Shift Increase the mouse resolution for small adjustments.

Control Lock editing to one parameter at a time (frequency/gain).

Alt Switch to the ”third” filter parameter. This applies to filter types which have a **Width** or a **Q** parameter. Holding **Alt** during dragging changes that parameter instead of frequency/gain.



For filters with a **Width** parameter, the bandwidth overlay with two circle-handles is drawn to adjust the bandwidth graphically by dragging.



If a filter has a **Q** parameter, a Q handle is drawn around the central filter handle. The Q parameter is adjusted by clicking and dragging up and down.

Numerical parameters on top of the filter editor can also be adjusted by clicking and dragging the number itself.
Alternatively any of the numbers can be double-clicked to enter a new value with the keyboard.

The **Channels** and **Stack** settings are changed by clicking on the desired option.

Channels determines on which channels processing happens. It is only available if apQualizr is running on a stereo track.
Stack changes filter characteristics by stacking multiple filters of the same type on top of each other. It can improve the accuracy of the filter calculations. The higher stack is set, the more cpu the filter uses.

To change the type of an already created filter, click the filter type name on the bottom left of the filter editor (for example ”High Shelf”). This will open the filter type chooser (similar to subsection 7.3) where a new filter type can be chosen.

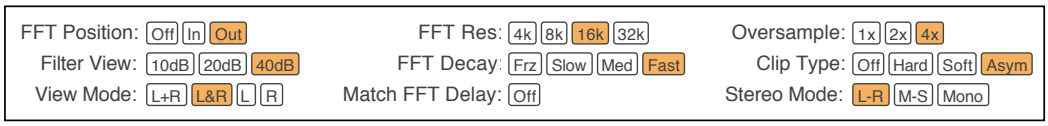
The **delete** button at the bottom deletes the edited filter and closes the filter editor. The **bypass** switch allows bypassing the selected filter.

7.5 Master Gain

The master gain handle at the right side of the display works similar to the filter editor. Dragging it up and down will change the overall gain of the filter sum curve and of apQualizr itself.

7.6 Global Settings

The bottom area of the apQualizr interface consists of a number of radio buttons to change global settings which apply to all filter bands.



7.6.1 FFT Position

This setting determines where the signal path is analyzed to produce the analyzer frequency graph. **Off** means there is no analyzer display which saves cpu cycles as long as the apQualizr editor is open.

7.6.2 Filter View

The decibel range of the filter display. It determines how many decibels are visible for the filter graphs and the filter sum curve. This is independent of the FFT decibel range.

7.6.3 View Mode

This setting is visible if apQualizr runs on a stereo track.

L+R The analyzer and the filter sum curve display a sum of the left and the right channel while the single filter curves use colors depending on their channels setting.

L&R The analyzer and the filter sum curve display both channels at once. The left channel appears green, the right channel red. The single filter graphs use colors depending on their channels setting. This mode leads to more than twice the cpu usage for analyzing and drawing!

L The analyzer and the filter sum curve display the left channel only.

R The analyzer and the filter sum curve display the right channel only.

Note: In M-S stereo mode, the buttons use **M&S** nomenclature.

7.6.4 FFT Res

The number of samples used to generate one frame of the frequency analyzer graph. Higher values mean more cpu usage and better frequency resolution while lower values mean better impulse resolution. **4k** stands for 4096 samples, **8k** for 8192 samples, **16k** for 16384 samples and **32k** for 32768 samples.

7.6.5 FFT Decay

The speed at which the analyzer curve 'falls' downwards. **Frz** means freeze - the curve does not decay at all. Over time the resulting graph is a graph of gain maxima per frequency. **Slow** stands for 8dB/sec, **Med** for 24 dB/sec and **Fast** for 100 dB/sec analyzer gain decay.

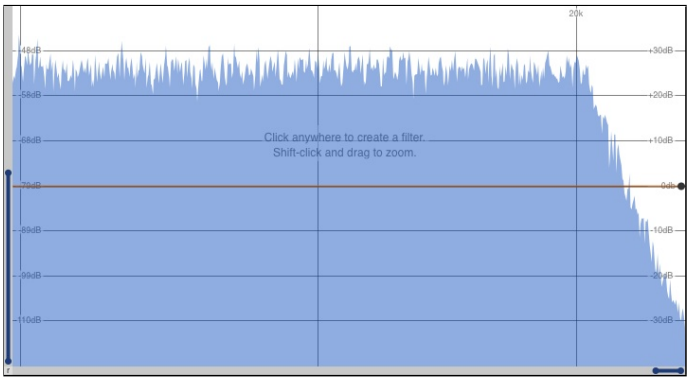
7.6.6 Match FFT Delay

This setting delays the audio running through apQualizr by half the FFT resolution. This leads to the analyzer curve being visually in sync with the audio. Usually the graph appears to be late compared to the audio. At the same time apQualizr reports the delay as plugin-latency to the host. Modern audio applications should be able to compensate for this latency. Some audio applications only reset compensation when audio is stopped and started again, some will not do it at all. This setting is only recommended if the visual delay is a problem.

7.6.7 Oversample

Using oversampling with apQualizr leads to heavy processing to convert the audio stream to a higher sampling rate. All the filters and the final clipper are then calculated at the higher sampling rate. As apQualizr uses high quality sampling rate conversion algorithms, using oversampling leads to a lot of cpu usage. If the host application is already running at a sampling rate of 88 kHz or more, oversampling is most likely not needed. The biggest advantages of using oversampling are more accurate filtering of high frequencies and better sounding output clipping.

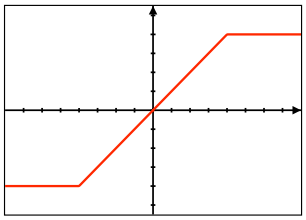
Oversampling involves two sample rate conversions of the audio material and these always need filtering in order to avoid unwanted conversion noise. A filter is required that cuts out everything above the maximum frequency of the end sample rate, which is half the host sample rate. Unfortunately, no perfect filters do exist, so the cutting has to start earlier than that. apQualizr uses a quite CPU expensive and very steep filter during oversampling. It cuts inside 1/20 of the topmost octave of the frequency range. For a 44.1 kHz sample rate this means the cutting begins at 20.05 kHz and reaches non-measurable silence at 22.05 kHz.



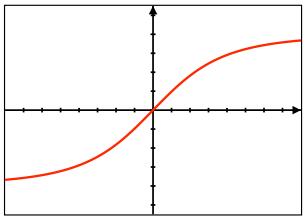
This screenshot shows apQualizr zoomed in to show extreme high frequencies. White noise is fed through and the oversampling filter is visible at work.

7.6.8 Clip Type

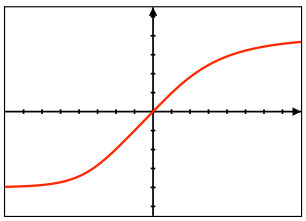
Using filters can lead to a signal boost resulting in very high sample values. Therefore apQualizr includes a shaping/clipping stage to make sure no sample exceeds the maximum value. If **Clip Type** is set to **Off**, nothing will be done to the output. This often leads to distortion in a later stage of the host application's signal path.



Hard clipping just cuts the signal. This results in a lot of artificial overtones, but it also allows the signal to pass completely unaltered as long as there is no clipping.



Soft clipping gradually reduces high output levels. It leads to less overtones than hard clipping and they tend to sound warmer. Soft clipping will kick in quite early and therefore slightly color even signals that are still far away from clipping.



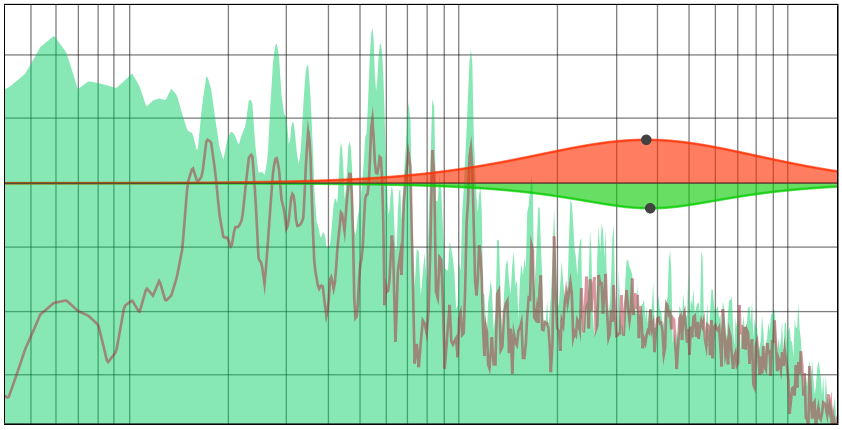
Asymmetrical clipping clips negative sections of the output signal harder than positive ones. This leads to more artificial overtones than soft clipping and a somewhat tube-esque sound.

7.6.9 Stereo Mode

This setting is only visible if apQualizr is processing stereo data. In older vst hosts that is always the case.

L-R apQualizr does regular stereo processing. Filters can be set to process the left or right channel only.

M-S The input signal gets converted from L-R (left-right) to M-S (mid-side). Then the filters are applied and the result is converted back to L-R.
M-S allows to change the stereo image of a stereo signal by frequency. An example: Low cutting a signal on the side-channel only will make the bass frequencies totally mono.
In M-S mode apQualizr draws the stereo analyzer differently. The side channel is drawn by a line only to make it easier to see the overall mid or side curve. Additionally newly created filters default to filtering the mid channel only.



Mono This forces apQualizr to only process mono even if the signal is stereo. This mode exists because some older hosts host every plugin as a stereo plugin or don't have any option to make a track mono. In these cases stereo processing would be a waste of cpu.

7.7 Popup Menu

The apQualizr popup menu is opened by clicking the apQualizr title at the bottom left corner of the apQualizr interface.

7.7.1 About apulSoft apQualizr v#.##

Opens the apQualizr splash screen. This is the fastest way to determine the exact version of apQualizr.

7.7.2 Visit the apulSoft Website...

Opens the system standard browser and visits `http://www.apulsoft.ch`.

7.7.3 Check for updates online...

Opens the standard web browser and visits the apQualizr version checking site on apulsoft.ch. The site shows whether the very latest version of apQualizr is running.

7.7.4 Open the apQualizr manual...

Opens the apQualizr pdf manual in the system standard pdf viewing app.

7.7.5 Enter/Update ID/serial...

Reopens the ID/serial input box where the current info can be looked up or edited.

7.7.6 Save current preset as 'Init'

Saves the current preset as the 'Init' preset which will be loaded initially whenever a new instance of apQualizr is created.

7.7.7 Remove all filters

On option mainly for the vst version. As vst uses a presets banks scheme, changes made to the factory settings stay effective when switching presets. Sometimes it can be bothersome to get a clean slate again as the **Init** factory setting has likely been changed. This menu entry removes all created filters and sets master gain to +0dB.

8 Filter Types

8.1 Peaking EQ

A peaking EQ with bandwidth parameter. The bandwidth determines the point where filter gain is half of the peak gain. Using stack with the peaking EQ results in a more bell-like curve with faster decay outside the bandwidth area. If the filter frequency comes close to half the sample rate, using oversample is recommended in order to get reasonable filter results.

8.2 1P Lowpass/1P Highpass

Six decibel/octave lowpass/highpass filters. These types use less cpu than the 2P lowpass/highpass filters. They don't have any gain, Q or bandwidth parameter. Using stack multiplies the filter slope steepness.

8.3 2P Lowpass/2P Highpass

12 decibel/octave lowpass/highpass filters with gain. These can be used like classic synthesizer filters, high gain values lead to resonance. Using stack multiplies rolloff, but also flattens the resonance part.

8.4 Bandreject

This filter type is used to eliminate specific frequencies from the signal. The 50 Hz / 60 Hz hum removal factory presets demonstrate how to remove electrical hum with its overtones.

8.5 Bandpass

A bandpass filter with bandwidth control. The bandwidth parameter controls the position of gain value $-3dB * stackfactor$. A more flexible bandpass filter can be realized by using a 2P highpass in combination with a 2P lowpass filter.

8.6 Low Shelf/High Shelf

Shelfing filters which have a Q parameter as well as a gain parameter. The gain parameter controls the amount of shelfing, using Q leads to resonance effects on both sides of the filter frequency. Using stack results in a steeper filter.

9 CPU Usage tips

apQualizr can use a lot of CPU as one of its main design philosophies was 'never sacrifice any kind of quality for cpu'. CPU usage can be kept reasonable by following a few simple guidelines:

- Only use oversample if high frequency content needs to be processed. Oversample will not improve the sound of low frequency filtering.
- Turn off the FFT analyzer if it is not needed. It uses a lot of CPU for drawing as long as the plugin window is visible.
- Turn off clipping if it is not needed.
- Use low settings for **FFT Res**.

10 Frequently Asked Questions (FAQ)

- *I lost my apQualizr serial info. What now?*
  Please contact apulSoft via `http://www.apulsoft.ch/contact.php` and let us know who you are (email-address / name). ApulSoft will retrieve your info and send it via Email.

- *Will there be a 64 bit native version of apQualizr?*
  Most likely yes. apQualizr already uses 64 bit processing internally. Once the major hosts are available in native 64 bit versions, apQualizr will be updated.

- *Is apQualizr compatible with Logic Node?*
  apQualizr for now is not compatible with Logic Node. In Logic Pro 8, apQualizr should be excluded from Node usage in the AU manager. It is something that would be added if there were many requests.
  If you need this, please contact apulSoft via `http://www.apulsoft.ch/contact.php`

- *Why did apulSoft rename the plugin in version 1.4?*
  Unfortunately someone came up with the exact same 4 letter name around at the time the plugin was originally released. While apulSoft is a one-man operation, they have got lawyers and a trademark registration and there was no way to keep using the old name under those circumstances.

- *When installing the renamed 1.4+ plugin, will my old projects using older versions still load?*
  The newly named plugin will install alongside the old, not replace it. If you have projects using the plugin with its original name, do not delete the original-named plugin to ensure compatibility. For new projects it is highly recommended to use apulSoft apQualizr as this will be the one that is going to be updated in the future. apulSoft is very sorry it had to come to this.

11 Changelog

- Version 1.0.0
  - Initial release.
- Version 1.1.0
  - Added compatibilty with Windows XP and Vista
  - Improved filter curve updating
  - Added parameter switching during filter handle draggin via Alt
  - Added M-S processing mode
  - BUGFIX: Filters disappear if negative frequency values are entered.
  - BUGFIX: Horizontal dragging on numerical values is flipped.
  - BUGFIX: Force mono mode can't be turned off once editor gets closed.
  - BUGFIX: Plugin-Bypass handling is broken in the AU version.
- Version 1.1.1
  - BUGFIX: Inserting apQualizr in stereo channels in SONAR 6 on windows.
- Version 1.1.2
  - BUGFIX: Crash in filter overlay drawing (bad crash on windows).
- Version 1.2.0
  - Added changing filter type by clicking the filter name.
  - Improved Filter creation popup behaviour.
  - BUGFIX: Crash when using samplerates $>$48 kHz on windows.
- Version 1.3.0
  - Added saving a default init preset.
  - Added zoom to the analyzer and the filter display.
  - Doubled the FFT resolutions.
  - Improved handle appearance and behaviour for filters outside the viewed area.
  - Improved analyzer calculations to produce a more steady cpu load.
  - Improved the downsampling filter of the 4x oversampler.
  - Faster analyzer graph drawing.
  - BUGFIX: Notch filters drew wrong if they were outside the viewed area.
  - BUGFIX: The mousewheel did not work on windows.
  - BUGFIX: Preset names did not show up in Digital Performer.
- Version 1.3.1
  - apQualizr is now compatible with the fxpansion vst2rtas wrapper.
  - New way of authorizing apQualizr (workaround for problems with Logic 8).

- Version 1.3.2
  - BUGFIX: Crash after removing the plugin from the host while the GUI is open.
  - BUGFIX: Incompatibility with some OS X intel VST hosts.
  - The demo version now remembers settings to avoid issues with authorized/non-autorized machines.
- Version 1.4
  - Due to some legal bullying and interesting use of trademark law, the plugin had to be renamed from its former 4-letter name to apQualizr. Unfortunately this means the plugin lost backwards compatibility with older versions. Owners of the 4-letter version should keep a copy around in order to load old projects.
  - Changed default value for oversampling to 1x for most presets.