



Pulsar Massive

User Manual



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Introduction

This manual describes the features and operation of the Pulsar Massive effect processor. To be sure you understand how to use your plugin and understand all its subtleties, please read it completely.

The information contained in this manual is believed to be correct at the time of publication. However, if an error has unfortunately crept into its contents, please let us know.

IMPORTANT: The prolonged use of amplified instruments, speakers or headphones may cause permanent hearing loss. Ensure you monitor your exposure level, and take regular breaks. In case of tinnitus or suspected hearing loss, please consult an ENT specialist.

Welcome

Our experience

Thank you for choosing Pulsar Audio quality!

With more than 15 years' experience in plugin development for the biggest names in the industry, we decided to create Pulsar Audio to push the quality requirements of our products even further.

For each product, our quest for excellence requires us never to rest on our technical achievements, and to expand our knowledge ever further.

Sound and science

With solid expertise in audio signal processing, but also in electronics, sound techniques and music practice, we take great care in modeling all the small details and imperfections of analog equipment that make the difference between a « mathematical » exact sounding algorithm and a rich, living and musical processing, and we produce this famous « 3rd dimension » sound so much sought after.

In addition, our close collaboration with music production professionals requires us to be rigorous in order to produce professional quality tools.

Our user interfaces

The user interface of a plugin is the link between the creative drive and the technical implementation; it must therefore be clear, intuitive, and as pleasant as possible to use. We take great care to create the most beautiful and fluid interfaces possible, with an emphasis on intuitiveness.

The search for the right equipment

Rarely do you find two analog machines that sound exactly the same. It is therefore important, when developing an emulation, to carefully choose the hardware units to be used as models. We only use units in perfect condition and measure them with the best recording equipment.

A final word

We hope you will enjoy this plugin as much as we enjoyed creating it. Be sure to visit our website www.pulsar.audio and find out about updates, new products, tips and other resources. There you will also be able to contact us to ask for help or simply to tell us about your experience!

The Pulsar Team

Passive equalizers

History

Filters are among the oldest electronic audio effects, dating back to the beginnings of phonographic recording and reproduction. They were first used to correct spectral distortions in sound reproduction systems, and later for aesthetic purposes. The first filters were fixed, but a variable equalizer arrived in the 1920s (created by John Volkman for the RCA company), kicking off a spectacular evolution of these tools throughout the 20th century.

Before the democratization of the transistor, the first equalizers were made of passive networks followed by tube amplification cells. For cost reasons, several filters were implemented on the same passive components network, often in parallel. The Pultec EQP-1 equalizer, still widely used today in studios around the world, is probably the best representative of this technology and has achieved mythic status among audio engineers.

Pulsar Massive is an emulation of a famous passive equalizer released at the dawn of the 21st century, itself largely inspired by the Pultec EQP-1. Passive equalizers are characterized by a rich and smooth sound. The use of inductances in the passive circuit causes characteristic behaviors, particularly in the low end of the spectrum: the higher the input level, the greater the likelihood that harmonics and 'dynamic' behavior will appear. This behavior allows, for example, to work on the low mids (100-500Hz) without the sound becoming muddy as is often the case when using other EQs in this frequency zone.

With Pulsar Massive, you get:

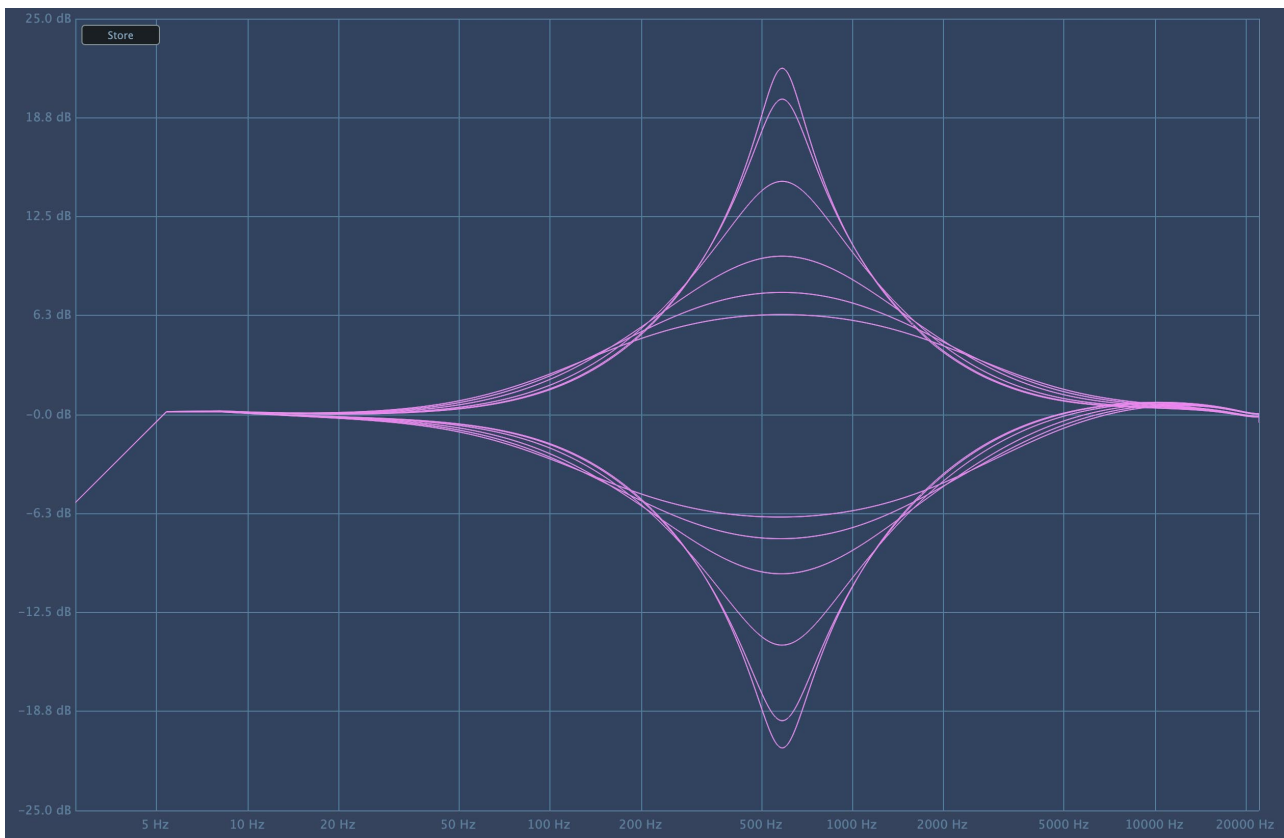
- The sound of a legendary EQ unit that's used in studios all over the world
- Musical equalization, either as an insert effect or on a bus
- Unique warmth and dynamic behavior (due to our emulation of the original circuit's inductors and tube amplifier stages)
- Very smooth and musical equalization curves
- An innovative workflow that allows you to tweak equalization directly on a graphical curve
- An organic analog sound paired with the advantages of digital (presets, automation of parameters, etc.)

Pulsar Massive's filtering forms

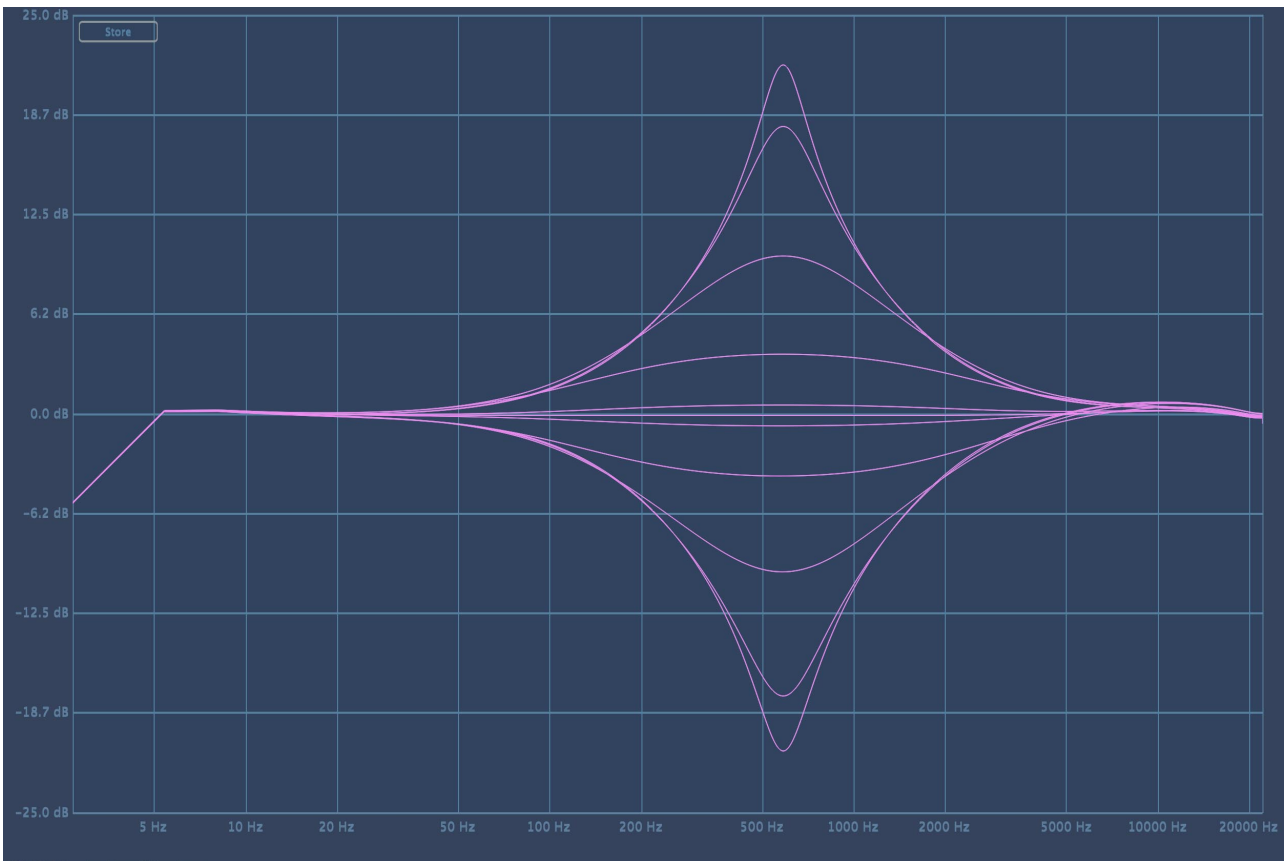
Bell filters

Pulsar Massive's bell filters differ from the classic constant-Q bell filters found in most of today's equalizers, in that the Bandwidth (ie, Q) also has an effect over gain:

- With a fixed Gain amount, adjusting the bandwidth setting will vary the effective gain of the filter. In other words, the actual gain amount of the bell filter can end up being different to the value shown on the Gain knob.
- When a band is modified from the curve, the vertical movements of the mouse are used to define the gain of the band. However, when the Gain parameter thus modified reaches its maximum (or minimum), and you continue to move the mouse vertically, it is the Bandwidth of the filter that is modified in order to achieve higher effective gain values.
- With a fixed Bandwidth amount, the Gain knob will not affect the width of a filter.



Bell filter with fixed Gain (20 dB) and different Bandwidth values (from 0 to 10)



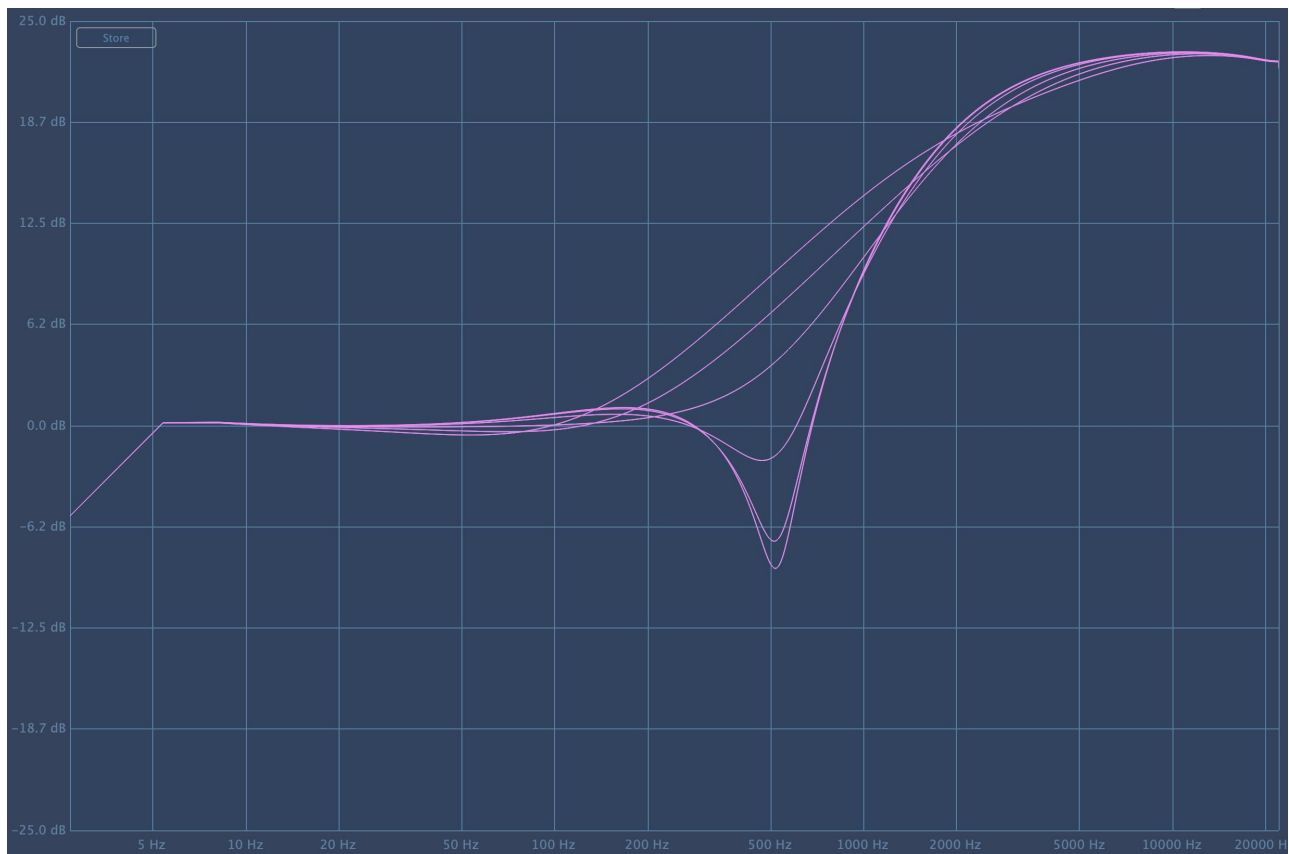
Fixed Bandwidth bell filter (Set at the maximum value of 10) with different Gain values (from 0 to 20 dB)

Note: Our modeling, accurate down to the level of the individual electronic component, faithfully reproduces all aspects of the original machine, including the differences between the frequencies displayed on the knobs (and which in the plug-in are also displayed in various other places), and the actual frequencies of the filters, which can sometimes be significant. For example, at the 82 Hz position, band 2 actually produces a resonant filter close to 100 Hz. We have chosen to keep these differences, which thanks to the curve display can be surprising, because they are part of the unique workflow of the original device that we want to reproduce in its digital emulation.

Shelving filters

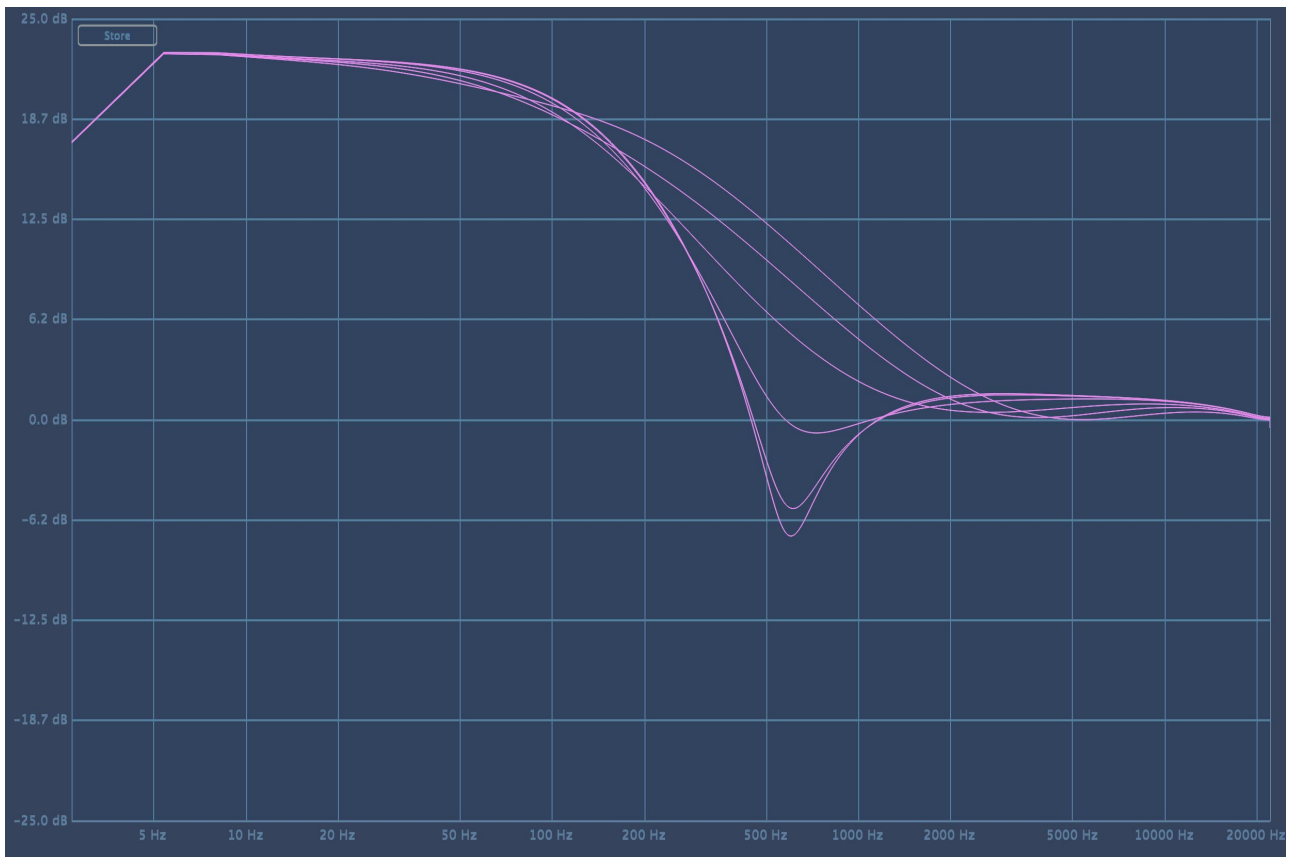
Pulsar Massive's high and low-shelf filters are also unusual when compared to those present in modern equalizers. With increasing bandwidth, a shelf will introduce a complementary bell filter, at the other side of the corner frequency – a scoop before a boosted shelf, or a bump before an attenuated shelf.

This type of filter allows, for example, in the case of a high-shelf, to add power in the very high end of the spectrum (frequencies above 8-10 kHz), while keeping harsh frequencies (between 2 kHz and 4 kHz) unchanged.



High-Shelf filter with fixed gain (20 dB) and different bandwidth values (from 0 to 10)

Note: This setting can be misleading. For example, when the Bandwidth parameter is set to maximum and you try to boost the high frequencies above 8.2 kHz, the shelf filter will also attenuate the frequency area around 8 kHz due to the added bell filter in the opposite direction. For this reason, you will often be surprised at your own frequency choices when using the Massive's shelf filters! The best frequency choices will often be higher than you would expect for low-shelves, and lower for high-shelves. The Pulsar display rack provides graphical feedback on the proposed EQ, and allows the user to better understand the filtering done by the plugin.



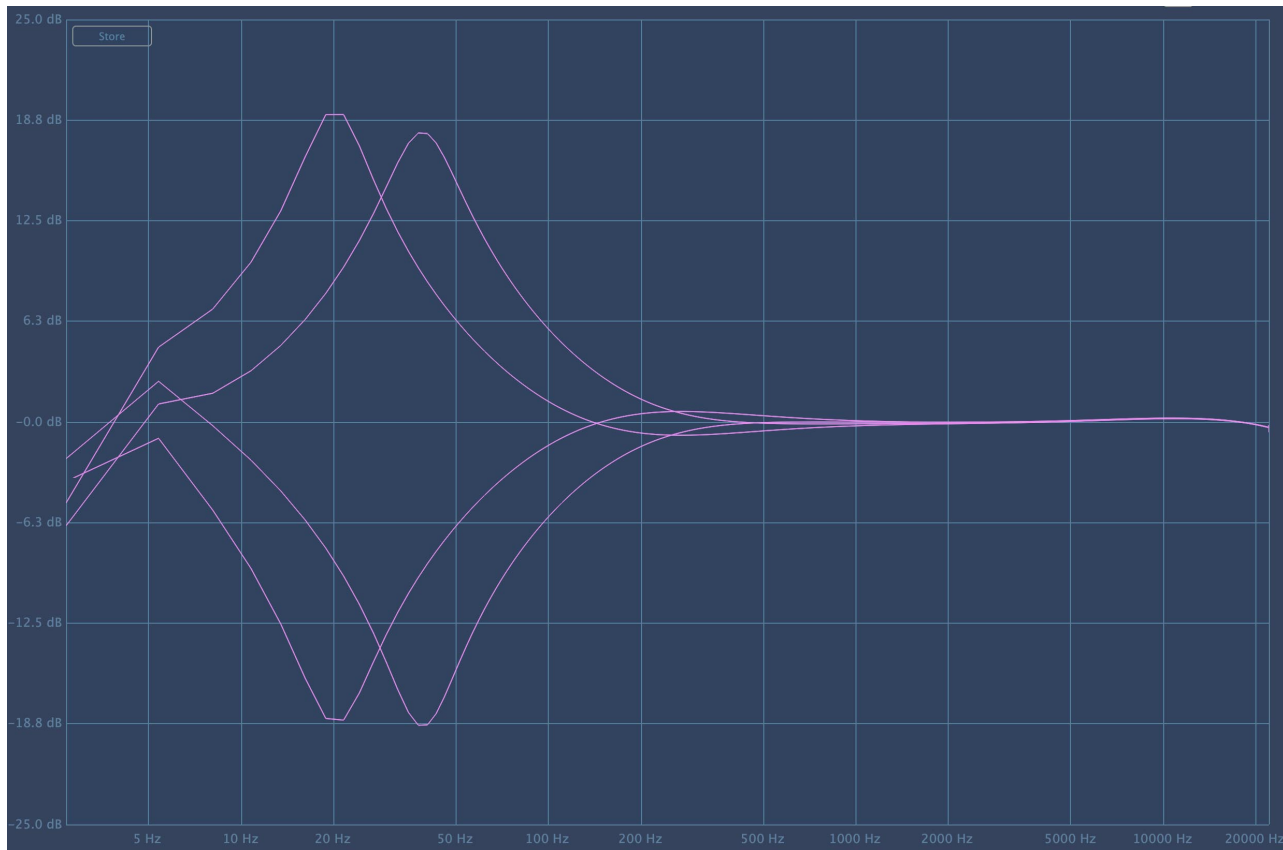
Low-Shelf filter with fixed gain (20 dB) and different bandwidth values (from 0 to 10).

Special modes

When using bands 1 and 4 of each channel in Pulsar Massive's Shelf mode, you'll notice that the frequencies 22 Hz and 33 Hz on the first band as well as 16 kHz and 27 kHz on band 4 change color and become orange. This illustrates that the behavior of the filters in Shelf mode is particular to these frequencies.

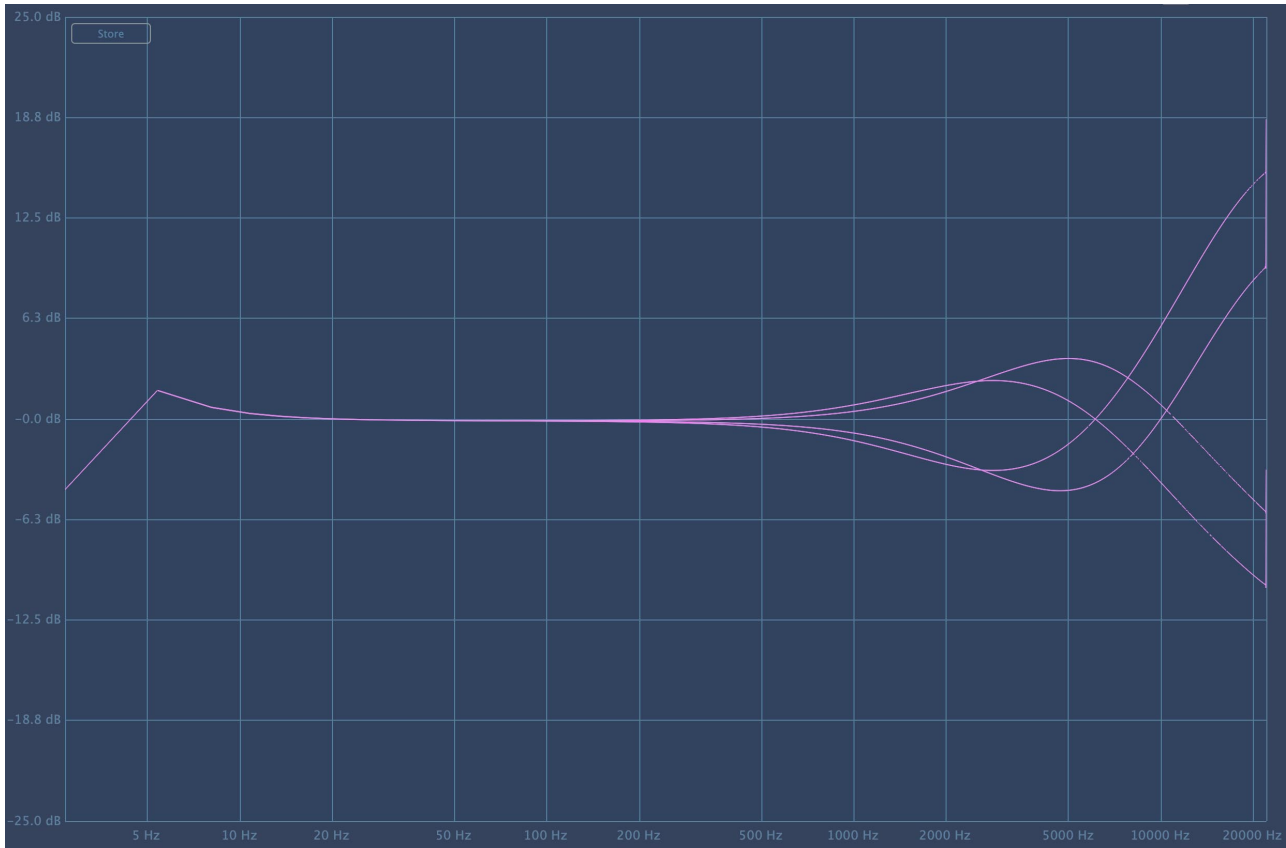
Indeed, as the frequencies approach the limits of human perception, the shape of the filter is reworked so that we don't amplify frequencies too far from the usual 20Hz-20kHz range. These new shapes (described in the screenshots below) no longer have the classic shape of Pultec filters, instead providing you with a unique way of working with the extreme frequencies of the audio spectrum:

The 22 Hz and 33 Hz low-shelf filters are quite similar to a bell filter. The subsonic frequencies are filtered in such a way that they do not overload the low end of the spectrum (which could be dangerous for the speakers, or affect dynamic processing carried out after Pulsar Massive in an undesired way).



Special 22 Hz and 33 Hz low-shelf filter modes with minimum bandwidth.

In the 16 kHz and 27 kHz high-shelf filters, the inverse bell filter added by the Bandwidth control has a higher gain than in normal modes and is located for both around 8 kHz. This enables you to obtain optimal control between the very high end of the spectrum and the high-medium frequencies where you might experience sibilance.



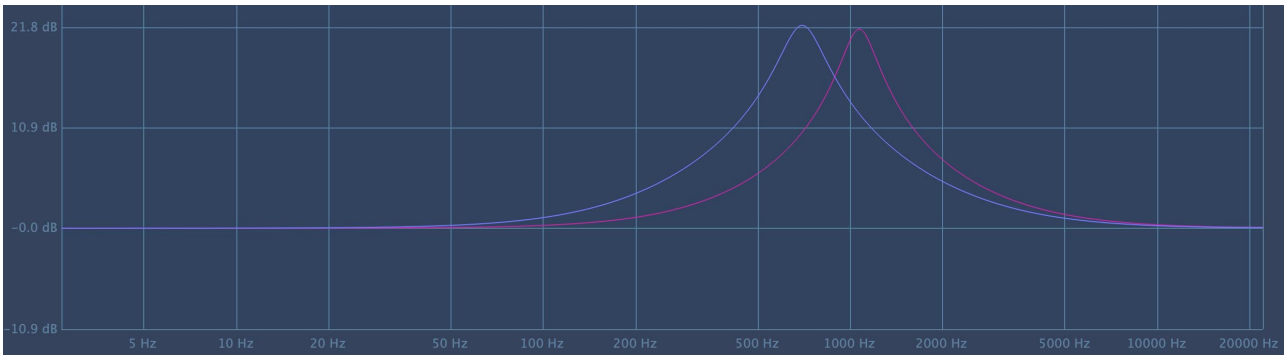
Special 16 kHz and 27 kHz high-shelf filter modes with Bandwidth set to minimum

Parallel equalization

The original machine that Pulsar Massive emulates is a parallel equalizer. If we were to take a quick look at the electronic structure of the unit, we would see that:

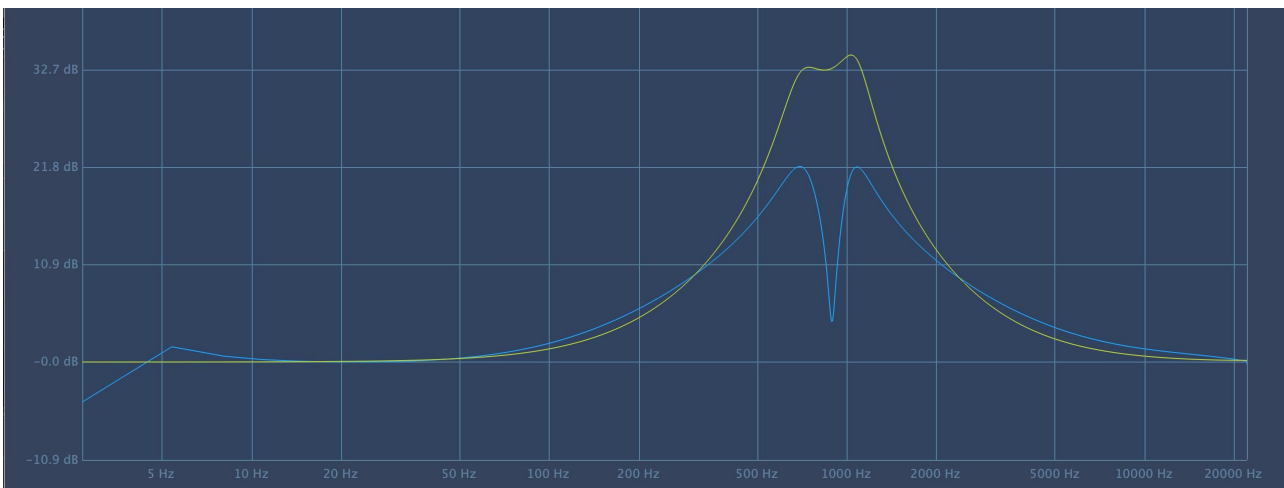
- The overall structure consists of two passive filtering networks (Boost and Cut) in series (processed one after the other), followed by two amplification circuits: one for positive gain filters and one for negative gain filters.
- Within each of the Boost and Cut structures, the filter blocks corresponding to the 4 bands of the EQ are wired in parallel, which essentially means that they all filter the same input signal, and the filtered signals are then summed. By contrast, in most other modern equalizers, the filters are successively applied to the signal and they are said to be in series.

Here is a quick example of parallel equalization. Let's take two bell filters of about 20 dB of gain, placed at about 600 Hz and 1 kHz. You can see the frequency responses below:



Two bell filters at close frequencies.

Now, take a look at the combination of these two bell filters performed with a series equalizer (in yellow), and with a parallel equalizer (in blue). You can see that with filters of 20dB maximum gain, the series equalizer gives a response that can reach up to 40dB. With a parallel equalizer, on the other hand, the response never exceeds 20 dB in total.



Difference between the combination of bell-shaped filters in series and in parallel.

Quick start

Installation

Pulsar Massive is available as a plugin in VST2, VST3, AU and AAX formats for use with all major DAW software such as Live, Cubase, Logic, Pro Tools, etc.

Installation from the supplied installer is automatic. The installer takes care of copying the different plugins as well as presets, manual, etc. into the appropriate locations.

Note: If you are using the VST2 format in Windows, you will be asked by the installer to specify the installation folders for the 32-bit and 64-bit VST2 plugins respectively. The paths that seem most appropriate for your computer will be recommended by default, but we advise you to check them before completing the installation. If the plugin is not installed in the same folder as your other possible plugins, your DAW software may not detect it.

Activation

All our plugins are protected by PACE's iLok system. For correct operation, we recommend you ensure that you have the latest version of the « iLok License Manager » software, available for free download at www.ilok.com .

You can choose between three activation methods:

- Activation on a hardware USB dongle such as iLok 2 or iLok 3, which will enable you to use your plugin on several machines (you can order a dongle online at www.ilok.com or buy it from your music retailer)
- iLok Cloud activation which will enable you to use your plugin on several machines but requires a permanent internet connection
- Machine activation, which does not require a dongle or a permanent internet connection, but only activates your plugin on one machine

Important: If you choose the iLok Cloud system, you have to open a Cloud session on your computer by going to the « File > Open Cloud Session » menu of your iLok License Manager. If you choose an iLok 2 or 3 dongle, you have to connect it to your computer before any operation.

When you purchase your software, you will receive:

- Either a license deposited directly onto your iLok account. Just go to the « Available » tab and drag it to the destination of your choice (here CLOUD for a cloud license, or iLok_Pulsar for an iLok 2 or 3 dongle)
- Or an activation code. Simply paste it into the « Licenses > Redeem Activation Code » menu to receive the license on your account, and drop it off at the destination of your choice (CLOUD or iLok 2 or 3 dongle)

iLok License Manager

All Licenses (126) Available (5) All Activations (1) Unavailable (72) Hidden (0)

pulsar
126 Licenses

Local

- CLOUD
35 Activations
- MacBook Pro de
0 Activations
- iLok_Pulsar
42 Activations

Valid Locations	Product Name	Publisher Name	Subtype	Expiration Date	Deposit Date	Type	Activ
	Pro Tools	Avid	Product	04/01/2019 19:59	04/01/2018 19:59	Subscription	0 of 1

ZERODOWNTIME INFORMATION

Export CSV

Show Details

iLok License Manager's "available" license tab

First steps

Load Pulsar Massive on a track of your choice in your DAW. A good starting point is to load a basic preset corresponding to the type of track (voice, guitar, bass, drums...). From there:

- Start playback and compare how the material sounds with and without Pulsar Massive, using the bypass button on the interface (the Power switch). You can activate the Auto-gain mode to compare at equivalent volumes.
- Adjust the gains of the different bands according to the desired result (the bands are sometimes named according to their role)
- Adjust the calibration (Drive) to make sure there is no audible saturation while enjoying the warm and dynamic sound of the Pulsar Massive

You can browse through the many available factory presets to find inspiration quickly, without getting into technical considerations.

The user interface



The user interface

The user interface consists of two distinct panels:

- The toolbar, common to all Pulsar Audio plug-ins (top)
- The control panel, specific to the plug-in. This panel is made up of two racks: the EQ control rack on the bottom, and the curve display and metering rack on the top

You can find a resizing control in the bottom right corner of the plugin interface. This is common to all Pulsar Audio plug-ins, and scales the interface to the exact size you want, from 67% to 200%, in increments of 1%.

Use of parameter controls

The parameter control knobs have several modes of use:

- The normal editing mode (use a classic mouse drag, or the mouse wheel)
- The fine editing mode (hold the Ctrl or Cmd key while dragging or while using the mouse wheel, or drag with the right mouse button)
- The « reset to default » action (double-click, or click while holding the Alt key)
- The « menu » action (right-click, or click while holding the Ctrl key)
- Only for some controls, the alternate edition mode (hold Shift while dragging), which can have various functions, for example to temporarily link two parameters

Parameter locking

It is possible to lock certain parameters, so that they are not changed when loading a preset. For example, one possible use of this feature is to set the input and output gains of a compressor to achieve the desired amount of gain reduction, lock these parameters, and then scroll through the list of factory presets to find the most appropriate tone.



Locking the Dual Input knob

To lock a control, right-click it with the mouse, or click while holding down the Ctrl key on the keyboard. If the control can be locked, a menu will appear offering to lock it. When a parameter is locked, a small padlock icon appears next to the control.

Using the GUI resize control

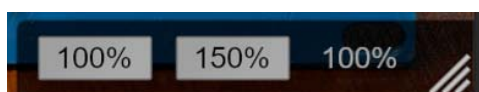
Located at the bottom right of the interface of all Pulsar Audio plugins, this control allows you to resize the plugin's interface to your liking. It comes in the form of three lines, like a classic resizing handle:



Resizing handle

Note that in some DAWs, this resizing can be problematic, depending on how the DAW developer has designed its windowing.

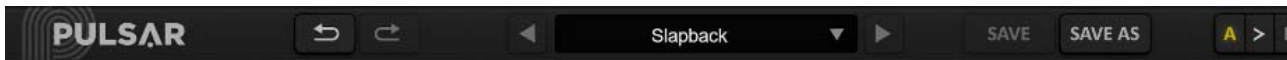
It is also possible, by clicking in the corner, to open a small popup window with buttons offering a choice of fixed size resizing (100% - 150%):



Resizing window

The Toolbar

Located at the top of the plugin interface, it contains all the functions relating to parameters, presets, communication with Pulsar Audio, etc.



The toolbar

Undo / Redo

The two arrow buttons on the left of the toolbar have the function Undo and Redo, i.e. respectively the cancellation and restoration of the last action. All parameter changes and more generally the state of the plugin are stored in a history. You can click on « Undo » at any time to return to the previous state (or to the nth previous state) and on « Redo » to return to the current state.

Note: a right-click on one of these buttons gives access to the list of stored operations.

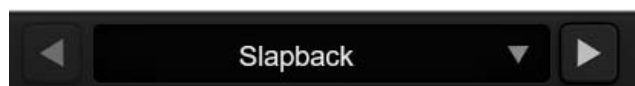


Undo / redo buttons

Preset Selection

The preset selection area, located in the center of the bar, allows you to:

- Read the name of the current preset. If an asterisk appears after the preset name, it means that the state of the plugin no longer matches the saved preset
- Select a preset from the list of available presets, arranged in sub-banks
- Delete the current preset (« Delete Preset » option)
- Rename or move a preset to another sub-bank (« Move / Rename Preset » option)
- Set the current preset as the one that will be loaded by default when creating a new instance of the plugin (« Set This Preset As Default » option)
- Open the presets directory. This can be handy for making backups of your preset files and restoring them. Note that renaming and reorganizing presets must be done from the plugin menu, not by using your system's file explorer.
- Restore factory presets. This will also overwrite any changes you have made to your factory presets
- Quickly navigate between the presets to find inspiration, using the left and right arrows



The preset selection area

Save / Save As

The Save button saves the current preset.

The Save As button saves the current state of the plugin under a new preset name.



Save and Save As buttons

A / B

This section allows you to compare 2 different states of the plugin, or 2 different presets. Slots A and B, accessible through these 2 buttons, represent 2 completely independent states.

For example, when state A is active, you can load a preset and/or make settings from the interface, then click on button B; then load another preset and/or make other settings; buttons A and B now allow you to quickly switch between the two states and easily compare the 2 presets or sets of settings.

It is also possible to copy the state A to B or vice versa using the > or < buttons located between A and B.



A, B and Copy buttons

Rack visibility button

This button allows you to hide some parts of the plugin interface. Using the rack visibility button, you can toggle between displaying both sections of the interface, displaying only the bottom rack (without curve editing and visualization), and displaying only the top rack.



The rack visibility button

Menu Button

The button located on the far right of the bar encompasses various options.



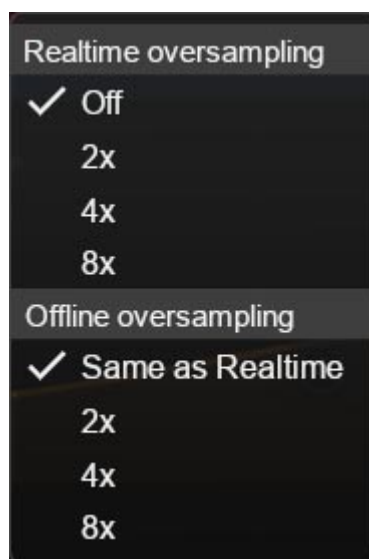
The Menu button

Oversampling settings

The first menu item is used to set the oversampling. Oversampling allows the sound to be processed at a higher sampling rate within the plugin, in return for higher latency and CPU consumption. Oversampling is disabled by default, as all Pulsar Audio products use advanced technologies that allow in most cases to process the sound without oversampling, with no compromise on quality. This makes oversampling useful mainly when you saturate a lot.

The maximum available oversampling rate is not the same in all Pulsar Audio plug-ins and depends on a trade-off between the need for oversampling and the CPU consumption induced by oversampling in this plug-in.

Please note that Pulsar Audio products use very high-quality linear phase upsampling and downsampling filters. This means that the x2 oversampling will generally be of higher quality than the x2 setting in a competitor's product, but will also be more CPU intensive.



Oversampling options

The "Offline oversampling" option allows you to choose an oversampling setting for final rendering (and other non-real-time processing) independent of the setting applied in real time. This enables to reduce the CPU consumption during the use of the plugin, while having the best quality during the final rendering.

Stepped Gains/Freqs/Bandwidths

Pulsar Massive's Frequency selection controls are continuous – you can target exact frequency and gain values of your choice and can even type them into the parameter value display fields. Certain studio hardware units, including the original, and especially units designed for mastering, have a fixed selection of centre/corner frequencies and gains that can be selected.

With Pulsar Massive, you can re-enable these discrete steps and choose their resolution.

For the gain knobs, you have the choice between continuous operation or 0.5dB / 1dB / 2dB / 4dB steps.

For the frequency knobs (including the LP/HP knobs), you can choose between the following options:

- No (default): The frequency knobs are continuous and all intermediate frequencies can be accessed
- Stepped: The frequency knobs are stepped like in the original hardware unit
- Half Steps: The frequency knobs are stepped like in the original unit with additional intermediate steps. For bands 1-4, the added frequency values correspond to the frequencies of the neighbouring bands.

For bandwidth controls, the plugin offers the following options:

- No (default): The bandwidth knobs are continuous and all intermediate values can be accessed.
- Stepped: The bandwidth knobs are stepped, 11 values can be accessed.
- Stepped: The bandwidth knobs are stepped, 21 values can be accessed.

Note: when controls are stepped, it is still possible to access intermediate values by using the fine editing mode when using the control. Just down the Ctrl key while dragging, or use the right mouse button to drag.

Other options

Other functions accessible through this menu are:

- Enabling / disabling the help balloons
- Access to the website
- Access to social media
- Access to communication with technical support
- Link to this user manual

The control panel



The Pulsar Massive control panel

Pulsar Massive's control panel is inspired by the famous hardware unit. For the plugin, some liberties have been taken to increase the sonic possibilities and to allow better navigation and user experience.

This panel consists of two racks:

- The EQ control rack
- The curve display rack

The control rack

This rack contains the main controls (switches and knobs) useful for setting up the equalizer. These correspond to the controls present on the original machine.

It is organized in two identical and symmetrical sections, left and right, which correspond to the settings of two channels in stereo (in mono, the right section is not used), as well as a central section with the controls common to both (Power, Auto-gain, Transformer, LR/MS, Link).



The control rack takes up the bottom half of the Pulsar Massive interface.

On each side, there are four almost identical vertical slices, representing the settings of each of the four equalization filters (In/Out, Bell/Shelf, Gain, Bandwidth (Q), Frequency).



Control strip of a band

Power

The switch in the center of the interface allows you to activate and deactivate the effect. You can use this as a bypass button within the plugin itself, rather than relying on your DAW's bypass functionality.

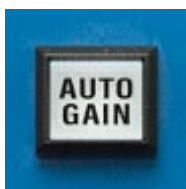


Power button

Auto Gain

With this setting activated, a compensatory gain will be applied, based on the settings of the various bands. This helps you keep a constant sound level while adjusting the EQ controls, helping you to judge your changes without needing to remember that "louder sounds better".

Note: the algorithm used does not perform actual real-time level measurements – it derives an estimate based on EQ settings only. This enables it to work on a wide variety of musical sources. Depending on the source, you will sometimes still need to make small adjustments with the Output Gain knob.



Auto Gain button

Transformer

This selector will give you the choice between several output transformer options:

- "OFF" mode: No output transformer in this mode, but the output amplifier circuit remains present
- "Transformer 1" mode: faithful emulation of the transformer and the output circuit of the original machine
- "Transformer 2" mode: AC output transformer, with a modified output circuit, in order to slightly accentuate the dynamic characteristics of the machine while gaining in harmonic richness



Transformer selector

L/R M/S

This switch selects the stereo management mode, and what signal each bank of four filters is actually processing.

- L/R corresponds to the traditional stereo mode (left / right channels).
- M/S stands for Mid / Side mode. In this mode, the two channels of the equalizer contain respectively the sound common between both left and right channels (Mid) and the difference between left and right channels (Side).

The M/S mode allows you to employ different equalization curves for material in the very center of a signal and the elements panned more to the sides. It is therefore recommended to disable the "Link" switch when working in M/S.



L-R / M-S switch

Link

This switch links the settings of the two banks of filters. If Link is activated, every setting changed in the left bank is also carried over to the right bank (and vice versa).

Note: If a parameter is set differently between the two channels, and then the Link switch is activated, the relative distance between the two channels is maintained. To reset the parameters of the two channels to the same settings, it is therefore necessary to disable Link beforehand.



Link switch

Listen

These headphone-shaped buttons allow you to solo a single channel and hear only the signal passing through the left or right filter bank. This parameter is particularly useful in M/S mode to fine-tune the equalization of each of these channels.



Listen switch

Channel bypass

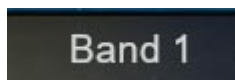
Bypass processing in either channel (left or right, mid or side) by clicking the relevant light. When the plugin is placed on a mono channel, these lights will not function.



Channel L (left channel) bypass switch

Band name field

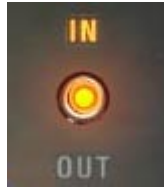
An editable text field above each band lets you name it. The default names are Low, Warmth, Presence and Air.



Editable band name field

Band bypass

Updating the original unit's Boost/Cut/Out switch, on Pulsar Massive this switch simply activates or deactivated the band.



Band bypass switch

Band filter type

This selector allows you to choose the type of filter for each band:

- Bell : selects a classic bell filter
- Shelf : selects a shelving filter (low shelf for bands 1 and 2, and high shelf for bands 3 and 4)



Band filter type switch

Gain knob

This knob adjusts the gain of the filter:

- Bell: as explained earlier in this manual, the actual gain will also depend on the Bandwidth setting. The actual gain value is very often quite different from the gain displayed by the knob.
- Shelf: the actual gain is quite close to the gain indicated on the silkscreen.



Gain knob

Frequency

This knob allows you to adjust the frequency of the filter.

- Bell: the adjusted frequency will be the center frequency of the filter (at the top).
- Shelf : the adjusted frequency will be the corner frequency of the filter, halfway to the filter's maximum gain output.

Note: it is possible that the actual frequency is slightly different from the frequency displayed. Indeed, on the original machine, small errors in the component values generate slight biases.



Frequency adjustment knob

Bandwidth

This knob has an action depending on the type of filter used:

- **Bell:** The higher this setting, the narrower the bandwidth. As said before, this setting also has an influence on the effective gain of the filter. The highest values of gain (+20dB) are reached when the Bandwidth parameter is set to its maximum (narrowest).
- **Shelf:** As explained at the beginning of this manual, this mode recreates the classic shape of the shelves inherited from Pultec EQs. With the bandwidth set to minimum, the filter behaves like a classic shelving filter. The more you increase the bandwidth, the more you deepen the effect of the reciprocal bell close to the shelving filter. The curves of these filters are shown in the section "Pulsar Massive's filtering forms".



Bandwidth adjustment knob

HPF

This knob allows you to activate the high-pass filter and to adjust its frequency.



HPF frequency knob

LPF

This knob allows you to activate the low-pass filter and to adjust its frequency.



LPF frequency knob

Drive

This knob sets the global calibration of the plugin. The higher the calibration, the more the circuits behave in a non-linear way, revealing all their character. However, you have to be careful, because a too high setting can induce saturation (but this can also be the desired effect).

The non-linear behavior of the circuits brings a particular sonic thickness (the inductors, the tube amplification stages, and the output transformer generating harmonics), but also a dynamic behavior of the bands in the low end of the spectrum. This behavior allows the machine to keep a clean sound at boost levels much higher than a classic EQ.

In some cases, the user may wish to use the frequency response of the parallel passive equalizer without the non-linear behavior of the machine. For this, the calibration can be set to the minimum.



Drive knob

Output Gain

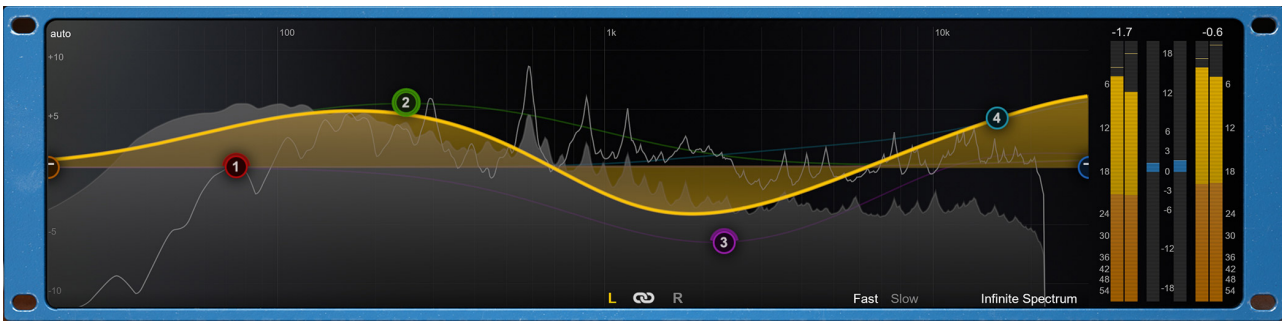
In the original machine, this knob is used to adjust the output gain, and thus possibly compensate for the gain lost or gained after filtering.

However, this gain is applied before the output block, which includes an amplifier stage and a transformer. A large value of this output gain may therefore lead to the generation of harmonics, or even saturation for extreme settings.



Output Gain knob

The curve display rack



Curve display rack

The top half of Pulsar Massive window is taken up by the curve display rack. This section allows you to:

- View and edit the frequency response of the equalizer channels.
- Visualize the spectrum of the output signal on several time scales.
- Visualize the input and output levels, as well as the level difference between input and output.

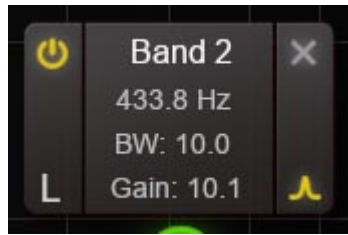
Equalization using the curve display

Here you can see the frequency response curve of the channel selected at the bottom of this rack (L or R, M or S), as well as the frequency responses of the individual bands.

- The L and R buttons (or M and S in M/S mode) select which channel is being edited. It is also possible to use the S button (as in "Switch") or the left and right arrows on the keyboard to control the active channel.
- The button in the shape of a chain link is used to activate or deactivate the link between channels. It is a repeat of the setting at the bottom of the hardware rack. It is also possible to use the L key (as in "Link") or the down arrow on the keyboard to activate/deactivate the link.
- The selector at the top left of the screen, which by default displays "auto", allows you to choose the range of the gain axis for the display of the EQ curve. The default behavior is to automatically switch between a range of +/- 12 dB and a range of +/- 24 dB, depending on the EQ settings, but the selector allows you to choose a more precise display range if necessary.

Different actions can be performed with the mouse:

- In the same way as for all other controls, hold down the **Ctrl** key or use the **right mouse button** to make fine adjustments.
- A **right button click** or a click with the **Ctrl** key pressed on a band allows to activate or deactivate the band.
- Hold down the **Shift** key on your keyboard while moving a band to activate the Band Solo function, which allows you to listen to the effect of that band alone on the input signal.
- Use the mouse wheel when hovering your cursor over a band to change its bandwidth.



Parameter window of an EQ band

The band parameters window appears when a band is hovered over with the mouse cursor.

- The button at the top left allows you to activate or deactivate the band.
- The letter at the bottom left indicates the channel being edited.
- The cross at the top right allows you to reset the band parameters to their default values.
- The button at the bottom right allows you to change the type of filter between bell and shelf.
- It's possible to click on any central field (name, frequency, gain, bandwidth) to enter values manually

Spectrum visualization

The spectral content of the signal processed by the machine is displayed behind the frequency response curve. There are a few options to refine the display:

- Fast and Slow modes determine the analysis window size, leading to faster or slower visual feedback.
- Infinite Spectrum allows the user to view the integrated spectrum from the beginning of the playback. This will appear as grey-shaded information behind the real-time white lines. If this mode is activated, you can click on the screen to reset the infinite spectrum calculation.

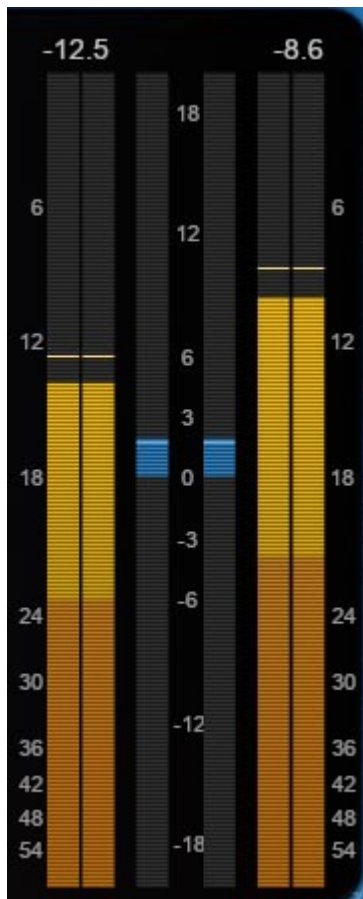
The signal displayed is not exactly the one at the output of the plug-in, but rather the one at the input to which we add the effect of the EQ. When there is not too much saturation, it is the same, but with the advantage that the effect of changes in EQ settings is immediately visible in Slow or Infinite Spectrum mode.

Level visualization

The vertical meters on the right of the interface are used to measure the input and output levels of each channel (left/right, or mid/side in M/S mode). There is a peak level reading (yellow); an RMS average level reading (orange), a "Peak Hold" bar that holds the highest peak level for a few seconds (thin yellow bars), and a reading of the "Peak Hold" level in dBFS above each meter.

In addition, a red light comes on when the peak level exceeds 0dBFS, and remains on until it is manually turned off by clicking on the indicators.

The central indicator (blue) displays the difference in RMS level between the input and output of the equalizer, to provide assistance in adjusting the output level. It should be noted that the human ear does not have a uniform sensitivity curve, so even with a level close to 0 dB displayed, there may be a slight difference in perceived volume.



Level visualization section

Minimum Configuration

This plugin is compatible with all major sequencers on the market (Cubase, Nuendo, Pro Tools, Logic Pro, FL Studio, Ableton Live, Bitwig, Digital Performer, Studio One, Reaper, Adobe Audition...)

Available formats:

- VST 2.4 (Windows: 32/64-bit, Mac: 64-bit only)
- VST 3 (Windows: 32/64-bit, Mac: 64-bit only)
- AAX (Windows: 32/64-bit, Mac: 64-bit only)
- Audio Unit (64-bit).



Windows

- CPU: Intel Core i3 / i5 / i7 / Xeon
- Memory: 4 GB RAM / 1 GB free disk space
- Operating system: Windows 7 and higher
- GPU: OpenGL 2.0 compatible GPU with up-to-date drivers
- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

MacOS

- CPU: Intel Core i3 / i5 / i7 / Xeon / Apple Silicon (M1)
- Memory: 4 GB RAM / 1 GB free disk space
- Operating system: 10.9 and higher
- GPU: OpenGL 2.0 compatible GPU with up-to-date drivers
- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

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