apulSoft apQualizr2 v2.6.0 Manual

modular equalizer audio plugin
(VST/VST3/AU/AAX)

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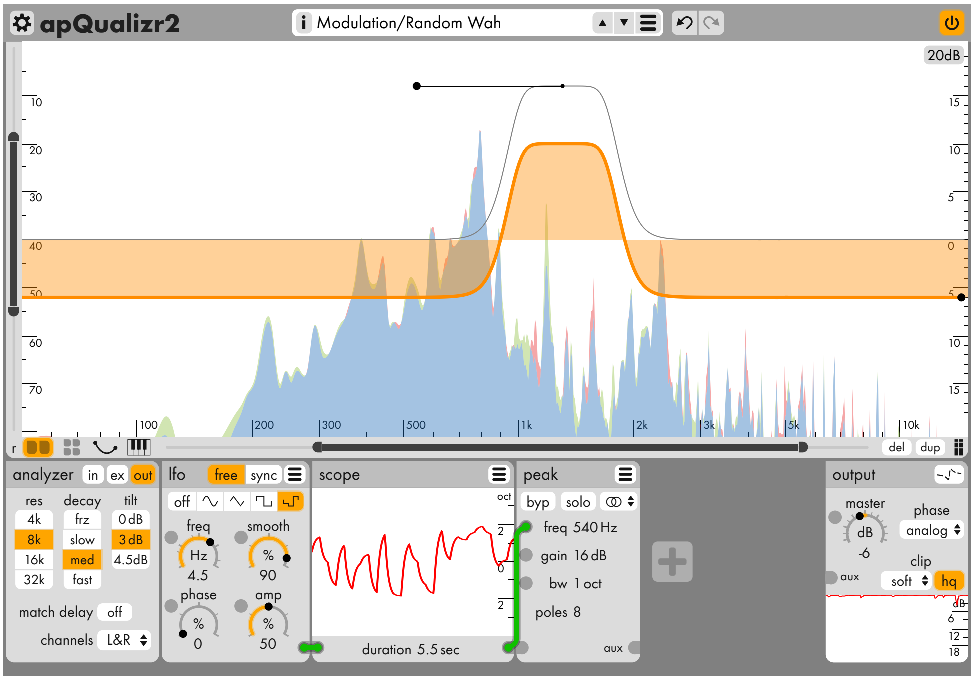
VST plugin technology by Steinberg.
AU plugin Technology by Apple.
AAX plugin Technology by Avid.
Manual written with LaTeX on September 8, 2023

Introduction



apulSoft apQualizr2 is an audio plugin based on a multiband equalizer layered on top of a frequency analyzer with elements of a modular effects system. It features graphical editing of top quality low latency filters and modular modulation of various filter parameters. It can be used for all standard eq applications and can produce many modulation and dynamics effects thanks to the dynamics and lfo modules included.

apQualizr2's filters are based on a unique method to match ideal frequency response curves over the entire frequency spectrum while maintaining minimal latency. Applicable filter types have adjustable steepness (adjustable number of poles) and can be individually switched to mid-side processing.

Contents

1 End User License Agreement

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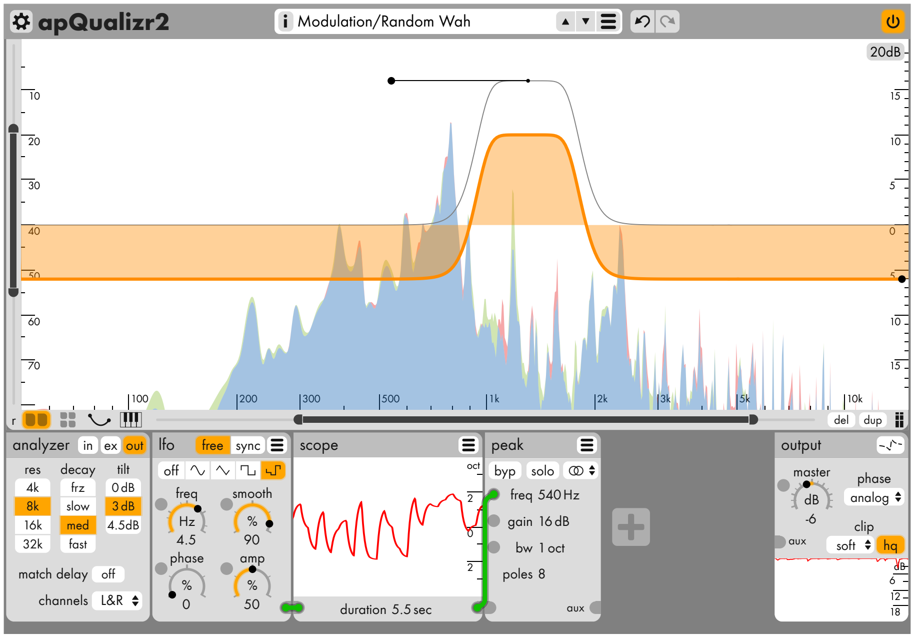
2 System Requirements

- macOS
  - macOS 10.11 or newer on an Intel or Apple Silicon CPU (64-bit only).
  - A host application compatible with VST, VST3, AU or AAX plugins running in 32-bit or 64-bit mode.
  - Pro Tools (AAX): Version 11 or newer. Pro Tools on Apple Silicon is natively supported.
- Windows
  - Windows 7 or newer. Both 32-bit and 64-bit versions of Windows are supported.
  - A CPU with SSE2 instruction support (Pentium 4 or newer).
  - A host application compatible with VST, VST3 or AAX plugins running in 32-bit or 64-bit mode.
  - Pro Tools (AAX): Version 10.3.5 or newer.
  - An application to view pdf files to read this manual.

3 Installation

- macOS
  - Quit all plugin host applications.
  - Double-click **apqualizr2-mac-(..).pkg**.
  - Follow the macOS installation procedure.
  - Open a host and create an instance of apQualizr2 in a plugin slot.
  - The apQualizr2 GUI will show a welcome screen with the options to run the plugin in demo mode or to buy or enter license information.
- Windows
  - Quit all plugin host applications.
  - Double-click the **apqualizr2-installer-win-(..).exe** to start the installation. On newer versions of windows it may be necessary to confirm the launch because of user access management.
  - Follow the installation procedure. During the install you have the option to set the path to the apQualizr2 data folder. That is where settings, presets and the manual will be installed.
  - If VST2 versions are installed, the installer will provide the option to select destination folders for VST2 plugins for both 32-bit and 64-bit.
  - Open a host and create an instance of apQualizr2 in a plugin slot.
  - The apQualizr2 GUI will show a welcome screen with the options to run the plugin in demo mode or to buy or enter license information.

4 Overview



The apQualizr2 user interface has three main parts. On top is a status bar with the settings button, plugin title, preset selection, undo/redo buttons and license information.

The large area below is where the analyzer graphs are shown and frequency response curves are shown and adjusted using handles.

At the bottom is the module tray. Between the two built-in analyzer and output modules, a module is shown for each created filter band and optionally more modules can be created by the user to modulate filter band settings. The modules can be moved using drag and drop. Module in- and outputs allow the user to create signal graphs as needed.

Most of the controls on the GUI can be dragged with the mouse to change values.

- If the Shift key is held down, draggable values snap to predefined markers at round values. On the frequency graph, holding Shift locks dragging to the frequency or gain axis.
- Holding Ctrl switches dragging to be scaled by 1/20 for fine adjustments.
- Alt -clicking and Ctrl -right-clicking make draggable values jump to their defined default values.
- Holding Alt while the mouse is above the main graph enables the band-pass listening mode where an additional variable bandwidth bandpass filter is applied to the output to quickly listen to specific frequencies only.
- Most of the value-based controls can be double-clicked to open a popup value editor to be used with the keyboard.

5 Top Bar

Click the **gear button** on the left or the plugin title to open the preferences/information dialog.

5.1 Preset Section/Undo/Redo

The white box shows the label of the current preset and can be clicked to edit the label. The "**i**" button on the left opens the preset description popup with the option to edit the text.

On the right side are up/down arrow buttons to **cycle** through presets. Depending on the global setting "cycle all presets", these buttons cycle through the folder containing the current preset or through the entire preset folder structure in the order displayed in the preset menu. The rightmost button opens the **preset menu**. It lists all available presets with folders as they are organized on disk. Below the presets the following entries are available:

save current preset.. The current state of the plugin is added to the preset menu. In the prompt that pops up, the preset name can be edited and folder paths can be added which automatically creates folders on disk if necessary. Presets in the menu can be overwritten by using the same name/path.

import preset.. Load a preset from a .apq2preset file anywhere on the local filesystem.

export current preset.. Store the current state of the plugin as a .apq2preset to any location on your local filesystem.

manage presets folder in finder/windows explorer.. This opens the filesystem folder that contains the presets shown in the menu. The usual file operations can be used to restructure this folder and therefore restructure the presets menu.

To the right of the white preset box are the **Undo** and the **Redo** button. These undo and redo that last actions when clicked. apQualizr2 supports unlimited undo and redo.

Note: Saving and importing of presets is only available in the full version of apQualizr2.

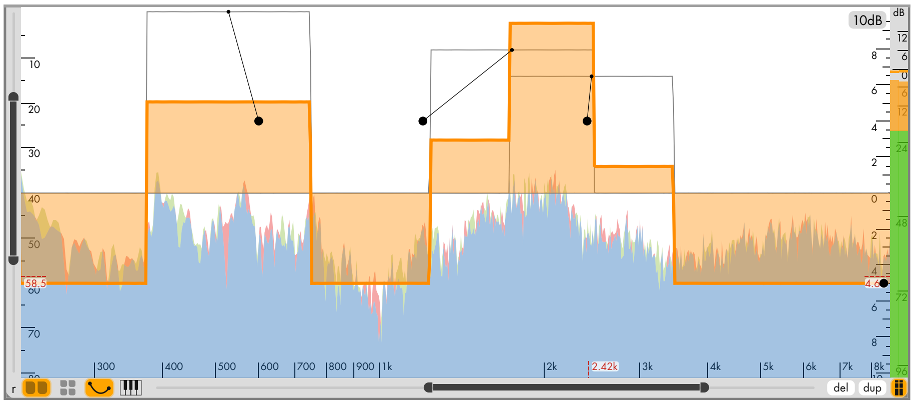
5.2 License

The look of the license section changes depending on your license status. In demo mode the →full version button is displayed. It brings up a dialog with options to buy a apQualizr2 license online, to enter the purchased license information or to keep running in demo mode.
Once full version of the plugin is unlocked, the license section displays the license ID.

5.3 Enable Processing Button

Toggle this button of to bypass all processing in apQualizr2. On compatible hosts, the button will hook up to the host bypass feature, in other hosts this button allows to automate bypassing the plugin without audible artefacts as the fade-in/out is handled by apQualizr2 if this button is used.

6 Frequency Analyzer/Frequency Response Curve Display



The main display of apQualizr2, where frequency graphs are shown, filters are created and edited by dragging on their handles.
On the left and bottom side of the graph there are range sliders which can be used to adjust the viewed frequency range as well as the displayed dB range of the analyzer graph. Additionally, the frequency range can be adjusted using the mouse wheel on the frequency ruler markings. On the right, an output level meter can optionally be displayed.

On the top right the +- dB range of the filter curve is shown and can be adjusted by a popup menu. If a filter handle or the selection rect is dragged near the top or bottom border, the filter range auto extends. mmoving towards the center line during a drag auto-contracts the range again. If filter handles are moved towards the left or right borders, the frequency range automatically scrolls.

Buttons on the bottom bar besides the frequency slider:

r Reset the graph viewport area to the values defined in your **Init** preset.

module single toggle By default, apQualizr2 displays one row of modules with horizontal scrolling at the bottom and this button is lit. Toggle the button off to use the entire plugin window for the frequency display.

module multi row toggle Switch to multi-row module mode with vertical scrolling. Multiple rows of modules are displayed for a better overview when creating complex presets. Toggle the button off to use the entire window for the frequency display.

cable dangle toggle Display connections as curved cables instead of routing between the modules.

piano ruler toggle Controls the visibility of the piano ruler display on the bottom of the graph.

del Delete the currently selected filters and/or modules. The same thing can be achieved by pressing Delete or Backspace on the keyboard.

dup button duplicates the currently selected filters and/or modules with identical settings. For filters this means two filter main handles lie on top of each other after this operation.

output meter toggle Display the output level meter on the very right side.

6.1 Filter Creation

Clicking on empty space on the graph brings up the filter creation popup (if nothing is selected). A list of the available filter types is shown. Clicking on one of the list entries creates a new filter of the selected type at the clicked frequency and gain location.
Right-clicking brings up the chooser as well, but regardless of selection state. It also immediately shows the menu, so Right-click + drag can be used to created a new typed filter band with just one mouse gesture.

Double-clicking on empty space on the graph creates a new filter band without showing the chooser. The filter type depends on the mouse location at the time the click happens. A symbol near the mouse cursor indicates the type of filter to be created on double-click. Below 100 Hz, high-cut and low-shelf filters are created, above 10kHz it's low-cut or high-shelf filters.
Using double-click-drag (+ mouse-wheel), the newly created band can be edited during the same mouse gesture.

Additionally, peak filter bands can be created by dragging the mouse on the filter sum curve. A dual-arc symbol indicates the readiness for filter creation.

6.2 Filter Types

Peak Boosts/attenuates the frequency band around a center frequency. It has an adjustable bandwidth in octaves that is defined by half the gain-value. Higher pole versions lead to a flat gain plateau around the center frequency.

Band Stop Cuts out a band of frequencies completely. Bandwidth defines the -3dB points. This type has no gain parameter.

Low Pass Cuts frequencies above a cutoff value and lets low frequencies through. The gain value adds resonance to the lowpass which boosts/attenuates frequencies around the cutoff frequency.

High Pass Cuts frequencies below a cutoff value and lets high frequencies through. The gain value adds resonance to the high pass which boosts/attenuates frequencies around the cutoff frequency.

Band Pass High and low frequencies outside the bandwidth area are removed from the signal and gain is applied. The bandwidth spans between the two -3dB points.

Low Shelf Low shelf filters boost or cut low frequencies by the amount set up by the gain parameter. There is a transition area which width depends on the number of poles used. The frequency parameter defines the middle of the transition region where half the gain is applied.

High Shelf High shelf filters boost or cut high frequencies by the amount set up by the gain parameter. There is a transition area which width depends on the number of poles used. The frequency parameter defines the middle of the transition region where half the gain is applied.

Tilt Shelf A Tilt shelf filter boosts or reduced the frequencies above the center frequency by a set amount while applying the opposite change below the center frequency. The more poles are used, the steeper the transition area gets.

Comb The comb filter produces many evenly spaced frequency spikes at the same time. The spikes are located at the odd harmonic frequencies of the main frequency. The frequency and gain parameter set up the lowest spike. As the spikes are evenly spaced frequency-wise they appear to get more and more dense towards higher frequencies as the frequency graph (and human hearing) work logarithmically. Technically the comb filter is implemented as a short delay with feedback.

Low Pass Variable Slope The **low pass vs** cuts frequencies above the cutoff value like the regular low pass and additionally allows to adjust the frequency/octave slope to any value, breaking the usual limitations of pole-based filters. As an added bonus it can do negative slopes, boosting the frequencies above the cutoff linearly.
The slope value parameter can be modulated by other modules and using negative values allow to create a unique variable-slope boost filter. The gain value adds resonance which boosts/attenuates frequencies around the cutoff frequency. **Note:** A slope of 6.0206 dB/octave is what traditional filter produce for each pole added.

High Pass Variable Slope The **high pass vs** cuts frequencies below the cutoff value like the regular low pass and additionally allows to adjust the frequency/octave slope to any value, breaking the usual limitations of pole-based filters.
The slope value parameter can be modulated by other modules and using negative values allow to create a unique variable-slope boost filter. The gain value adds resonance which boosts/attenuates frequencies around the cutoff frequency. The high pass variable slope filter includes a dc removal stage at 5Hz to avoid generating/amplifying DC offsets when using negative slopes. **Note:** A slope of 6.0206 dB/octave is what traditional filter produce for each pole added.

Spectral Tilt The spectral tilt filter allows to precisely tilt the entire frequency spectrum (down to 10Hz) by a set amount of dB per decade. It includes a dc removal stage to avoid generating/amplifying DC offsets when using negative slopes. **Examples:** A -10dB/decade spectral tilt filter turns white noise into purple noise (also called a pinking filter). A -20dB/decade spectral tilt filter can turn a square wave into a triangular wave (also called an integrator).

Harmonic Peaks A bank of up to 16 two-pole peak filters with configurable spacing. This filter enables sculpting the overtones of a musical note with just one module. Check the Harmonic Filter Band module description for information about all the parameters.

Harmonic Bands A bank of up to 16 two-pole bandpass filters with configurable spacing. The sub-bands are processed in parallel, it is like up to 16 bandpass filters mixed together. Outside the peaks area, the falloff is 6dB/oct. This filter can for instants produce a musical sound from noise by letting just an overtone structure through. Check the Harmonic Filter Band module description for information about all the parameters.

Harmonic Notches A bank of up to 16 two-pole bandstop filters with configurable spacing. This filter can be used to filter away a complete overtone series from a signal. It's great to get rid of hum, but can also be used to cut away specific notes. Check the Harmonic Filter Band module description for information about all the parameters.

Gain A filter band that just changes gain over the entire frequency spectrum. Useful for dynamics processing and to completely mute the main chain (at -∞ gain) for parallel processing using the aux connectors.

6.3 Editing Filters

Filter frequency and gain can be changed by dragging the main filter handle on the graph. Clicking or dragging a handle will automatically select the filter band.

Selected filters show additional handles depending on the capabilities of the filter type. Most filters support changing their steepness/polescount. In that case a rotary handle is displayed around the main filter handle which can be used to change the pole count by dragging up/down or left/right.

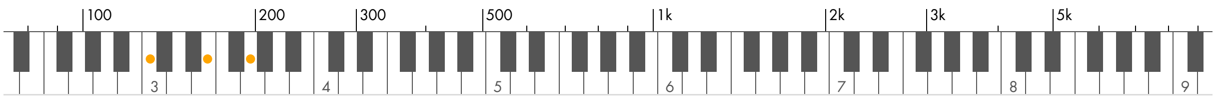
Since version 2.2, the poles count can be increased past the previous maximum of 16. In that case filter bands enter the **++** mode where filters become extremely steep. The cost of this is 0.01 dB ripple over the entire frequency range per **++** band and such extreme filters tend to ring quite a bit. Additional peak/band stop bands can of course be used to fight this ringing.

Filter types with a bandwidth parameter show a green area that show the set bandwidth. Dragging the two thin lines on the left and the right border with the mouse changes the bandwidth of the filter.

6.4 Keyboard and Mouse Shortcuts During Filter Editing

Mouse drag on empty area	Rubber-band selection of multiple filter bands.
Shift -Click	Select multiple filters to edit them together.
Ctrl -A	Select all filter bands at once.
Ctrl -Click	Bypass/Reenable the selected bands.
Double-Click	Bypass/Reenable the selected bands.
Alt	Holding Alt switches to band-pass listening mode which applies a variable bandwidth band pass filter to the plugin output. Use Alt -mouse wheel on empty space to quickly change the bandwidth.
Right-Drag	Dragging a filter handle with the right mouse button switches to bandwidth adjusting mode.
Ctrl -Drag	Switch to velocity-based value dragging. Slow movements allow for tiny adjustments.
Shift -Drag	Lock dragging to the frequency or gain axis depending on which one has the larger change at the time of pressing Shift
B	Holding B while dragging a filter handle switches to bandwidth adjusting mode.
Mouse Wheel	Turning the mouse wheel while the mouse pointer hovers over a filter handle adjusts the bandwidth for filter types with bandwidth property and the number of poles for other types. While holding Alt the mouse wheel adjusts the bandwidth of the frequency band solo filter.

6.5 Piano Ruler



If the the piano ruler toggle button left of the bottom scroll bar of the graph view is enabled, a piano ruler is displayed at the bottom of the display. Using the preferences setting about A4 tuning, the position of each musical note is calculated so the keys match the frequency position of their fundamental note. The piano ruler displays an orange handle for each filter center frequency. If these handles are dragged, the frequencies are quantized to musical pitches. The quantization is chromatic when dragging over the upper half and c-major diatonic on the lower half. Double-clicking a note allows to create a 6 dB peak filter for that note.

7 Module Tray

The module tray is shown as the bottom section of the user interface if the module tray toggle button at the bottom left of the graph display is on.

The module tray always starts with the analyzer module at the left and ends with the output module on the right. Between these, a module is displayed for each filter band created. Additional modulation modules live in the tray as well. The order can be freely changed by dragging and dropping modules. Processing always happens top->down (in multi-row display mode) and left->right for each row.
New modules can be added by clicking on the large plus button. A popup menu with the available module types is displayed to choose what new module to create. Modules can also be created by right clicking unconnected module inputs and choosing the module to create in the input context menu.

7.1 Module Interface

Each module has a title bar with its name and a module menu (for user-created modules). Double-click the names of user-created modules to edit them. If the name of a filter module is changed from its default, it is displayed near the filter handle on the big graph view.

The module menu allows to **duplicate** and **delete** modules. Some modules feature additional entries in the plugin menu described in the module types section below. In multi-row module viewing mode, **start new row** moves the selected module and any following modules to the next row.

Below the title bar is the individual interface of each module. Dragging on the space between controls allows to reorder modules.
Custom module inputs are displayed on the left side of modules and custom outputs on the right side as darker rounded areas. Some modules have knobs with additional modulation inputs on their top left corner. These look like dark circles. Dragging the mouse on an empty input or output creates a new cable. When dragging on an input/output with an existing cable, the connection gets lifted and the cable can be attached to another connector. Hold Shift to override this and instead create a new

cable from the same connector.
Right-clicking an unconnected input/output brings out options to directly create and connect new modules.

7.2 Cables

Create modulation connections by clicking and dragging on module inputs and outputs.
Multiple cables can start and end at the same connector. When dragging on an input/output with an existing cable, the connection gets lifted and the cable can be attached to another connector. Hold Shift to override this and instead create a new cable from the same connector.
Right-clicking a cable brings up a cable popup menu where the cable color can be changed, the cable can be replaced by a module or the cable can be deleted.
Any output can be connected to any input as long as there is no feedback loop.

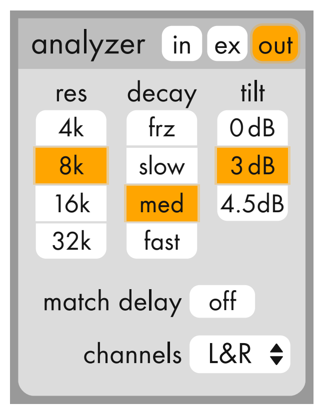
7.3 Cable Dangle

The regular cable display algorithm routes cables around modules to keep controls visible. This can lead to many cables being displayed on top of each other. The cable dangle button (right of to the module tray view button) switches to displaying cables as (catenary) curves. This might draw over some controls, but gives a much better overview for complex presets.

8 Module Types

8.1 Analyzer

The analyzer module is always the first module in the module tray and controls various analyzer-related settings.



in enables the input analyzer right at the plugin input before any processing is applied by apQualizr2.

ex enables the external signal (sidechain) analyzer that displays the signal connected to the sidechain input. How to set up a sidechain connection depends on your host.

out enables the output analyzer which is places just before the plugin output after all the processing has happened including output gain and output clipping.

res This setting defines the number of samples used as FFT block size. This is directly proportional to the diplayed frequency resolution. The higher the resolution, the more detailed the frequency graph gets. However at the same time the larger blocks mean the display reacts slower to changes and the transients get smoothed more. Higher resolutions also require more processing power.

decay The speed at which the frequency graph moves downward. **Frz** means all freqs stay at their maxima.

tilt This setting optionally applies a tilt to the graphical output. At the **0 dB**/Oct setting the display is mathematically accurate and white noise gets displayed as a horizontal line. Music tends to fall off towards higher frequencies and the **3 dB** and **4.5 dB** settings allow to compensate for that. Natural sound sources then appear more like a horizontally balanced graph.
Note: Tilt is only applied to the analyzer graphs, not the filter frequency curves.

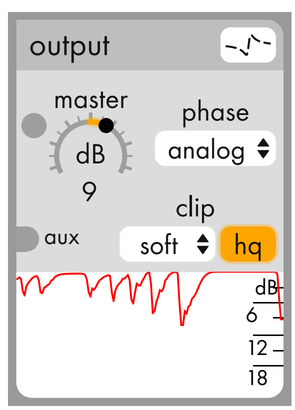
match delay This toggle switch enables an internal delay which is set to half the fft block size. At the same time this amount of delay time is reported to the host application as plugin latency. The intention is to better align the visual fft graph with the audible sound. To make it work, the host application needs to support plugin latency compensation.
Notes: Many hosts do not support changing latencies while the audio is running. In this case the host transport needs to be restarted for things to work. This setting is definitely not recommendad for live usage!

channels This popup menu is only available if apQualizr2 is used on a stereo channel and shows the various stereo view modes.

- **left & right** Two analyzer curves are displayed for each analyzer. A green tint is used for the left channel, a red tint for the right channel.
- **left/right avg** For each frequency value, the left and right channels gains are added and multiplied by 0.5.
- **left/right max** For each frequency value, the maximum gain or the left and right channels is used.
- **left only** Only the left channels curve is displayed.
- **right only** Only the right channel curve is displayed.
- **mid & side** The input signal is converted to a mid and a side signal. The side signal is displayed as a black line, while the mid signal is shown as a yellow area.
- **mid only** Show the mid channel only. This is the same as Left-Right Mix.
- **side only** Show the side channel only.

8.2 Output

The output module is the last module in the tray and is always present.



The **master** gain is applied before clipping and has a modulation input.

The **phase** popup button allows choosing from multiple phase correction engines: minimal, analog, mixed and linear. They are described in detail in **section 9 Phase Engines**.

On the top right, the **phase graph** toggle button enables displaying the main chain phase response as a dashed line. The phase display has a range of -180° to 180°- if the phase change gets larger, it is wrapped around.
Note: Displaying the phase curve is a very heavy operation and might cause the gui to refresh slower. It is not recommended to leave the toggle on.

The **aux** input allows mixing in a signal before the master gain is applied. This can be parallel filtered signals or the sidechain.
Note: Phase correction is not applied to anything connected to the aux input.

On the **clip** popup menu, four clipper modes can be selected:

off No clipping is applied. **Note:** If extreme filter settings are used, very loud signals can occur!

hard Any value above 1.0 (0dB) is hard clipped at 1.0 (0dB). This produces harsh distortion for large values.

soft Soft clipping makes sure no sample goes over 1.0, but values are gradually softened over the top 12dB.

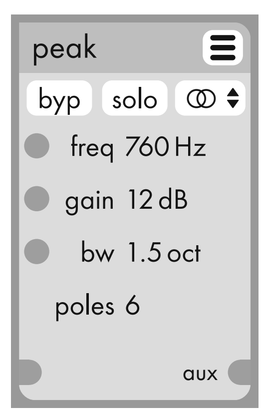
asym Asymmetrical soft clipping lowers levels for the top 9dB of the positive signal side and for the top 15dB of the negative signal side.

The **hq** button enables aliasing suppression for the clipper. The signal is oversampled and the clipping is applied to a continuous model of the audio stream to suppress unwanted mirrored frequencies. Due to Gibbs' phenomenon from the downsampling filters, abrupt jumps in the signal can still exceed the -1.0/1.0 range after clipping. Antialiasing leads to additional processing latency. In minimal phase mode, the extra latency is 6 samples, while in the other phase correcting modes linear-phase oversampling adds 125 samples of latency.

The lower part of the output module is a graphical display that shows how much clipping has been applied over time. For stereo instances, it shows the maximum clipping applied.

8.3 Filter Band

For each filter band created on the main graph view, a filter band module is automatically created. It allows to edit filter parameters and add modulation cables to them. Depending on filter type, different parameters will show up. A description of all filter types can be found in **subsection 6.2 Filter Types**.



The filter module menu on the top right has an extra option to change filter type while maintaining filter parameters.

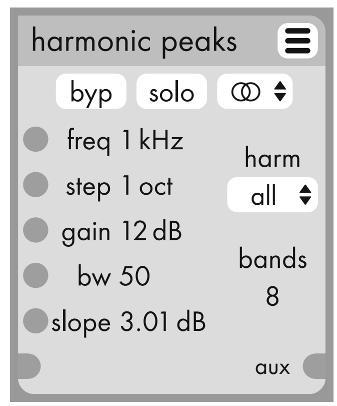
The **bypass** button disables processing of the filter band and just feeds the

input to the output.
The **solo** button acts like bypass was enabled on all bands which don't have
solo enabled.
In the **stereo** configuration menu, processing can be limited to the left, right, mid or side channels.

At the bottom, the filter modules have an **aux** signal input and output. If cables are connected to these, the filter is removed from normal processing and directly processes the connected signal (or the plugin input if the aux input is not connected) and feeds the result to the aux output. This enables pre-filtering input of detection modules (like dynamics) and parallel processing using the aux in of the output module.

8.4 Harmonic Filter Band

This extended version of the filter band module is created for each harmonic filter created on the main view. Frequency and gain controls work the same as in the regular module. The stereo setting differs from the regular filters. It is set using a popup button on the top-right corner.



The **step** parameter controls the harmonic series. It is the base value that gets multiplied to determine the frequencies of all the bands using the series selected in the **harm** menu:

all 1x, 2x, 3x, 4x, 5x, ... all harmonics

odd 1x, 3x, 5x, 7x, 9x, ... odd harmonics

even 1x, 2x, 4x, 6x, 8x, ... fundamental + even harmonics

cnst 1x, 2x, 4x, 8x, 16x, ... constant distance in octaves (logarithmic)

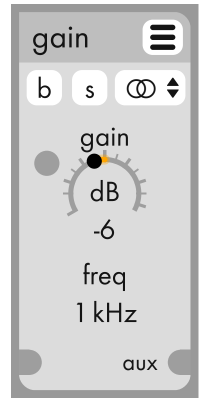
The **bw** parameter controls the relative bandwidth of all bands. At 100 the half-gain/-3dB point is at the center frequency of the next band, 1 is very narrow and produces long ringing for all harmonic filter types.

The **slope** parameter controls the gain of all upper bands. It is a value in dB/octave that gets applied to each subband based on its frequency. The slope is displayed as a straight line starting at the main filter handle.

The draggable **bands** control sets the number of subbands to create. CPU usage of the harmonic filter depends on the number of bands and can get quite high at the maximum number (16).

8.5 Gain Filter Band

This simplified version of the filter band module is created for each gain filter created.

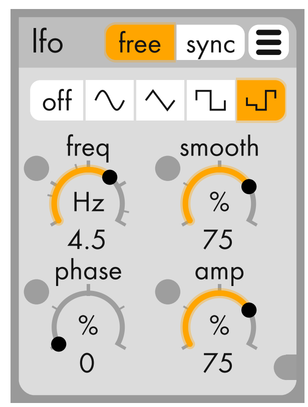


The gain filters just make the signal louder or softer. The textbffreq parameter only exists for the visual positioning of the filter handle.

Gain filters can be used to completely mute the main chain with gain set to the -∞ minimum.

8.6 LFO

The LFO (Low frequency oscillator) module generates various repeating signals. In **free** mode, the LFO generates its own tempo based on a Hertz value and in **sync** mode the LFO can sync to the host beat tempo. In that case, the frequency knob adjusts the ratio to the host beat.
Note: not all hosts supply beat information.



The LFO module supports these waveform types:

off The module is turned off and outputs a stream of 0.

sine A sine curve.

triangle A triangle wave.

square A square wave switching between amp and -amp (defined by the **amp** knob).

random hold At the beginning of each oscillator cycle, a random value is generated and then held for the cycle duration.

The four main controls of the LFO module are controlled by knobs and all can be modulated by connecting a cable to the modulation inputs next to the knobs.
The **freq** knob adjusts the LFO frequency or its ratio to the host beat tempo.
The **smooth** button defines the amount of smoothing applied to the LFO output. The smoothing is always relative to the oscillator frequency so the waveform shape does not change if the frequency changes.

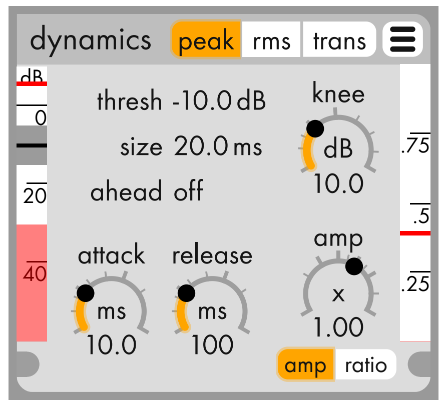
The **phase** setting offsets the waveform by a % value. A 50% phase setting means the waveform starts at its middle. Multiple LFOs with the same settings usually run in sync and using phase fixed ratios can be obtained. **Note:** If the frequency knob is adjusted, LFOs might loose sync and only restarting host playback will make then sync again.

The **amp** knob sets the LFOs amplitude.

On the bottom right is the one output of the module to be connected to modulation targets with a cable.

8.7 Dynamics

The dynamics module converts an audio signal coming from the cable connected to the input on the left into a dynamics signal that can be used to modulate parameters on other modules. It can be a envelope follower, a peak hold detector, a transient detector or a complete compressor control circuit.



There are two meters on this module. The one on the left shows the dB level of the incoming signal. Layered onto that is the threshold value showing the knee area as a gray bar. The threshold can be adjusted by dragging it with the mouse. The right one shows the output signal. Depending on where the output cable is connected to, different value units are displayed.

On the module title bar one of three main modes can be selected. **Peak** is the peak hold mode to capture fast transients in the audio. It means the maximum sample values are held for the time set up by the **size** parameter. **rms** stands for "root mean square". This means for the duration of the **size** parameter, squared samples values are summed and the sum is divided by the size. RMS mode is good for slower material and is a way to measure the energy contained in the signal. **Trans** mode uses a special algorithm to detect transients in the audio. Transients are steep rises of energy in the signal. In this mode the **size** parameter defines the scale of transients to capture. The transient mode only captures transients above the threshold.

The dynamics module has one input and one output. If nothing is connected to the input the dynamics module connects by itself to the plugin input. Once something else is connected to the input (such as a filter band aux output), the connection to input is removed.

In the center of the dynamics module, multiple parameters can be adjusted:

thresh The threshold value in dB. This is also shown as a line on top of the input level meter. It determines the level above which dynamics processing starts.

size The size parameter works differently depending on the dynamics module mode (set on the module title bar). In **peak** mode, it is the duration, for which maximum samples are held. In **rms** and

trans modes it is the size of the processing window, the duration of audio that is considered for each output value.

ahead Stands for Lookahead. This is the time the dynamics module "looks into the future" during the processing so modulation can occur before transients have already happened. If lookahead is set above zero, the dynamics module will introduce latency in the signal chain. Hosts with plugin delay compensation can make up for that by sending audio earlier, but not all hosts support changing latencies while transport is running. To make sure, things are running in sync, restart the host transport.

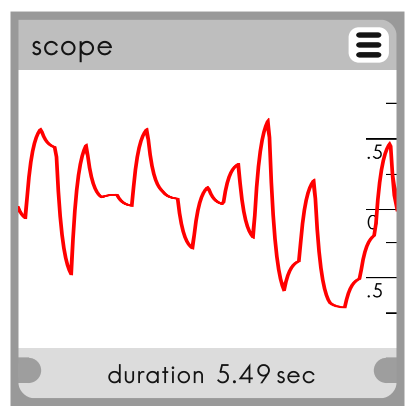
attack/release The attack/release times of the built in envelope follower. These set the time the output signal takes to rise/fall by 10 dB and can be used to smoothen the output signal.

knee The soft-knee control defines the area over which the dynamics processing gradually sets in in dB. 10 dB means dynamics processing begins 5 dB below the threshold and the full ratio/amp setting is reached 5 dB over the threshold. This value is shown as a grey area on top of the threshold meter/slider combo view.

amp/ratio The dynamics module has two ways of multiplying the output signal which can be switched below this knob. In **amp** mode, the value above the threshold is taken and multiplied by the amplification value. In **ratio** mode, the output signal is multiplied in a different way to become a gain reduction signal that can be used to drive the gain of filters or the master output to get compressor-like behaviour.

8.8 Signal Scope

The signal scope module can be used to visualize signals. It automatically picks up the signal unit if available and its view duration can be adjusted at the bottom.

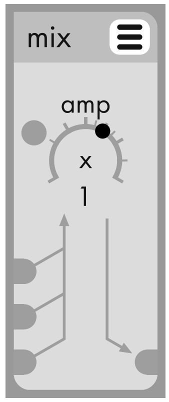


The module just feeds the signal through and can therefore be placed between a modulation source and target.

To visualize the signal going through a cable, right-click the cable and choose "Insert Signal Scope". If a scope module with cables connected on both sides is deleted, a shortcut cable is created.

8.9 Mix

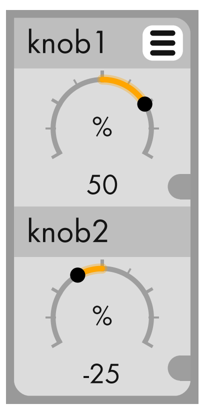
The mix module allows to mix up to three signals together and multiply the result with an amplication factor (**amp**) which can itself be modulated by a signal.
The multiplied sum is fed to the output on the bottom right corner.



Note: newer versions of apQualizr2 allow to connect multiple cables to one input and have a dedicated gain filter type. In most cases, this replaces the split module.

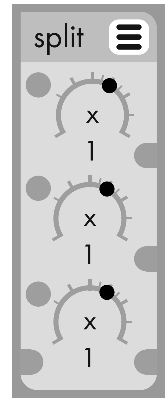
8.11 Knobs

The knobs module provides two large knobs that can be used as a source for modulation. Both knobs can be renamed and output a smoothed stream of the knob value. The knobs module exposes the knob values as automatable parameters to the host - one knob parameter can be used to automate multiple module parameters at once.



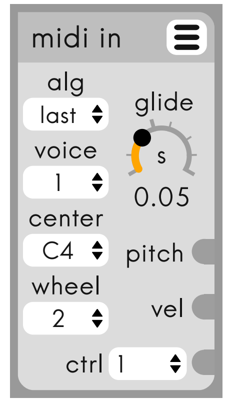
8.10 Split

The split module outputs the input signal to three separate outputs. Each output can be amplified by an individual amp factor.
These factors each have modulation inputs.



Note: newer versions of apQualizr2 allow to connect multiple cables to one output and have a dedicated gain filter type. In most cases, this replaces the split module.

8.12 Midi In



Use the midi in module to modulate parameters by midi. Check your host application's manual to figure out how to route a midi stream to an audio insert plugin, in this case to apQualizr2. Some hosts do not support sending midi to audio effects at all, some require to patch a virtual cable, some just send midi to all effects on the same channelstrip and allow to assign a midi input to that.

The midi in module supports polyphony using multiple instances of the module set to different voice numbers. The module auto-detects how many midi in modules are created and checks the voice settings. If multiple voice settings are found, it uses various algorithms to determine which pitch to output. **Last** is the smartest algorithm that prioritizes the last pressed midi note while keeping held note assigned to their voices as well as adding priority to close voices as well as the lowest and highest notes currently held. **Low** and **high** just sort the notes by pitch value and ignore holding keys.

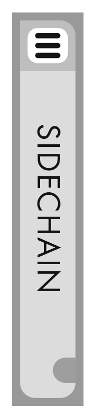
The pitch **center** settings defines the midi pitch for which a zero output is produced. If the pitch output modulates a filter band that is set to the same pitch, the center frequency will correspond to the midi notes fed into the plugin. Click the center value to enter any midi note name or midi note number.

The **wheel** setting sets the number of semitones the pitch bend wheel maximally affects pitch.

The **glide** algorithm uses linear interpolation over time between held notes. It releases whenever no note is output for the selected voice. It does work with multiple instances and polyphony, but the behaviour might be unexpected because voice switches can happen.

The **vel** output (velocity) uses logarithmical scaling for the midi velocity values (an industry pseudo standard). Velocity values 0...127 are mapped to 0dB ... 40dB gain if the output is connected to a gain input. To scale this to a larger or smaller range, use a mix module.
The midi **controller** output maps values 0..127 to the unit modulation range in apQualizr2. Controller numbers not preset in the popup can be entered numerically after clicking on the number.

8.13 Sidechain



Using the sidechain module, the sidechain input of apQualizr2 can be routed to other modules. To use it, a sidechain connection needs to be properly set up in your host application. Consult the host manual to find out how to do that. Sidechains are also called auxiliary inputs or external inputs for plug-ins.

apQualizr2 can handle mono and stereo sidechain input. Cables connected to the sidechain module output carry mono or stereo signal depending on the setup. If the sidechain configuration is not done properly, the module outputs silence only. It is possible to visualize the current sidechain signal using the **ex** button on the analyzer module. In order to route the sidechain to the output of apQualizr2, connect the sidechain to aux in of the output module.

9 Phase Engines

All filters in apQualizr2 are mathematically modeled in analog space with the frequency linearly going 0 to infinity. The analog models are transformed to digital filters using apulSoft's special method which maps one analog IIR biquad to two digital IIR biquad sections to produce a filter matching the frequency response of the original model closely over the entire frequency range. The resulting filters are minimal phase/have minimal latency.

Starting from v2.5, apQualizr2 includes multiple phase correction engines. These trade additional latency for specific phase targets. The mode is selected using the **phase** popup button on the (rightmost) output module. The phase change applied can be viewed using the **phase graph button** located on the top right of the output module. **Note:** Phase correction is only applied for the main chain filters, not for filters in aux mode which are calculated in parallel.

mode	latency	pre-ringing	algorithm	use-case
minimal	0 ms	none	IIR	universal, limited CPU budget, fast modulations
analog	$\sim$5 ms	minimal at very high frequencies	IIR/FIR	phase undistorted by frequency modulation, ideal "analog" sound
mixed	$\sim$22 ms	up to 15ms for resonant filters	IIR/FIR	mixing channels without phase-issues, preserving transient shapes, perfect frequency response.
linear	$\sim$178 ms	up to 150 ms	FIR	linear phase required, no modulation used

Note: If mid/side filter bands are used at the same time as left/right bands, the latency and cpu usage of the FIR phase correction engine is doubled!

9.1 Minimal Phase

The original apQualizr2 mode using digital IIR filters with minimal latency. This is the most cpu friendly mode with no additional latency. The filters sound close to the analog models, but the phase gently curves towards 0 near half the samplerate. Use this mode unless there is a good reason to use any of the other mode.

9.2 Analog Phase

By definition, digital filters need to have phase zero at half the samplerate. The analog phase mode pushes the adjustments reach phase 0 to the very top of the frequency response. In analog mode, 100% of the frequency response and 99% of the phase response is identical to the analog models. Therefore the relative phase response of filter bands remains constant if the main frequency changes - just like real-world analog gear. The cost for this is some latency and cpu usage to generate an additional FIR filter to adjust the top-end phase.
Note: the analog phase mode does not correct filter bands in ++ poles mode and comb filter bands.

9.3 Mixed Phase

The mixed phase mode makes gentle filters linear-phase while resonant filters work the same way as in analog phase mode. The frequency response is perfectly matched to the analog models. It causes 22 ms of latency, but provides most benefits of linear-phase filtering while still being able to do all kinds of frequency response modulations. The pre-ringing introduced is below the time resolution of human hearing and having linear phase on gentle filters also means transients preserve most of their shape (minimal/analog phase tend to slightly time-smear the highest frequencies). If multiple microphones were used for the same acoustic source, the mixed phase mode can prevent phaser-like effects when mixing filtered channels.
Note: the mixed phase mode does not correct comb filter bands.

9.4 Linear Phase

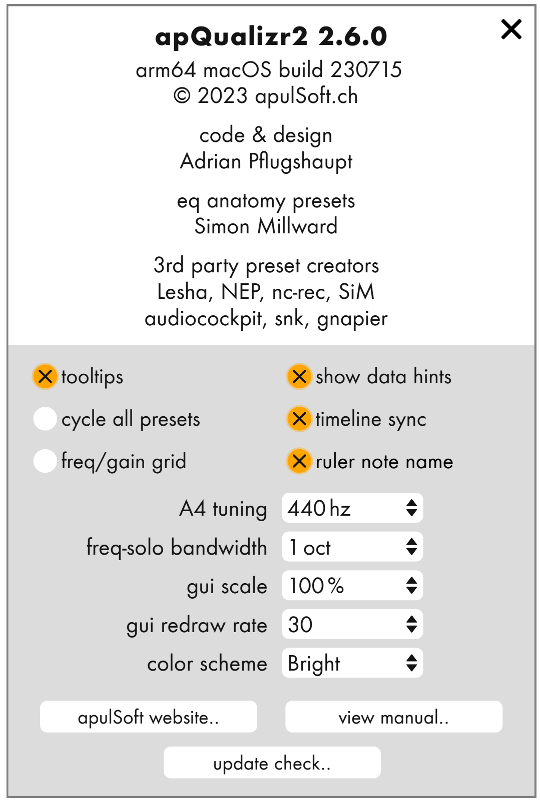
In this mode, all filters are converted to linear-phase FIR filters to ensure linear phase. To get reasonable bass response, FIR filter sizes in the region of 1/3 seconds are used, therefore linear phase mode leads to 1/6 seconds of latency. With resonant filters (steep slopes), pre-ringing can get extensive in linear phase mode. Resonances can start 1/6 second before the transients that cause them!

As generating the long FIR filters required is CPU-intensive, the filter reponse is updated only a few times per second (if any change happens due to modulation). Instead of the smooth frequency response transitions of the other modes the filter response crossfades to the new settings. That's why fast modulation does not work well with linear phase mode and gets sort of fade-quantized over time.

Linear phase mode is the only mode with an overall limited impulse reponse. This sets a limit to the maximum resonance. No filter can ring longer than 1/3 seconds.
Mixed phase mode is a better choice for almost all cases unless a requirement is to have no phase change at all or if pre-ringing is a desired effect.

10 Plugin Settings & Information Page

The top section of the settings dialog shows some basic information about the plugin.



In the middle there are some global settings. These apply to all instances of apQualizr2. All instances using the same plugin format in the same host will update immediately, others update once the plugin is reloaded.

tooltips If this is activated, orange rectangles with little hint texts pop up if the mouse hovers in place over a control for one second.

show data hints Display hint texts on the gui on how to create content if there is none.

cycle all presets When switching presets using the preset cycle buttons, cycle through the entire preset structure including folders instead of staying in the current folder by jumping between the first and last preset.

timeline sync Sync LFO position to the global host beat position. This makes LFOs recalc their position every time the host time jumps. Turning this off makes the LFOs use the project bpm value only - which might be necessary if a host does not send correct position information.

freq/gain grid If this is toggled on, frequency and gain grid lines are drawn on the main apQualizr2 frequency/gain view.

ruler note name Display the note name with an offset in cents on top of the main display frequency ruler for the current mouse x position.

A4 tuning If you use the piano ruler or enter note names to get specific frequencies, this setting is used to convert between (midi-based) note names and frequency values. The right side popup button shows some commonly used values or any number can be entered using the keyboard.

freq-solo bandwidth Holding Alt or S while the mouse is above the main graphical display temporarily adds a bandpass filter to listen to a frequency band only while everything else keeps working normally. This setting defines the bandwidth used for the filter and can also be adjusted using the mouse wheel while holding the key.

gui scale Choose how large the plugin GUI should be drawn in %. The right-side popup features a few presets and it is also possible to just enter any value between 25 and 500. Some hosts might only display the plugin correctly at the new size once the plugin window is closed and reopened. In extreme cases, the host might need to be restarted.

gui redraw rate The number of interface redraws per second. A slow computer might not be able to reach high rates. High refresh rates will only work well with small host audio buffer sizes.

color scheme Switch between multiple color schemes for the interface. The menu will show all installed schemes and new ones can be added by the user by renaming and editing the existing scheme files. These are located in a folder called ColorSchemes next to the apQualizr2 presets folder. Use the **manage in finder/explorer..** entry of the presets menu to navigate to the presets folder. The color schemes use an xml based format that can be edited in any text editor. More information can be found inside the **Bright.xml** file.

At the bottom of the preferences dialog there are four buttons:

apulSoft website.. Opens http://www.apulSoft.ch in the default browser.

view manual.. Opens the apQualizr2 manual in a pdf viewer application.

update check.. Opens a special page on the apulSoft homepage and sends version information. The homepage checks the version against the latest release and provides links to downloads if newer versions are available.

11 Init and Factory Presets

apQualizr2 installs a number of example presets by default. All factory presets include descriptions that are accessible by clicking the **i** button on the left side of the current preset name.

11.1 Init Preset

A special preset is the **Init** preset. The first time the plugin is opened it is auto-generated from the plugin's default values. Every time a new instance of apQualizr2 is created, the **Init** preset is loaded. This allows you to set up your personal default values by overwriting this preset once the plugin is in the desired default state. The "reset viewport" button in the bottom left corner of the main graph sets the visible ranges to the values defined in the **Init** preset.

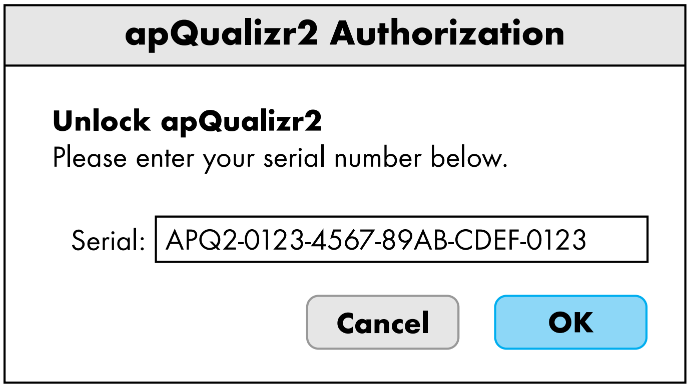
11.2 EQ Anatomy Presets by Simon Millward

The EQ Anatomy presets indicate the key frequency bands where popular musical instruments may be boosted or cut. The effect of each band's boost or cut is indicated by a label. These presets may be used as a guide to create your own filtering curves on top, or each band in the curve may be individually switched on or off using the Bypass switch for each filter (use Ctrl-Click/Double-Click on the filter handles in the display).

The EQ Anatomy curves were created using typical instrument sounds in each category but may need tweaking to achieve the desired effect with your own sounds.

All EQ Anatomy presets created by:
Simon Millward
(as published in 'Fast Guide to Cubase 6')

12 Unlocking the Full Version of apQualizr2



Once you bought an apQualizr2 license via 2Checkout from the apulSoft homepage (which can be opened from the demo welcome screen or the global settings dialog) there are two ways to enter your information and unlock the plugin.

- When you open first apQualizr2 plugin interface, a demo welcome screen appears with a **enter serial..** button. Click this button open the serial entry dialog.
- If the plugin is running in demo mode, the →full version button is displayed on the top right which brings up a license dialog where the **enter serial..** button can be used to open the serial entry dialog.

Just enter the serial exactly as received and click **OK**. to unlock the full version.

In case the serial is not accepted, check the following things:

- The serial needs to be an apulSoft apQualizr2 serial consisting of **APQ2** followed by 5 sections of 4 hexadecimal digits (0-9, A-F).
- If copy/paste was used, try typing manually as copy/paste sometimes copies more than was intended (white spaces, tab stops, etc).

12.1 Mid-2023 Serial Scheme Switch

If you bought apQualizr2 before mid-2023, you received an id (your email-address) and a serial number to unlock the plugin. With the new scheme, the id is no longer required, but the already received serial alone unlocks apQualizr2. The transition happens automatically on the first launch of a newer version, and thereafter, the ID is no longer displayed on the interface.

If you run into any trouble during the transition, please contact apulsoft:
https://www.apulsoft.ch/contact.

13 Frequently Asked Questions

- **I lost my serial. How can I retrieve it?**
  Just head to https://www.apulsoft.ch/contact and get in touch. Please add enough information to locate your order in the database and you will receive your serial as soon as possible.

- **What to do if the window size does not match the GUI size after adjusting the GUI scale?**
  Depending on how the host application handles resizing of plugins triggered by the plugin, changing the GUI scale might not immediately work correctly. Any change to the GUI scale is stored in a global preferences file that is read whenever a new instance of apQualizr2 is created. If this problem occurs, first try to just close and reopen the plugin window/editor. If that does not help, set the desired scale on the settings pane and then restart your host application. As long as GUI scale is not changed again, window and content should match.

14 Changelog

- Version 2.0.9
  - Initial public release.
- Version 2.1.0
  - Undo/Redo support.
  - Knobs module added.
  - Improved Installers to work around issues with AAX code signing.
  - Filter and FFT display frequencies always extend to at least 55
  - BUGFIX: Sometimes filters accidentally faded in during the first 250 ms of playback.
  - BUGFIX: Various GUI pixel tweaks.
  - BUGFIX: support for switching sample rates not working in some hosts.
- Version 2.1.1
  - Switchable color schemes. Switchable in preferences.
  - Optional Frequency/Gain grid. Switchable in preferences.
  - BUGFIX: Horizontal mousewheel events did no longer scroll frequencies.
  - BUGFIX: The master gain handle on the main display did not show modulation correctly.
  - BUGFIX: The Mix module amp factor only worked for integer factors.
  - BUGFIX: One pole low- and highpass filters were producing weird responses for negative gains.
- Version 2.2.0
  - New midi in module.
  - New ++ poles mode for most filter types.
  - CPU usage optimization.
  - Reduced noise for high order filters.
  - Filter modulation is much more stable in general. Frequency movement no longer leads to gain buildup and drop.
  - BUGFIX: Automatable parameters now show up in the AAX automation menu.
  - BUGFIX: Fixed bad filter calculations for some extreme cases.
  - BUGFIX: Holding Alt only enters frequency solo mode if the plugin window is visible.

- Version 2.2.1
  - New spectral tilt filter with freely adjustable slope.
  - New tilt shelf filter.
  - New output clipper antialiasing algorithm.
  - Filter CPU optimizations.
  - Improved comb filter algorithm.
  - BUGFIX: various filter bugs solved.
  - BUGFIX: latency compensation sometimes not working.
- Version 2.2.2
  - New filter types: low- and highpass varislope.
  - Improved output clipper antialiasing suppression algorithm.
  - On Windows, text is rendered with more contrast.
  - BUGFIX: antialias output setting lost when restoring preset.
  - BUGFIX: peak filter not visible in secondary filter type menus.
  - BUGFIX: some buttons did not pick up colors from alternative color schemes.
  - BUGFIX: sound issues when using the spectral tilt filter with very low tilt settings.
  - BUGFIX: issues with automation and the vst3 version
- Version 2.2.3
  - New enable-button on top right which is hooked up to bypass on modern hosts.
  - All internal signals now use 64 bit precision.
  - Analyzer decay now compensates for FFT size.
  - Module controls can be changed using the mouse wheel.
  - Improved filter quality and performance.
  - Improved quality of the output clipper antialiasing.
  - Better ctrl-drag fine adjustment mode.
  - Additionally allow to use holding the 'S' key for the bandpass solo feature.
  - Improved the way the LFO module resets on transport start/stop.
  - BUGFIX: Issues with HiDPI drawing in the windows vst3 version.
  - BUGFIX: Increased CPU usage after loading big presets.
  - BUGFIX: Fixed loading old presets with comb filters included.
- Version 2.2.4

- Support for Notarization on OS X.
- BUGFIX: filter response weirdness when using peak filters with very low bandwidth in ++ poles mode.

- Version 2.3.0
  - New harmonic filter types.
  - High frequencies displayed in kHz + support for entering values like ”2k”.
  - Frequencies can be entered using pitch names like ”a4” or ”bb-1”.
  - BUGFIX: Crash when quitting WaveLab on OS X with multiple VST3 instances of apQualizr2 open.
  - BUGFIX: High samplerate projects did not reopen correctly when using the VST3 version in Cubase.
  - BUGFIX: Crash in FL Studio when bypassing apQualizr2 in the fx rack.
  - BUGFIX: Audio stutters with clipper antialiasing enabled in some VST3 hosts.
- Version 2.3.1
  - Compatibility with Apple Silicon Macs.
- Version 2.3.2
  - Added a few missing presets to the macOS installer.
  - BUGFIX: installer signing issues on older macOS versions.
  - Improved timing accuracy of host-automated parameters.
- Version 2.3.5
  - Added a preferences option to disable hint texts on empty displays.
  - Improved performance of the spectral tilt filter.
  - New cable dangle mode.
  - Multiple cables can now be attached to the same pin.
  - New optional output level meter.
  - BUGFIX: Occasional crashes when switching presets.
  - BUGFIX: Could not enter very long preset names and email addresses on windows dialogs.
- Version 2.4.0
  - New multi-row module view mode improves routing overview for complex presets.
  - Modules can now snap right or left to adjacent modules.
  - The add module button in the tray now includes the filter band module.

  - Double-click-drag creates and edits a new filter band in one mouse gesture.
  - New: Create filter bands by dragging from the sum curve.
  - Add dynamic gain modulation with one click from the filter handle context menu.
  - The preset cycle buttons can optionally cycle through the entire preset structure.
  - LFO module: optionally resync on playback time jumps (including cycle playback)
  - Provide better info to hosts about changing parameter names.
  - BUGFIX: module parameter cleanup issues when creating more than 32 modules.

- Version 2.4.1
  - Added 3dB and 6dB filter response curve viewing ranges.
  - BUGFIX: tooltip scaling on windows when using high-dpi displays.
  - BUGFIX: preferences label no/show data hints.
  - BUGFIX: rare high frequency filter matching issue.
- Version 2.4.2
  - Zoom frequency range using the mouse wheel while hovering over the frequency ruler.
  - BUGFIX: Filter band bypass state mismatch between processing and gui.
- Version 2.5.0
  - New phase correction engine with minimal, analog, mixed and linear phase modes.
  - Improved harmonic filtering accuracy.
  - Improved output clipper quality.
  - The stereo mode can be changed in the filter band context menu.
  - Hide vertical scrollbar in multirow mode with <3 rows.
  - Optional display of the note name on the frequency ruler mouse guide.
  - BUGFIX: module drag-image size and resolution on HiDPI screens.
  - BUGFIX: filters internally reverting to defaults when switching AUX mode.
  - BUGFIX: harmonic bands filter low frequency stability.
  - BUGFIX: modulating the harmonic step did not correctly update the display.
  - BUGFIX: wrong poles value display after switching filter types.
  - BUGFIX: easier selection of short cables.

  - BUGFIX: sharpness of display rulers on HiDPI screens.
- Version 2.5.1
  - BUGFIX: Reenabled 64-bit audio host calls on vst and vst3.
- Version 2.5.2
  - BUGFIX: Work around a plugin loading issue on macOS 10.11 and 10.12.
  - BUGFIX: Loading plugin settings not working correctly in Cubase and FL Studio when phase correction modes are used.
- Version 2.5.3
  - AAX version compatible with Apple Silicon native Pro Tools.
  - Added double-click filter creation type zones with mouse cursor hints.
  - Keyboard display works much more like the main display including filter creation, context-menus and multi-selection.
  - Filter bands created during freq-solo audition mode (holding alt) use the current freq-solo bandwidth.
  - All phase-correction modes and the frequency analyzer use less CPU.
  - Better display of numerical parameters. Exact values no longer display trailing zeros.
  - Added ctrl-right-click as an alternative default value mouse shortcut.
  - Midi input modules can now be created from input pin context menus.
  - Auto-expansion of the filter view range during drag operations is reversible when dragging back near the center line.
  - macOS installer no longer requires Rosetta2 installation on Apple Silicon macs.
  - Added guarding against invalid sample values.
  - BUGFIX: The phase graph display didn't correctly update after samplerate changes.
  - BUGFIX: Removing a connection deactivated modulation for pins with multiple connections.
  - BUGFIX: Dynamics ratio default value is now 1:1.
- Version 2.6.0
  - User-ID no longer required/displayed on the GUI.
  - Added a sidechain module to provide the sidechain signal.
  - Added a sidechain (ex) analyzer display option.
  - Added an aux input to the output module to route parallel signals to the output.

- Added a gain filter band type.
- Band pass and harmonic band filters now have -inf dB as the minimum gain.
- Improved spectrum graph drawing.
- Improved handling of drag-to-create from the sum curve.
- Keyboard entry now accepts both ',' and '.' as decimal separators.
- Updated AAX SDK to work around automation issues in newer versions of Pro Tools.
- BUGFIX: For zones of very low gain, analog and mixed phase correction was broken on Windows.
- BUGFIX: Channel layout error when switching presets in Wavelab 11.
- BUGFIX: The phase plot didn't work correctly in linear mode for bands in mid or side configuration.