Equalizer Pro
Installation and user manual

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2 Setting up the software for first use

2.1 Installation on desktop (Windows, macOS)

Download and install the freely-available trial/evaluation versions of the software. This will allow you to test the plugins prior to making any purchase decisions. trial/evaluation versions can be downloaded from the ToneBoosters.com website. Trial/evaluation plugins may have one or more of the following limitations:

- Trial/evaluation versions will not store nor save settings.
- Trial/evaluation versions will show a reminder to purchase a license.

2.1.1 Installing the trial/evaluation software

- The software will be installed in the locations given in the table below.

Operating system	Folder(s)

Windows*	`C:\Program Files\Common Files\VST2\ToneBoosters\`
	`C:\Program Files\Common Files\VST3\ToneBoosters\`
	`C:\Program Files\Common Files\Avid\Audio\Plug-Ins\ToneBoosters\`
	`C:\Program Files\ToneBoosters\`

Mac OS	`/Library/Audio/Plug-Ins/Components/`
	`/Library/Audio/Plug-Ins/VST/`
	`/Library/Audio/Plug-Ins/VST3/`
	`/Library/Application Support/Avid/Audio/Plug-Ins/ToneBoosters/`
	`/Applications/ToneBoosters/`

2.1.2 System requirements (Windows)

- Microsoft Windows 10 or higher
- 64 bit host program that supports 64-bit VST, VST3 or AAX plugins

2.1.3 System requirements (mac OS)

- macOS 10.13 or higher; Intel or Apple Silicon processor
- 64-bit host program that supports 64-bit VST, VST3, AAX or Audio Unit plugins

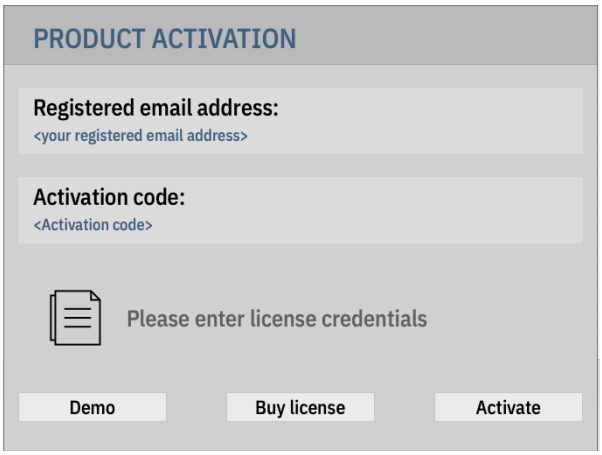
2.1.4 Configuring your host program / DAW

After installation of the trial/evaluation plugins, you may have to inform your host program about the presence of new plugins. Most host programs require you to provide the folder where plugins are installed.

- Consult your host program manual how to configure plugin folders. On Windows, make sure you add the following VST scan path to your host program settings if not already present:
  `C:\Program Files\Common Files\VST2\`
- Refresh and/or re-start your host program to allow it to scan for new plugins on your computer.

2.1.5 Activating a license

To run software in registered mode without demo limitations you can acquire an activation code in the ToneBoosters online store. The activation code is sent by email after purchase and consists of a long alphanumerical string that should be copied into the designated field when opening a plugin from within a host program. The user interface that has the entry field for your email and an activation code is shown in the figure below. Make sure that the email address entered is the same as the email address used for purchasing the software.



After entering your email and the activation code in the designated fields, click on 'Activate' to activate your license. Alternatively, without a valid activation code, you can try the plugin in demo mode.

2.1.6 Installing updates

Simply download and run a more recent software installer from www.toneboosters.com to update.

2.2 Installing on iPad, iPhone or iOS device

2.2.1 Installation

Follow the instructions from the Apple App store to install the app on your iOS device. The following versions will be installed:

- Stand-alone app, compatible with inter-app audio (IAA) protocol
- Audio Unit v3 (AUv3) app extension

2.2.2 System requirements for iOS devices

- iPad or iPhone; iPad highly recommended
- iPadOS or iOS version 12.0 or up
- Host program compatible with AUv3 plugins

2.3 Multi-channel and external side chain support

Some plugins support multi-channel audio input, possibly including 5.1, 7.1, and Dolby Atmos channel-based formats. Plugins may also provide support for an external side chain input. In general, the following restrictions apply for the various plugin formats:

- VST: only mono or stereo input; no external side chain input
- VST3: can typically run up to 16 channels, and can have external side chain inputs
- AAX: can typically run up to 16 channels, and can have external side chain inputs
- Audio Unit: can typically run up to 16 channels, and can have external side chain inputs
- iOS stand-alone app: only mono or stereo input; no external side chain input
- iOS AUv3: mono or stereo input, with external side chain support.

Not all plugins will support multi-channel audio or external side chain input; some plugins provide high-channel count input. Please refer to the web page and manual of the respective ToneBoosters product to see what configurations are supported. Please refer to your host program's manual to see what channel and external side chain input configurations are supported by the host program.

2.4 Sample rate support

- Desktop plugins (VST2, VST3, AU, AAX for macOS or Windows) typically support any sampling rate between 32 kHz and 384 kHz.
- iOS stand-alone and AUv3 typically support any sampling rate between 32 kHz and 96 kHz.
- Android stand-alone apps typically support sample rates between 24 kHz and 48 kHz, as determined by the operating system.

3 User interface common controls

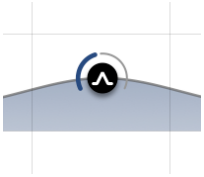
3.1.1 Controlling Knobs and sliders

The various knobs and sliders on the graphical user interfaces (GUIs) of the plugins can be controlled by left-mouse clicks (for switches) or left-mouse drags (for rotary controls and sliders). The following key combinations apply that modify the behavior of the GUI elements:

3.1.2 Sliders

- Drag up/down or left/right to change the value
- Hold 'Shift' for fine-control micro edits
- Double click to reset to the default value
- Click on the numeric value to allow manual entry of values trough the keyboard.

3.1.3 Draggers / nodes



- Drag left/right or up/down to change the position and value
- Hold 'Shift' for fine-control micro edits
- Double click to reset to the default position
- Use mouse wheel to change the rotary value (for example the Q value)
- Hold 'Control' to allow for vertical movement only
- Hold 'Alt' to allow for horizontal movement only
- Left mouse click to activate (if supported)
- Right mouse click to de-activate (if supported)

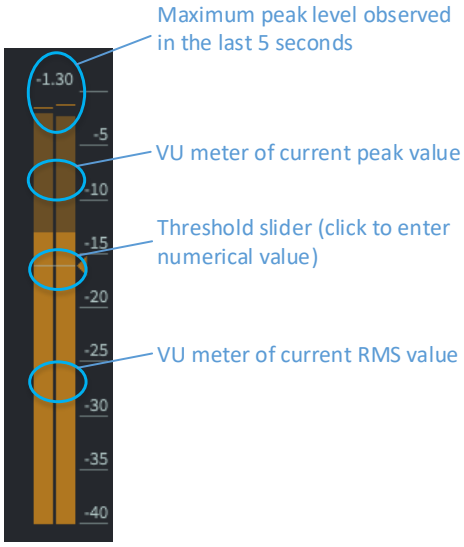
3.1.4 Lasso

Some plugins may provide a lasso function to select multiple draggers/nodes and move them all simultaneously. To activate the lasso, drag around the nodes you want to capture. To release the lasso, click once outside the lasso range.

3.1.5 VU meters

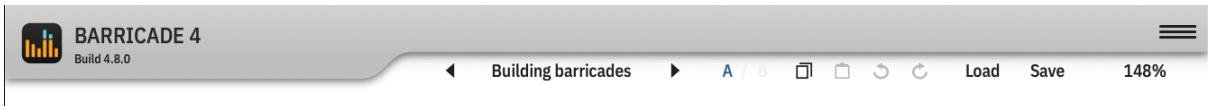
Plugins equipped with VU meters will typically show both peak and RMS levels in the same meter. The VU meters will show two segments with slightly different colors. The lower segment is the RMS (root-mean-square) value; the peak value is shown by the upper segment. Peak hold values will be shown numerically and by two horizontal lines in the meter. RMS levels are unweighted and computed with a 400-ms time constant.

Click on the VU meter to reset the peak hold value (if supported).



3.1.6 Tool bar

ToneBoosters apps and plugins have a common tool bar at the top of the user interface.

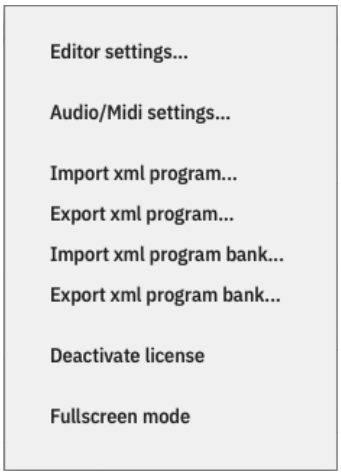


The tool bar shows the plugin version number, and will also indicate if a newer version is available. The second row contains the following tools:

- **Program selector**: clicking on the program selector can trigger one of two different actions, depending on the app/plugin's architecture:
    - For plugins/apps that can hold multiple programs simultaneously, a dropdown menu will appear that lists all programs currently loaded. Click any of them to switch between currently loaded programs.
    - For plugins/apps that can hold only one program at a time, a preset manager will appear to browse and load programs from disk (factory or user programs).
    - Double click on the program name to edit it.
- **A/B toggle** to quickly toggle between two configurations (A and B). This toggle button can be used as follows:
    - Click on the A/B toggle switch to activate configuration A.
    - Modify any plugin parameter that you would like to be set for configuration A.
    - Click on the A/B toggle switch to activate configuration B.
    - Modify any plugin parameter that you would like to be changed for configuration B.
    - You can now quickly switch between configurations A and B by clicking on the A/B toggle switch.
- **Undo**. This will undo previous changes. You can also use control-z or command-z.
- **Redo**. This will redo changes that were undone previously. You can also use control-y or command-y.
- **Copy**. This will copy all current parameter settings to the internal clipboard. You can also use control-c or command-c.
- **Paste**. This will copy the settings in the clipboard to all parameter settings. You can also use control-v or control-v.
- **Load**. This will bring up the program manager to browse or load a program.
- **Save**. This will bring up the program manager to save the current program.
- **Scale factor** for the user interface
    - Click to open a drop-down menu with commonly used scale factor settings;
    - Double click to enter a numerical value.

3.1.7 Hamburger menu

The hamburger menu opens up a drop-down menu with various options.



- **Editor settings**: open a new dialog to change a variety of editor settings
- **Audio/Midi settings**: open a panel to set up audio and midi inputs (only when opened as app; not available when opened as plugin)
- **Import / export xml program**: import or export the current preset/program as xml file using the native file browser. You can use these exported xml file to import presets on other computers or platforms.
- **Import / export xml program bank:** import or export all currently loaded presets/programs as xml file using the native file browser. This option is not available in plugins that only have one preset slot in a preset bank.
- **Deactivate license**: remove your license from the current computer. Only available if software licenses were activated. Not available on iOS.
- **Fullscreen mode**: switch to fullscreen mode. This option is only available on macOS and Windows.

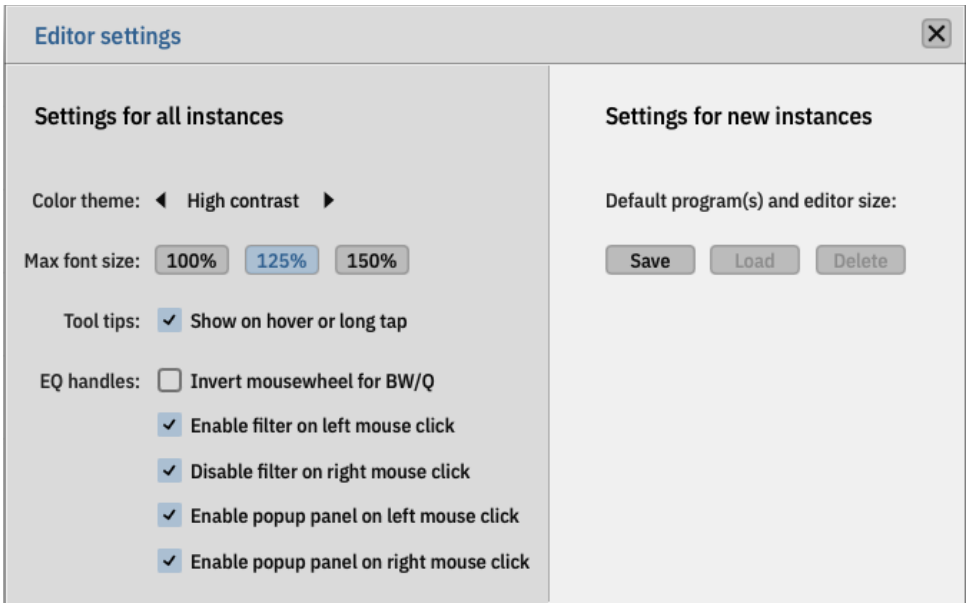
Programs and program banks are imported and exported as xml files. XML files are intended to be cross-platform compatible, e.g. a program (bank) exported on iOS can be imported into the desktop version of the same plugin and vice versa.

3.1.8 Editor settings dialog

The editor settings dialog holds a variety of configurable settings that determine how a plugin editor behaves and interacts with the mouse.

The left half of the panel has settings that apply to all (current and future) instances, and includes setting a color theme, a maximum font size, and select triggers related to mouse button and mouse wheel interactions.

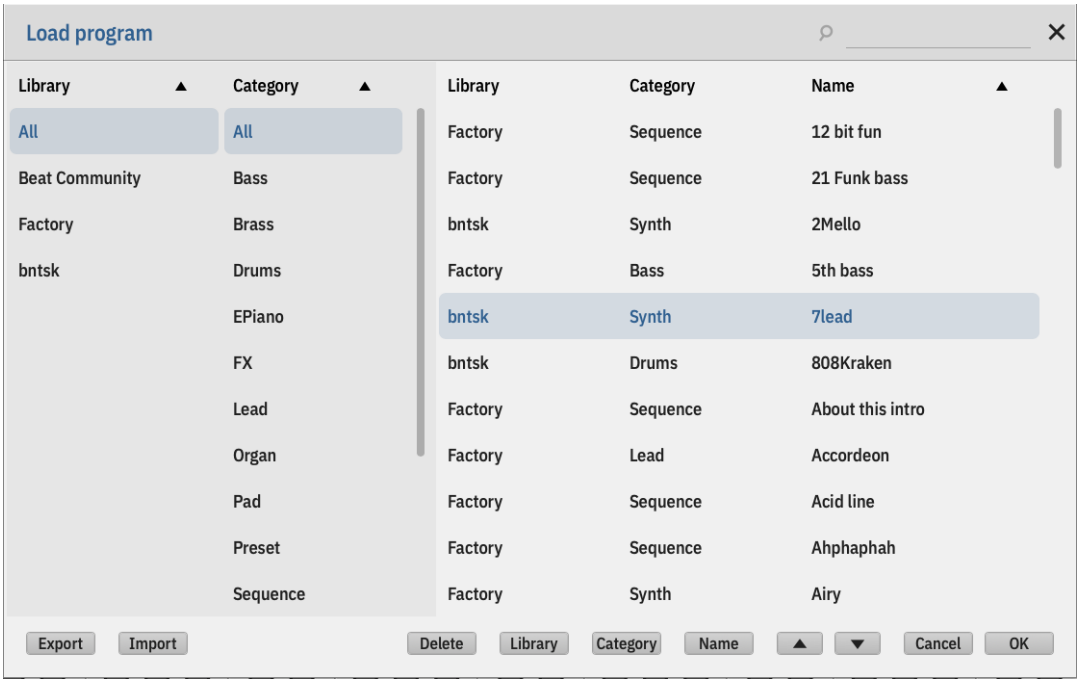
The right half of the panel allows you to save, load and delete default program(s) and the editor size for **new** instances when created in your digital audio workstation. If the 'Load' and 'Delete' buttons are disabled, no default program(s) and editor size were saved.



3.1.9 Program manager

The program manager allows you to load, categorize and browse preset and user programs. The two columns on the left allow you to select a library and a category as filters; the resulting set of available programs will show up in the 3 columns on the right.

- Click a program to load it, while keeping the program manager open.
- Double click a program to load it and close the program manager.
- Use the Exit button (cross in the right top corner) to exit the program manager.
- Use the Ok button to close the program manager.
- Use the Cancel button to close the program manager and revert to the program that was loaded prior to opening the program manager.
- Use the up/down buttons to navigate to the previous or next program in the list.
- Use the Name button to rename a program.
- Use the Category button to change the category of a program.
- Use the Library button to change the library name of a program.
- Use the Delete button to delete a program.
- Use the search text entry at the top to filter the list based on a text query.
- Drag a program to a different category or library to move it (this does not work with factory libraries)



3.1.10 Export programs in the program manager

The program manager can export all currently visible patches in one single zip file. You can use this zip file for backup or to share programs with other users.

- To export all patches, click on 'All' in both the Library and Category selectors. Then click on Export to export all patches visible in the list.
- To export one library only, select the Library, select 'All' in the Category selector. Then click on Export to export all patches from the selected library.
- You can make any selection using the Library and Category selectors and text search query method to create a subset of patches. This selection can then be exported as one zip file.
- When you click on Export, you will be asked where to store the zip file with the patches on your computer or mobile device.
- Export will maintain the library and category structure of the exported programs.

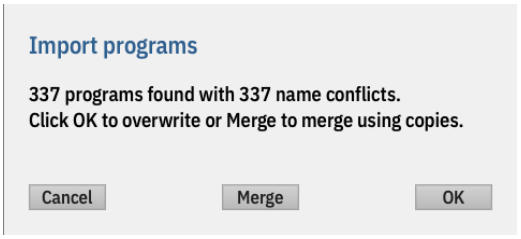
3.1.11 Import programs in the program manager

Use the Import button to import a zip file containing patches that was created using the Export programs option.

- Click on Import to select a zip archive stored on your computer or mobile device.
- After a short analysis time, you will see how many patches were found in the selected zip file.

If you are importing programs that cause a name conflict (e.g. having the same combination of library, category and program name as programs already saved), you will be asked how these conflicts will need to be handled:

- Click Ok to overwrite existing programs with the imported programs;
- Click Merge to automatically create a copy of an imported program if there is another program with the same name that was already present.
- Click cancel to abort import of programs.



4 Equalizer Pro

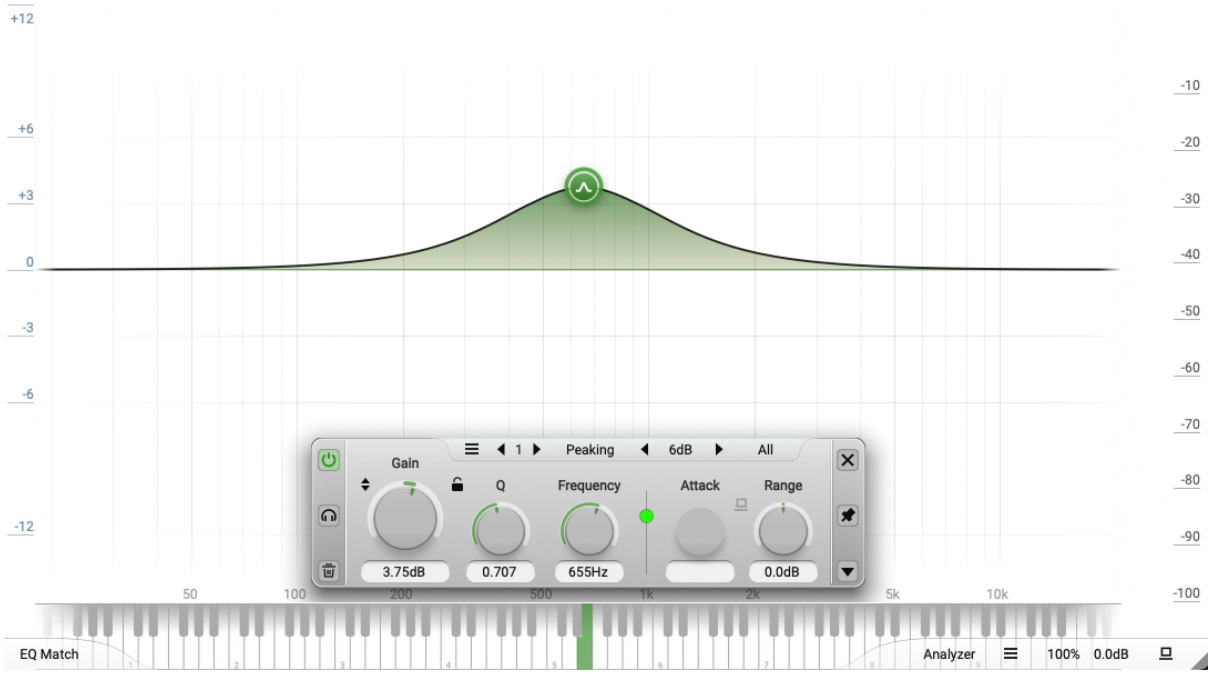
4.1 Controlling filter parameters

4.1.1 Introduction

In Equalizer Pro, one or more filter sections can be enabled that each have a specific frequency response. The frequency response (in dB as a function of frequency) of each active section is shown in the plug-in's user interface. Each section will have a handle with an icon to indicate the filter type that you can drag around to change the frequency and the gain parameter, or use the mouse wheel to change the Q (or bandwidth) value.

Use can use control/alt/shift modifier keys on your keyboard to enable zoom/fine edit or unidirectional edit (e.g. allowing modification of gain only or frequency only, while keeping the other parameter fixed).

When a node is selected by clicking on it, a section control panel will appear with advanced settings that are associated with the selected filter section.



The filter section control panel has three buttons on either side. The buttons on the left, from top to bottom, will:

- Enable or disable the filter section
- Switch to 'Audition' mode to listen to the affected frequencies of this filter in isolation
- Delete the filter section

On the right-hand side, the three buttons will:

- Close (remove) the section control panel
- Dock / follow the section handle, or allow you to move the panel when not in docking mode
- Minimize the panel

The middle section of the control panel has sliders and numerical displays for gain, frequency, Q, attack (or release), and range parameters, and a level VU meter for filter types that have a threshold parameter, more details on this below. You can double click any of the numerical displays to manually enter values. Double click a slider to reset the parameter value. The gain slider has further 'invert gain' and 'lock gain' controls.

In addition, at the top of the control panel there is a filter section edit menu button, a selector of the filter section index, a filter type selector menu button, a filter slope selector menu button, and a spatial configuration button.

- Some controls will be disabled automatically depending on the selected filter type. For example, notch filter types will not have a gain slider.
- Use the 'invert gain' button to quickly change the gain from positive to negative values or vice versa to listen to the opposite effect.
- Use the 'lock gain' control to lock the gain value when dragging a section node around. The gain slider itself will always allow you to change the gain, irrespective of the use of the 'lock gain' option.

The 'lock gain' control has 3 different states that can be changed by clicking on the button multiple times:

- No lock: the equalizer handle can be freely moved up and down to change the gain
- Lock: the equalizer handle can only be moved horizontally
- Snap to zero: the equalizer handle will 'snap' to zero gain

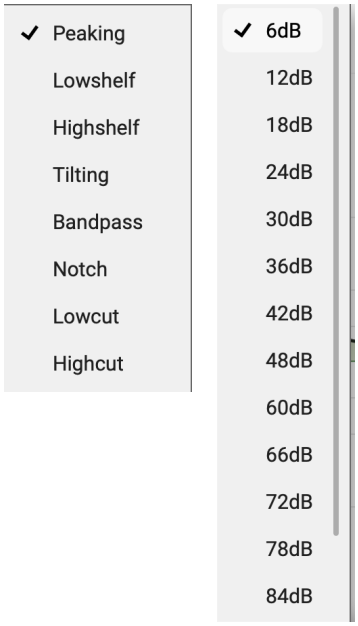
4.1.2 Filter section edit menu

The filter section edit menu provides some useful filter edit operations. These include resetting a filter section, removing a filter section, inverting the gain value, or setting the gain value to 0 dB.

Further options include:

- Duplicate in place: add a new filter section with identical settings as the currently selected one.
- Duplicate +1 oct: add a new filter section with identical settings as the currently selected one, but with a frequency that is one octave higher.
- Duplicate -1 oct: add a new filter section with identical settings as the currently selected one, but with a frequency that is one octave lower.
- Split L/R: this option will only be enabled when running on a bus or track that has two or more channels, and splits the current section into a left-channel and right-channel filter section.
- Split M/S: this option will only be enabled when running a bus or track that has two or more channels, and splits the current section into a mid channel and right channel filter section.
- Split frequency: this option is only available for peaking filters, and splits the filter into two peaking filters with a very similar overall frequency response. This option works best for higher-order filters.
- Make lowshelf: this option is only available for peaking filters, and converts a peaking filter into a low-shelf filter with the same spectral shape above the center frequency as the original peaking filter.
- Make highshelf: this option is only available for peaking filters, and converts a peaking filter into a high-shelf filter, with the same spectral shape below the center frequency as the original peaking filter.
- Make dynamic: change the filter type into a dynamic filter. This is only available for peaking, lowshelf and highshelf filters.
- Make static: change a dynamic filter type into a static filter. This option is only available for dynamic filters.

4.1.3 Filter type selection menu

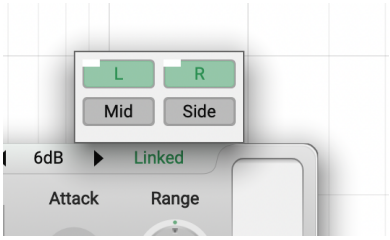


The filter type menu allows selection of one of the many filter types available in Equalizer Pro. You'll find all the basic filter types such as peaking, shelving, tilting, bandpass, notch, low-cut and high-cut filters in the menu, and a range of context-aware filters in the Peaking, Lowshelf and Highshelf submenu – more on those below.

Use the filter slope menu to select the filter steepness, expressed in decibels per octave. The slope can be varied between 6dB/oct up to 96dB/oct.

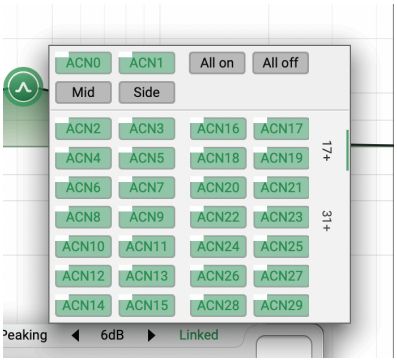
4.1.4 Spatial filter configuration

Each filter section can be configured to run on all channels, a single channel, or a set of channels. Additionally, the first two channels can be configured to run in mid/side mode, filtering mid only, or side only. Click on the Spatial Configuration button to bring up the Spatial Configuration editor.



For a stereo bus, four buttons will appear to configure the filter to apply processing to left (L), right (R), Mid or Side channels.

The input channel buttons have a channel activity indicator in the top-left corner to indicate whether a channel contains a signal or whether it is silent.



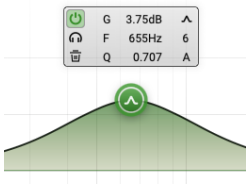
Channel configurations with more than 16 channels will trigger an extended spatial filter configuration panel, with tabs to go through the channels, and additional 'All on' and 'All off' buttons to enable or disable all channels simultaneously.

The Spatial Configuration editor will display channel names if these are provided by the plug-in host, or show numbers if unavailable.

4.1.5 Separate EQ curves for left, right, mid and side channels

When Equalizer Pro operates on a stereo bus, and you have selected one or more filter sections to operate on left, right, mid or side, a separate overall curve will appear to reflect overall EQ for a specific spatial configuration. This option will not be available for mono busses, nor for multi-channel bus formats.

4.1.6 Filter section handle context menu



When clicking on a filter section handle, a context menu will appear containing a range of convenient options to quickly modify the frequency, gain, Q, filter type, slope, and channel configuration. In addition, an on/off, audition and delete button are provided as well.

4.1.7 Filter section handle Q drag mode

Double-clicking on a filter section handle will enable the Q-drag mode. In this mode, any drag operations will only change the Q value of a filter, without modifying frequency or gain. This is especially useful on touch screens to quickly modify Q. Double click again to disable the Q drag mode. The Q drag mode status will be shown above the section handle.

4.1.8 Filter section handle modifier keys

- Shift: fine edit mode. Mouse movements are converted to small changes in gain, frequency or Q for fine-control edits
- Control: vertical mode. Filter section handles can only be moved vertically, keeping their frequency parameter constant. In this mode, you can use the mouse wheel to change the dynamic range parameter value.
- Alt: horizontal mode. Filter section handles can only be moved horizontally, keeping their gain parameter constant.

4.1.9 Filter section handle mouse wheel functionality

- For peaking and shelving filter types, and notch / bandlimit filter types, the mouse wheel can be used to change the Q value.
- For low-cut and high-cut filter types, the mouse wheel can be used to modify the filter steepness (slope) parameter.
- For dynamic filter types, and using the Control key modifier, the mouse wheel can be used to change the dynamic range parameter value.

4.2 Adding new filter sections

You can add new filter sections by double clicking an empty area. The new filter type will depend on where the double click occurs:

- Double-click on an empty area in the main window. Depending on where you click, a different filter type will get instantiated.
- Double-click in the lower-left corner to create a new low-cut filter.
- Double-click in the lower-right corner to create a new high-cut filter.
- Double click at low frequencies, at the upper ¾ of the gain range to create a new low-shelf filter.
- Double click at high frequencies, at the upper ¾ of the gain range to create a new high-shelf filter.
- Double click anywhere else to create a new peaking filter.

Alternatively, you can ‘nudge’ the overall filter curve to create a new filter type. Nudging at low frequencies will create a new low-shelf filter. Nudging at high frequencies will create a new high-shelf filter. Anywhere else will create a new peaking filter.

- Note that if no filter section is visible, the overall equalizer curve will be hidden automatically and nudging that filter curve isn’t available.
- The maximum number of supported filter sections is 24.

4.3 Changing multiple filter sections simultaneously

Drag around two or more filter section handles to select them with the lasso function. One two or more sections are selected, you can modify frequency, gain, and Q values of all selected sections simultaneously by dragging the lasso around, or using the mouse wheel function.

Micro edit (using shift), or unidirectional dragging modes using control/alt keyboard modifier keys are also available when working with the lasso function.

4.4 Static and dynamic filters

4.4.1 Gain and Range controls

The peaking, lowshelf and highshelf filters can be used as static and as dynamic filters. In dynamic filter mode, the filter section gain will depend on the main bus input signal level. In static filter mode, the filter curve will be constant, irrespective of the input signal characteristics.

- To change a peaking, lowshelf or highshelf basic filter from static to a dynamic type, choose a non-zero value on the 'Range' slider.
- Resetting the 'Range' rotary slider value to zero (for example by double clicking the slider) will revert back to a static filter type.

When a filter section is in dynamic mode, the 'Range' parameter value determines the change in gain that is applied in addition to the 'Gain' parameter value. In other words, the momentary gain can be different from the value set by the 'Gain' control. For example:

- With a Gain control value of +3dB, and a Range value of -2dB, the momentary gain will be +3dB for input signal levels below the threshold, and will gradually decrease by -2dB to a final (target) value of +3-2 = +1dB if the input signal level is substantially above the Threshold. Thus, in this example, the effective behavior is a dynamic *compression* with a dynamic range of -2dB.
- If we change the Range value to +4dB, the total (momentary) gain will again be +3dB for signal levels below the threshold, gradually changing to +3+4=7dB for input signal levels substantially above the threshold. Therefore, with a positive Range value, the filter acts as an *expander*.

To summarize, and as a general rule for dynamic filters:

- A positive 'Range' value adds momentary gain for signals above the threshold value, creating an expansion effect.
- A negative 'Range' value reduces momentary gain for signals above the threshold value, creating a compression effect.
- The effect of the Range parameter is always added to the Gain parameter value.

4.4.2 Threshold and Attack controls

The section control panel contains a VU level meter that shows the input signal level when a filter section is set to dynamic mode. In addition, the threshold value, above which the Range will get into effect, is available as vertical slider within the VU meter.

The Attack control provides means to determine how quickly the momentary gain will respond to increases in signal level. A long attack time will take a longer time for the change in gain denoted by the Range parameter to kick in.

Dynamic filters do not have a ratio or a release time. Ratio and release parameters are fully automatic, and signal dependent to increase your workflow and get results quickly.

4.5 Advanced filter types

Equalizer Pro contains a variety of innovative advanced filter types that provide more control over specific signal elements or characteristics than static or dynamic filters. You can find these in the 'Advanced' filter type submenu.

4.5.1 Brickwall filters

Brickwall filters come as low-cut and high-cut variants, and provide steeper (more aggressive) removal of low or high frequencies than the basic low-cut or high-cut filters, even when set to the highest filter steepness value. Brickwall filters therefore do not have a steepness parameter.

4.5.2 Makeup filters

Makeup filters are available for peaking, lowshelf and highshelf variants, and apply a loudness compensation gain to reduce the change in overall loudness as a result of EQ-ing. For example, for a peaking filter with a positive gain value, the overall EQ curve will be shifted downwards slightly. The amount of compensation depends on the gain, frequency and Q values of the filter. When running in dynamic mode, the automatic makeup gain

compensation is computed from the static part only (e.g. excluding any effects of dynamic, signal-dependent gain offsets).

4.5.3 Transient filters

Transient filters are provided as peaking, lowshelf and highshelf variants, and aim at equalizing transient components of the input signal only, while keeping elements that have a sustaining or decreasing level in a mix untouched. To control what signal elements are considered transient, Transient filters have two additional controls:

- A Threshold parameter: signal level changes that are above the Threshold level are considered transient elements and will be modified by the Transient filter type; and
- A Release parameter which determines the time required to transition from a transient section to a non-transient section. A higher value will cause the transient gain to be applied longer across time than a shorter value.
- Transients can be reduced or increased in level, by setting the Gain parameter to a negative or positive value, respectively.

4.5.4 Sustain filters

Sustain filters are the complimentary filter to Transient filters. Sustain filters will only equalize during sustaining or decreasing signal levels, and will leave transient components untouched. Similar to the Transient filter type, Sustain filter types have two additional controls:

- A Threshold parameter: signal level changes above the threshold will be considered transient and will not be filtered by sustain filters.
- A Release parameter which determines the time for the sustain filter to kick in after a transient occurred.
- Sustaining elements can be reduced or increased in level, by setting the Gain parameter to a negative or positive value, respectively.

4.5.5 Sibilance filters

Sibilance filters aim at enhancing or reducing frequency components that are higher than expected, based on the observed input signal. Sibilance filters can for example be used as a very flexible de-esser that can work on any frequency range. To control what signal elements are considered sibilant, Sibilance filters have two additional controls:

- A Threshold parameter: signal levels above the threshold are considered sibilant.
- An attack parameter, determining how quickly the Sibilance filter reacts to the presence of sibilant components.

4.5.6 Background filters

Background filters will equalize signal components that are relatively low, while keeping high signal levels untouched. In combination with a positive 'Gain' value, low signal levels will cause a positive momentary gain, acting like an **upward compressor**. A negative 'Gain' value, on the other hand, will attenuate low signal levels. This filter type is especially suitable to enhance or reduce background signal components. To allow for more precise control, Background filters have two additional parameters:

- A Threshold parameter: input signal levels below the threshold are considered to be background components and are therefore equalized, while signal levels above the threshold are unaffected.
- An Attack parameter determining how quickly the filter should respond to signal level reductions, and start to apply the momentary gain.

4.5.7 Gate filters

Gate filters have some resemblance with Background filter types, as they aim at equalizing signal levels below a specific threshold. Gate filters will however respond to changes in signal level in a much more pronounced way than background filters, where the 'Gain' parameter kicks in very quickly for input signals below the threshold, and will be released as soon as the input signal level reaches the Threshold value. To provide further control, Gate filters have two additional controls:

- A Threshold parameter: input signal levels below the threshold will be equalized, while signal levels above the threshold are not filtered.

- A release parameter which determines how long it takes for the gain to go from the set value to a gain of zero dB whenever the input signal level is above the Threshold value.

4.5.8 Direct filters

Equalizer Pro uses advanced spatial signal decomposition methods to separate direct and ambient signal components from stereo or multi-channel content. Direct filters will equalize the resulting direct components only, without applying gain or attenuation to the ambient components. The Direct filter type will be unavailable when the input signal is monaural.

4.5.9 Ambiance filters

Ambiance filters complement Direct filters; Ambiance filters only equalize the ambient signal components while not filtering the direct components. Ambiance filters are unavailable when the input signal is monaural.

4.5.10 External side chain filters

External side chain filters operate in an identical way as dynamic filters, with the exception that the signal level is determined from the wide-band external side chain input (if provided by the host program).

4.5.11 Unmask / gain riding filters

Unmask filter types are ideal for automatic gain riding of elements in your mix. The unmask filter type computes the amount of auditory masking caused by the *external side chain* input signal. The unmask filter type allows to compensate for such masking automatically by increasing the momentary gain of a filter section of the track it is applied to.

When the side chain input signal gets louder, the amount of masking will *increase*. Conversely, if the track that Equalizer Pro is operating on gets louder, the amount of masking will *decrease*. The unmask filter type will show the amount of auditory masking continuously in the panel's VU meter.

You can set the threshold value in the VU meter to determine at what masking level, masking compensation should start. Whenever the amount of masking is *above* the set threshold level, the unmask filter will increase its momentary gain value. Use the 'Range' control to determine the maximum change in momentary gain you will allow to compensate for masking effects. If the 'Range' control is set to zero, no masking compensation will occur and the filter type will essentially become a static filter.

When the 'Range' control is set to a *positive* value, the momentary gain applied by the filter will be equal to the value set by the 'Gain' control when the amount of masking is *below* the threshold, and will increase when above the threshold, with a maximum increase amount determined by the positive 'Range' control.

Similarly, for a *negative* range value, and when the amount of masking is below the threshold, the momentary gain will be below the value of the 'Gain' control. When masking increases above the set threshold, the momentary gain will increase and top out at the value given by the 'Gain' control.

4.5.12 Linear phase filters

Linear phase filters are provided for a range of filter types, such as shelving and peaking filters. Linear phase filters are typically used for mastering purposes. Due to the inherently symmetric impulse response of linear filters, these filters will introduce a latency. This latency can be compensated for by your digital workstation host program. You can choose the trade-off between impulse response length and latency in a dedicated 'Linear Phase Settings' panel. This panel will allow you to choose a range of quality/latency settings, and will also depict the total latency introduced by one or more linear phase filter types. Linear phase filters can be freely mixed with other, zero-latency filters.

4.6 Interactive piano keyboard

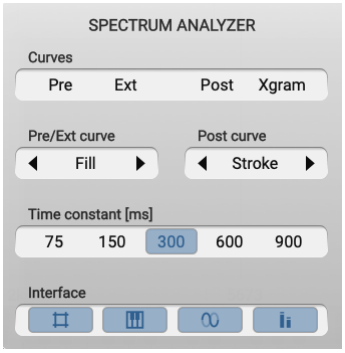
The piano keyboard can be used in various ways:

- Drag a highlighted note to change the frequency of a section, with frequencies quantized to MIDI note frequencies
- Double-click on a non-highlighted note to create a new equalizer filter section.
- When hovering the equalizer graph, the piano keyboard will highlight a note corresponding to the frequency of the current mouse position.

4.7 Configuring the frequency spectrum analyzer

The visual equalizer gain range can be modified simply by dragging the gain axis on the left of the user interface up or down. Similarly, the dynamic range of the input/output VU meters and the spectrum analyzer can be modified by dragging the level axis on the right of the user interface up or down.

4.7.1 Spectrum analyzer configuration panel



The frequency spectrum analyzer can show:

- The input frequency spectrum
- The frequency spectrum of the external side-chain input
- The output frequency spectrum
- The output spectrogram

To enable, disable or configure the analyzer, open the configuration button at the bottom of the user interface to open the configuration panel.

The Spectrum Analyzer configuration panel has controls to select what curves are shown, and how the data is visualized.

- Select ‘Pre’ to show a pre-equalizer frequency spectrum
- Select ‘Ext’ to show the frequency spectrum of the external side-chain input
- Select ‘Post’ to show the post-equalizer frequency spectrum
- Select ‘Xgram’ to show the output spectrogram

The third row represents the averaging time constant for the spectrum analyzer, which can be set to values between 75 and 900ms.

The 4 buttons at the bottom modify how the interface operates:

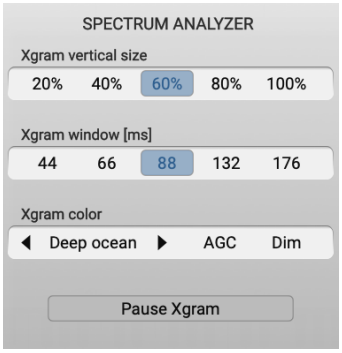
- Enable or disable cross-hair functionality when hovering the frequency spectrum
- Enable or disable the piano keyboard
- Enable or disable visualization of dynamic equalizer gains
- Enable or disable the input/output VU meters

A second tab in the Spectrum Analyzer configuration panel has even more settings to configure the spectrum analyzer and gain visualization. These include:

- Tilt: set the spectral tilt in dB/octave
- Resolution: set the resolution in octaves, or use ‘Lin’ to switch to a linear frequency resolution
- Dynamic range: set the visible (vertical) dynamic range
- Gain range: set the visible equalizer gain range

The visual dynamic range can *also* be modified by dragging the gain axis or level axis up and down.

A third tab is dedicated to the spectrogram settings.



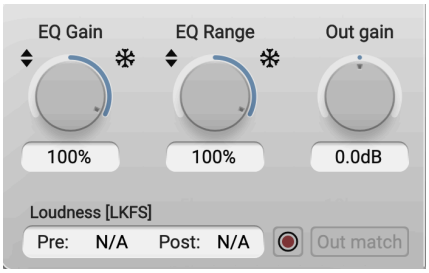
- Xgram vertical size sets the height of the spectrogram
- Xgram window sets the spectrogram temporal analysis window duration
- Xgram color allows to switch the color scheme, apply automatic gain control (AGC) or dim colors.
- Pause temporally pauses the spectrogram from scrolling.

4.8 Gain scaling and loudness management

4.8.1 Gain and dynamics scaling

Click on the output level button at the bottom-right of the user interface to bring up the level control panel. This panel has an output gain slider, as well as two gain scale sliders.

- The EQ Gain scale slider determines what percentage of gain is applied in individual sections. For example, if a specific peaking filter section has a gain value of 10dB, and the EQ Gain scale control is set to 30%, the actual gain applied by that specific section is 30% of the parameter value of 10dB, which equals 3dB.
- Similarly, the EQ Range slider scales all Range parameter values across all sections.
- Use the flip button to quickly toggle between a positive scale and negative scale value to hear the opposite effect of your equalization settings.
- Use the Freeze button to modify all gain values to their scaled variants, and reset the scale parameter back to 100%.



4.8.2 Loudness management

ITU bs.1770-compliant loudness analysis is available both pre and post equalizer. Click on the record button to reset and start loudness measurement, and click again to stop the measurement. The measured pre and post-equalizer loudness values are shown in realtime.

When measurement data is available, the 'Out match' button will become available to modify the Out gain slider value to match input and output loudness.

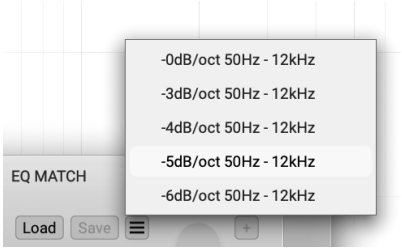
4.9 EQ learn and match

The EQ Match panel allows learning and matching of (average) frequency spectra across tracks. The EQ Match panel has two sections:

- Reference: this section represents the signal that has the desired frequency spectrum that you want to match.
- Input: this section represents the signal that the equalizer is operating on that needs to be matched to the reference signal.

Hit the record button for the Reference or Input section to start capturing the frequency spectrum. The numerical display shows how many seconds were captured. Capture will automatically pause if an input is silent. As a guideline, try to capture a reasonably long section, with a duration of at least 10 seconds, preferably longer.

- The Reference spectrum can be captured from the main bus input, or from an external side chain input by enabling the ‘Ext’ button.
- The Input spectrum can be captured pre or post equalizer. Click on the ‘Post’ button to select learning post EQ.
- Reference and input spectrum can be saved to disk or loaded from disk to allow you to create your own library of spectrum captures.
- The Reference spectrum has an edit menu with a range of ‘ideal’ spectra with varying slopes as a starting point.



Captured signal spectra will be visualized as overlay to the equalizer. It is therefore recommended to disable the Spectrum Analyzer and Spectrogram features when working with the EQ Match feature.

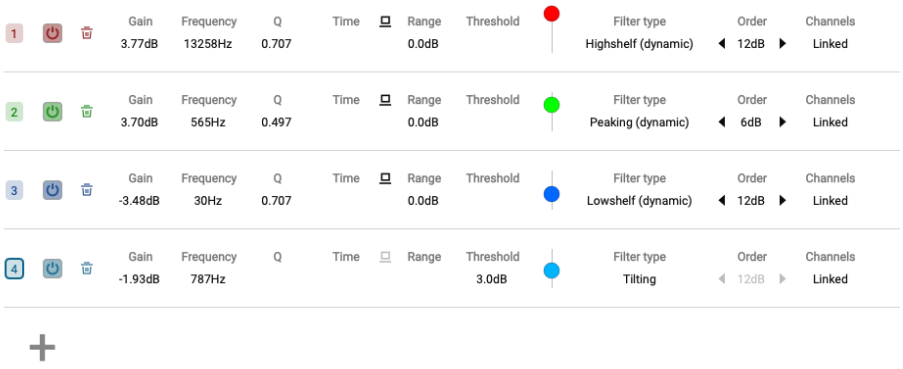
After capturing both Reference and Input data, the measured difference curve will be visible, and the ‘Amount’ control becomes available. Use the ‘Amount’ control to add or reduce the number of (new) filter matching sections to get a closer match to the measured difference curve. After dialing in a specific number of filter matching sections, you can freely modify these sections manually to adjust the effect.

4.10 Bypass

Use the bypass control in the bottom-right corner of the user interface to quickly bypass the equalizer. When in bypass mode, the user interface will dim.

4.11 Table view

Equalizer Pro can list all parameters of all filter sections in a table to get a quick overview of all settings. Enable the table view with the ‘Table’ button in the lower right corner of the user interface. The table view is fully editable in the same way as the graphical view. Click on the ‘plus’ button at the bottom of the table to instantiate a new filter section.

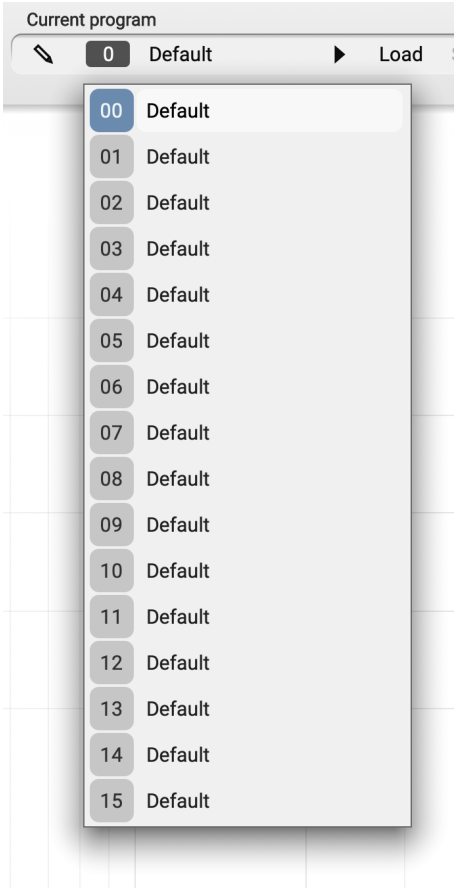


4.12 Program banks

Equalizer Pro has a program bank with 16 program slots, numbered 00 to 15. Each of these 16 program slots can contain a complete set of filters, parameters, spectrum analyzer configurations, program name, etcetera.

You can select a different program from the program bank in various ways:

- Use the 'Next' button right to the name of a program to advance to the next program in the program bank.
- Single-click on the name of a program to see a drop-down menu with all programs. You can select another program by clicking on any of the programs in the drop-down menu.
- Send a MIDI program message (on any MIDI channel) with program number 0 to 15 to select the corresponding program number.



Rename a program by double-clicking the name in the toolbar, or use the Name edit button that is shown left of the program slot index.

When a specific program slot is selected, you can edit that specific program independently from the other programs in the program bank. This includes loading and saving the program using the Program Manager by clicking on the Load or Save buttons.

Last but not least, you can import and export the complete program bank using the 'Import xml program bank' and 'Export xml program bank' options in the hamburger menu. These options allow you to load or save all programs in a bank in one xml file.

4.13 EUCON external controllers

Equalizer Pro supports the use of external controllers and hardware mixer integration using the EUCON protocol. EUCON is a high-speed Ethernet-based control protocol that enables external control surfaces to connect and communicate with host programs such as Avid Pro Tools and hosted AAX plugins. EUCON control surfaces that support the 'EQ' tables can modify frequency, gain, slope and Q values of Equalizer Pro remotely.

Equalizer Pro contains a dedicated 'EUCON EQ' preset that is recommended as starting point for EUCON control. This preset consists of 7 sections and includes a high-pass filter, a low-pass filter, a low-shelving filter, a high-shelving filter, and 3 peaking filters. All of these filters can be controlled from the compatible EUCON control surface. If the EUCON control surface supports visualization of the EQ curve, you should be able to see the (static) EQ curve on your control surface as well.

Consult the manual of your EUCON-compatible control surface, and Avid Pro Tools to set up EUCON control surface functionality. Equalizer Pro is also compatible with the EUCON control iPad app for iPadOS.