Pulsar 8200

User Manual

www.pulsar.audio

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Introduction

This manual describes the features and operation of the Pulsar 8200 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

Pulsar W495

History

The Pulsar W495 plugin is directly inspired by the Neumann W495b equalizer, a "Danner Cassette" format equalizer (an ancestor of the rack 500!), used in Neumann broadcast consoles and Neumann VMS vinyl presses/mastering consoles (it is said that 90% of vinyl records produced between the 1970s and 1990s were mastered with these Neumann modules!)

In the late 1960s, several inventors, including Daniel N. Flickinger, George Massenburg and Burgess McNeal, worked on equalization circuits that offered a constant and adjustable Q, and George Massenburg eventually expounded the concept of the parametric equalizer in a scientific paper published BY the Audio Engineering Society in 1972.

With the creation of GML, Inc. and the release of the GML 8200 equalizer, George Massenburg did not create the first parametric equalizer on the market, but one of the most mythical, remarkable for its ergonomics as well as for the transparency of its electronics, which makes it an exceptional machine still present today in many prestigious studios.

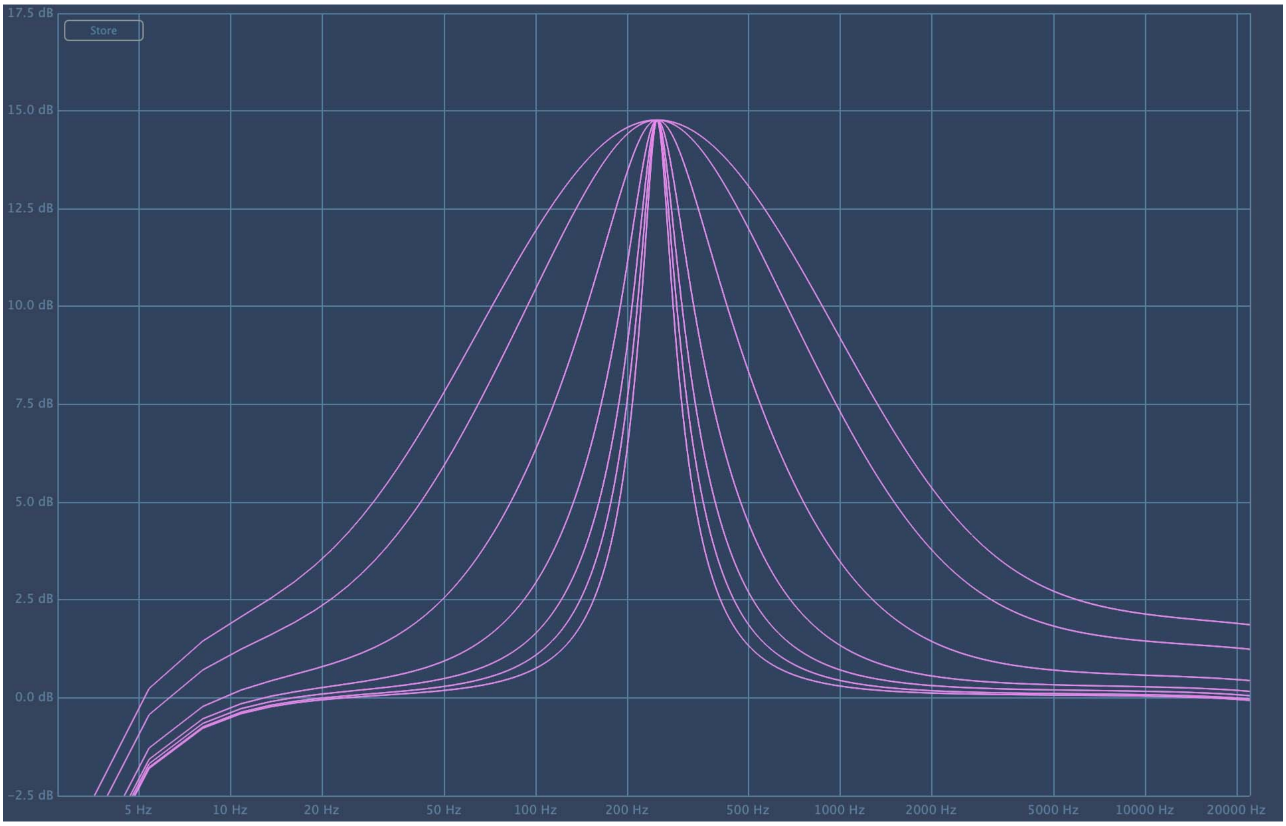
The Pulsar 8200 will offer you:

- The sound of a mythical machine, present in studios all over the world
- Musical equalization, both on an instrumental track and on a bus
- Very smooth and musical equalization patterns, also capable of very precise corrections
- Editing on the curve for fast workflow
- Additional bands for more flexibility
- A simple and powerful de-esser to attenuate whistling and other high frequency transients
- The advantages of analogue sound together with the advantages of digital (presets, parameter automation, etc.)

Filter forms of the Pulsar 8200

Bells filters

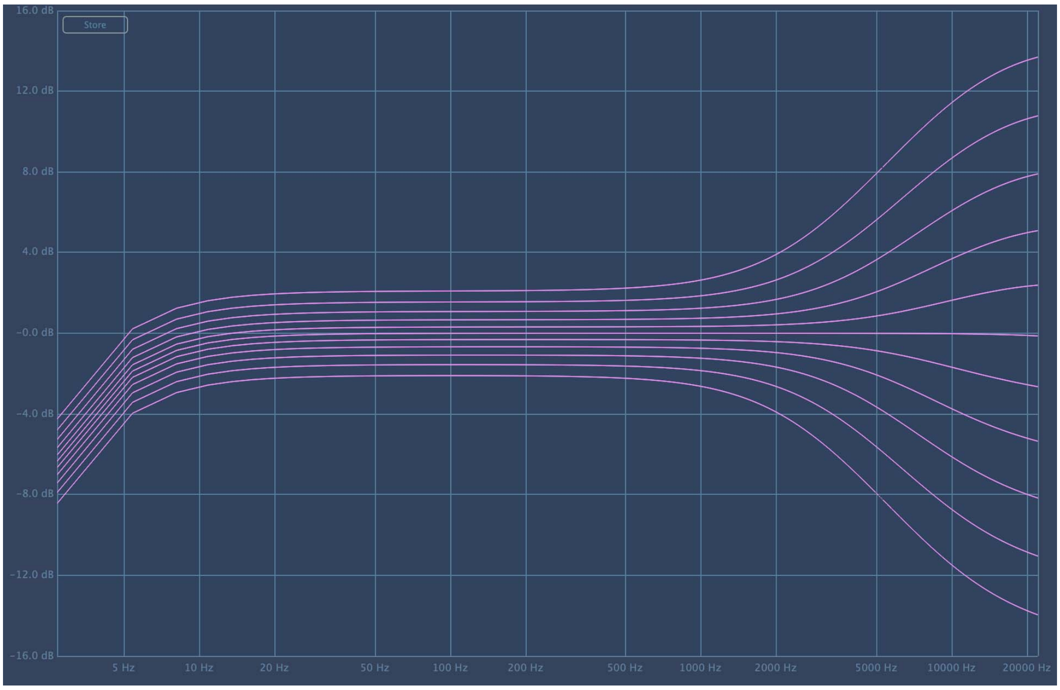
The bell filters in Pulsar 8200 are quite similar to the classic constant-Q bell filters found in current equalizers. The only difference is that in the 8200, in addition to filtering, an overall gain is added which depends on the Gain and Q of the filters used (the higher the Gain and the lower the Q, the more overall gain is added).



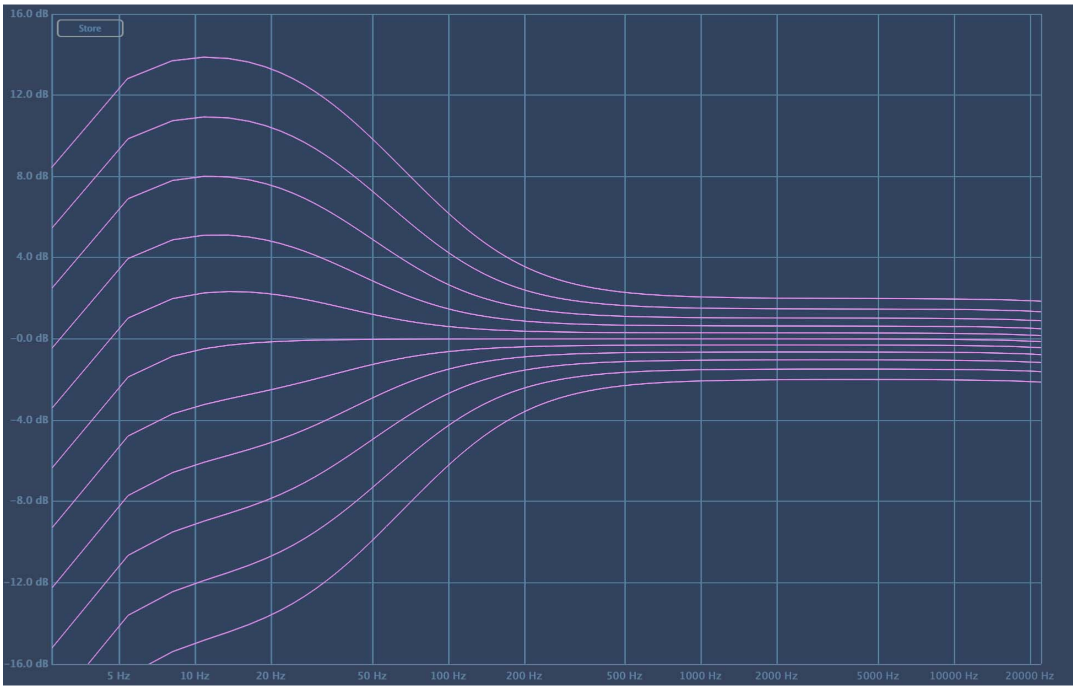
Fixed gain bell filter (15 dB) with different bandwidth values (from 0.4 to 4)

Shelf filters

The high-shelf and low-shelf filters in bands 1 and 5 of the Pulsar 8200 are very similar to those found in "modern" equalizers, except for the addition of overall gain, just as in the bell filters.



High-shelf filter with different gain values (from -15 dB to 15 dB)

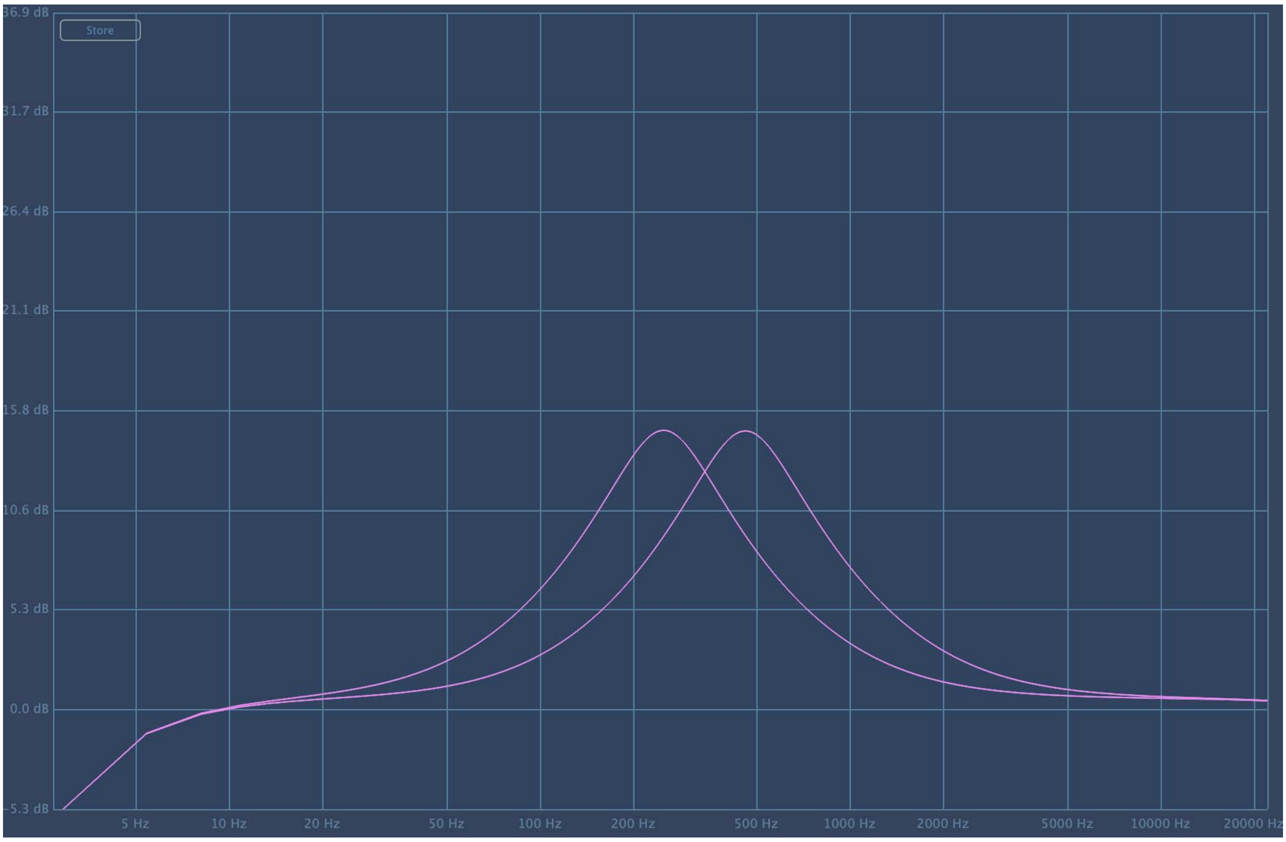


Low-shelf filter with different gain values (from -15 dB to 15 dB))

Parallel equalisation

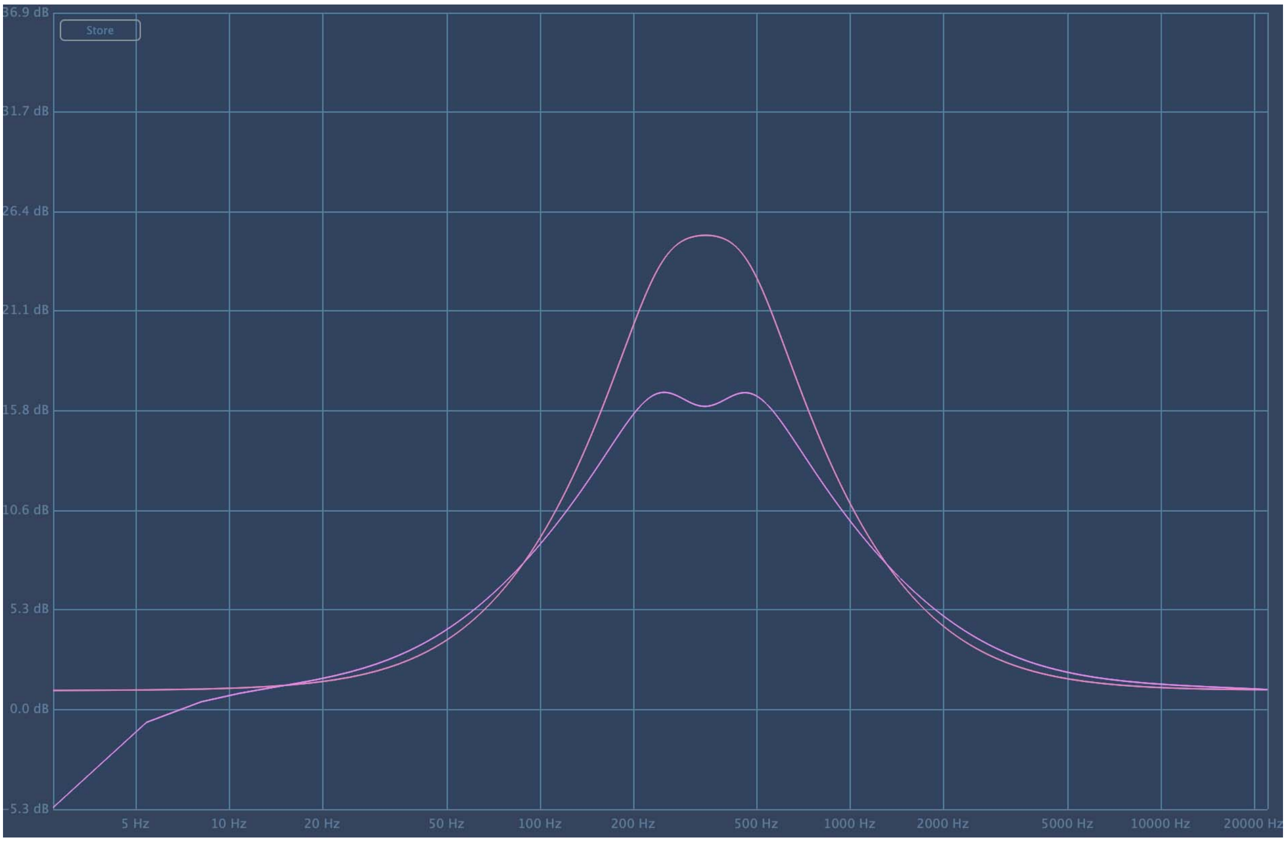
Pulsar 8200 is a parallel equalizer; the electronic component blocks corresponding to the five bands of the equalizer are wired in parallel, which essentially means that they all filter the same input signal, and the filtered signals are then summed. This is different from most modern equalizers, where the filters are applied successively to the signal and said to be in series.

We will now illustrate the difference in correction that parallel equalisation produces compared to serial equalisation. Consider two bell filters of about 15 dB gain, placed at about 250 Hz and 450 Hz, whose frequency responses are as follows:



Two bell filters at close frequencies

Here is the result of combining these two bell filters, with a serial equalizer (pink), and with a parallel equalizer (purple). It can be seen that with filters of 15 dB maximum gain, the serial EQ gives a response of up to 30 dB. With a parallel equalizer, on the other hand, the total response will never exceed 20 dB.

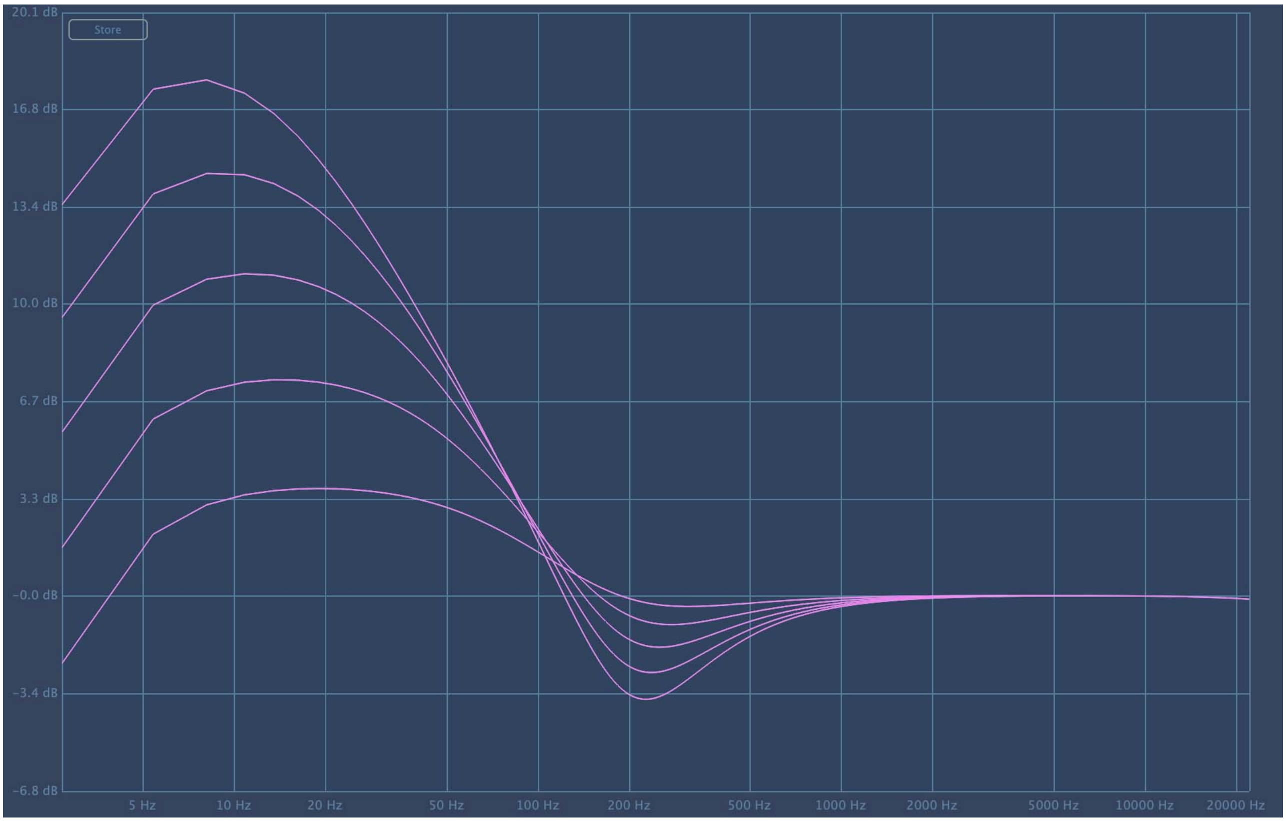


Difference between the combination of two bell filters in series and in parallel

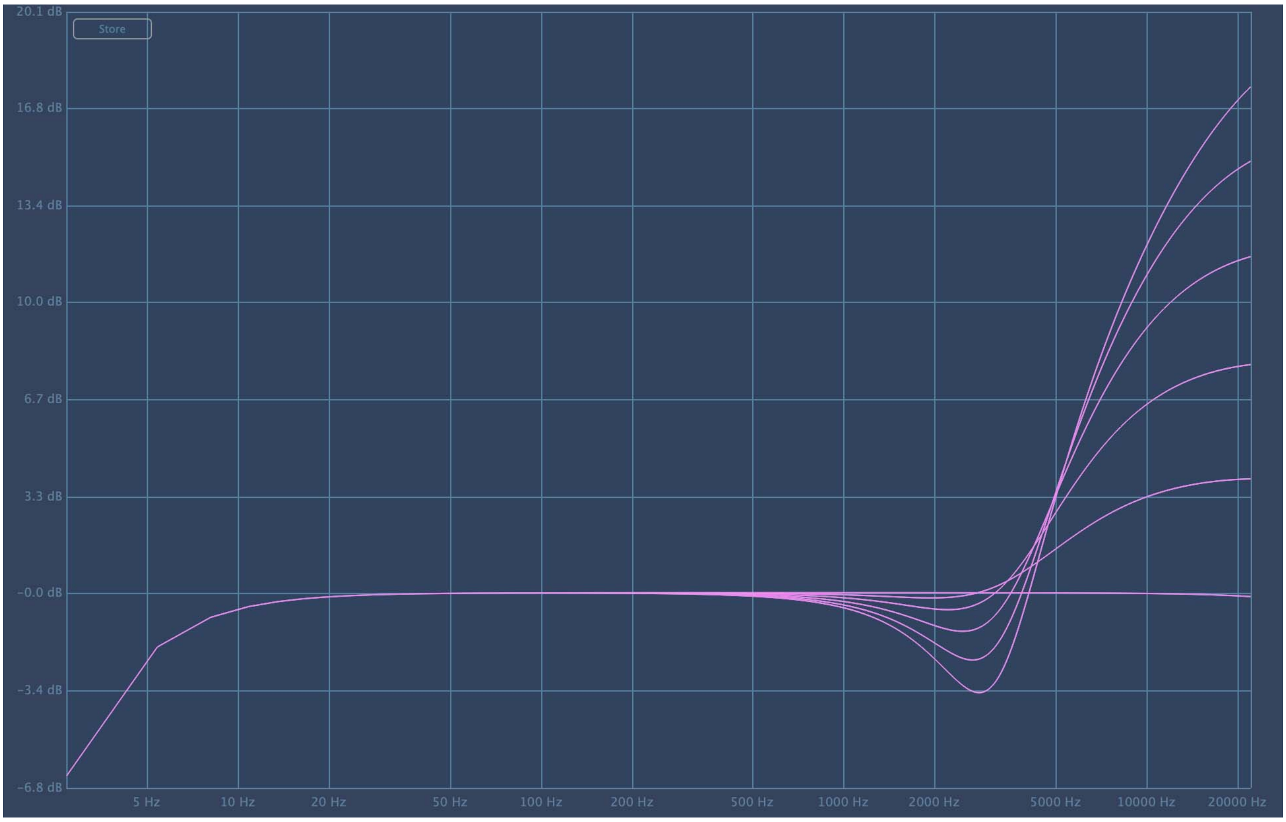
Additional filters

As usual, we have added a number of features to the Pulsar 8200 to increase the flexibility and sound range of the original machine, including a number of additional filters:

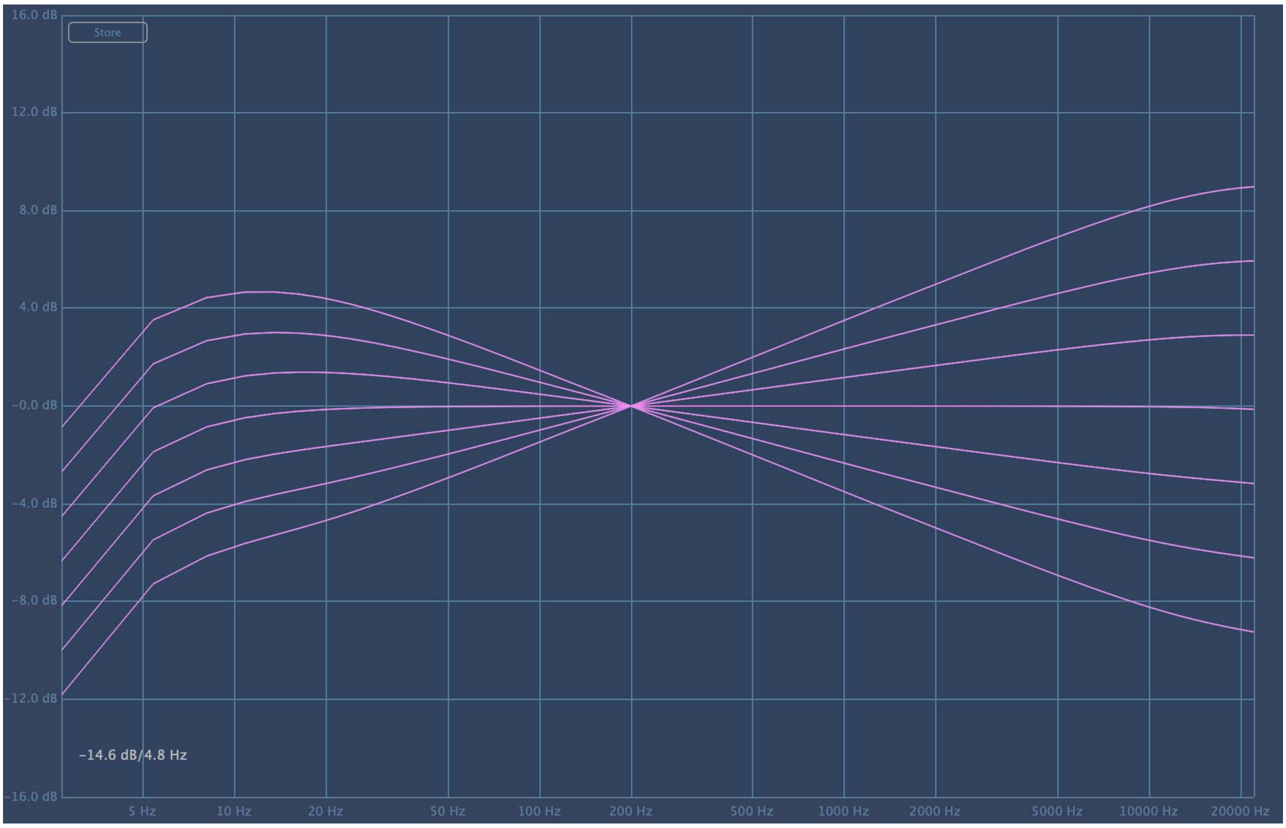
- High-pass and low-pass filters with a slope of 12 dB per octave, common in today's equalizers
- A "Sub" filter, inspired by the boost/cut combo of passive EQs such as the Pultec EQP-1A, to boost the low end of the spectrum without the "muddy" side of conventional shelf filters
- A "Tilt" filter, consisting of a constant slope between 20 Hz and 20 kHz, to subtly adjust the bass/treble balance
- An "Air" filter, also inspired by the boost/cut of passive EQs such as the Pultec EQP-1A, to boost the high end of the spectrum without the "aggressiveness" of conventional plate filters



Sub" filter with different gain values (from 0 dB to 20 dB)



Air filter with different gain values (from 0 dB to 20 dB)



Tilt filter with different slope values (from -1.5 dB/Octave to 1.5 dB/Octave)

The De-eesser

A de-esser has also been added to complete the package. A de-esser is a special kind of compressor that focuses on a specific frequency range. It is used to reduce high-frequency transients or unpleasant sibilance.

The de-esser is generally used for vocal mixing, and is the most common mixing tool for dealing with sibilance ("sss" or "chhh") in a vocal take, as it often gives a more natural result than a simple equalizer. However, this kind of device can be very useful on other types of sources that present aggressiveness in a transient way at the top of the spectrum: cymbals, hi-hats, synths, keyboards, guitars\'85 and it can even be useful on a bus mix.

The De-esser works with a detection filter that can either be focused on the unpleasant frequency (in bandpass mode) or set to a wide range of frequencies (in highpass mode). A parameter is then used to adjust the "amount" of signal removed when these frequencies reach too high a volume. The de-esser acts, depending on the mode chosen, as a negative-gain bell filter (in bandpass mode) or as a negative-gain high-shelf filter (in high-pass mode) that would remove the annoying frequencies automatically.

Quick start

Installation

Pulsar 8200 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 .

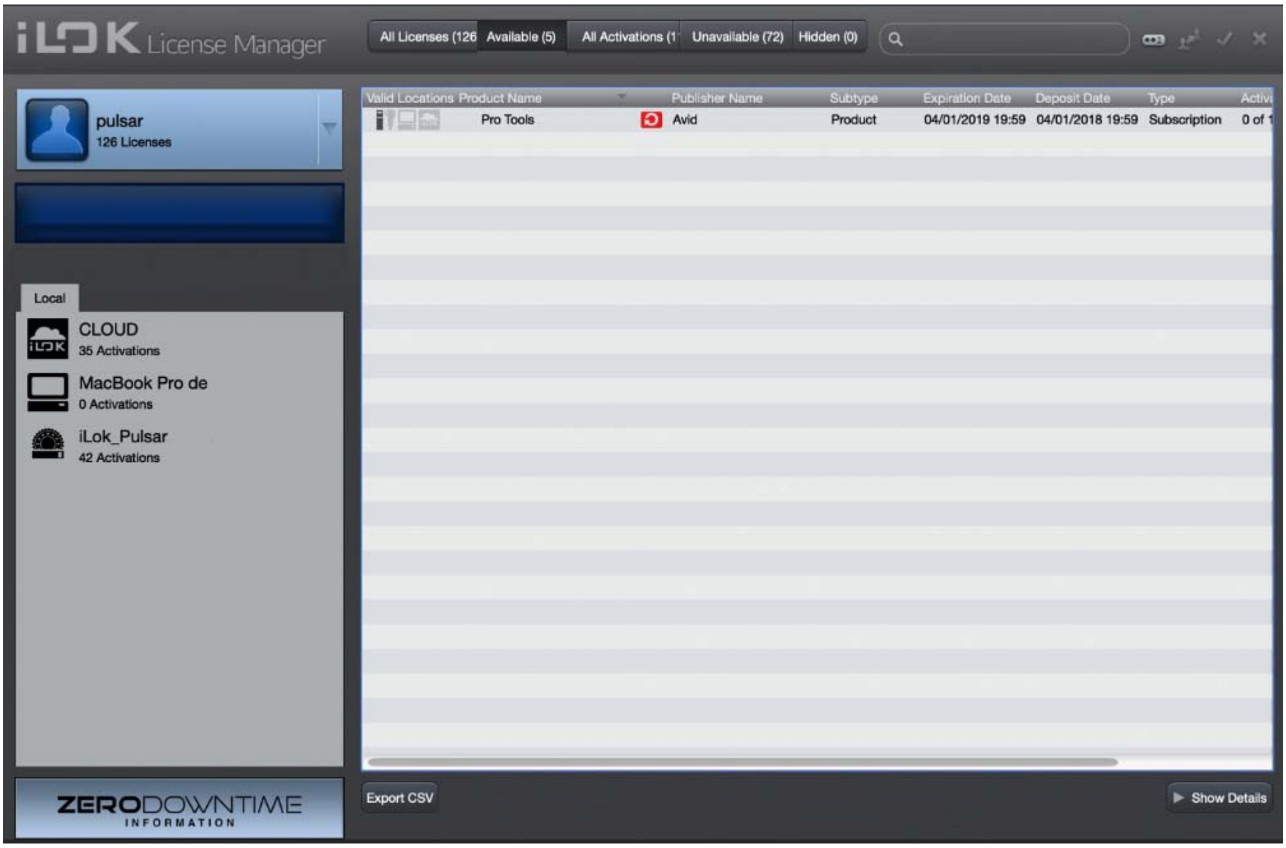
You can choose between three activation methods:

- 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’s "available" license tab

First steps

Load Pulsar 8200 onto a track of your choice in your DAW. A good starting point is to load a basic preset that matches the type of channel (vocals, guitar, bass, drums...). From there :

- Adjust the gain, frequency, and bandwidth of the different bands according to the desired result. (The bands are sometimes named according to their role, for a better understanding of the use of the preset.)
- Start playback and compare with and without using the bypass (power button). You can activate the Auto-Gain mode, to be able to compare at equivalent volume
- Adjust the frequency and amount of the de-esser to selectively attenuate aggressive transients over a localized frequency band

This allows you to browse through the many factory presets available, so you can 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 throught 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.

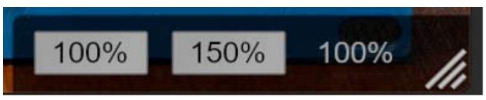
Using the GUI resize control

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.

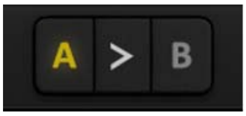
Save and Save As buttons

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.

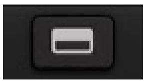
It is also possible to copy the state A to B or vice versa using the > or < buttons located between A and B.



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.

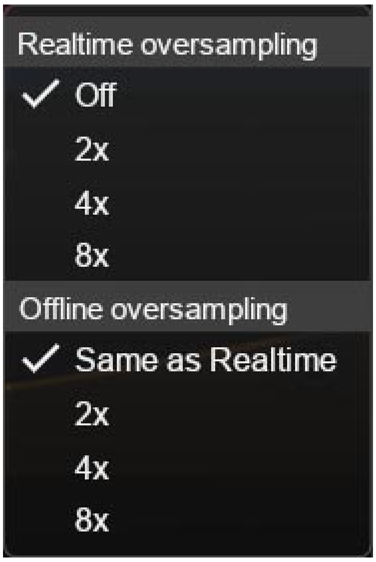

The Menu button

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

By default in Pulsar 8200, controls are moved continuously, allowing you to select any frequency, gain or width value on a continuum. You can enable discrete steps and choose their resolution, making those controls move in increments for a more 'old-school' workflow.

Gain knobs offer the choice between continuous operation or 0.5dB / 1dB / 3dB steps.

Frequency knobs (not including the LP/HP knobs) offer the following options:

- No (default): The frequency knobs are continuous and all intermediate frequencies can be accessed.
- Stepped: The frequency knobs are stepped. The exact steps for each filter vary but should be visibly logical as the band moves through the frequency graph on the upper rack. (

For Width (Q) 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, with six values accessible.

Note: when controls are stepped, it is still possible to access intermediate values by using the fine editing mode when using the control. Just hold 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 8200 control panel

The 8200's control panel is directly inspired by the GML 8200, with a few liberties taken to expand the sound palette as well as to offer plug-in users better ergonomics.

This panel consists of two racks:

- The EQ control rack
- The display and control rack on the curve

The control rack

This rack contains the main controls useful for setting up the equalizer.

It is organized in two identical and symmetrical sections, left and right, which correspond to the settings of two stereo channels (in mono, the right section is unused), as well as a central section with the common controls (Power, Auto-gain, De-esser, LR/MS, Link).

Control rack

On each side, on the upper part, one can note the presence of five almost identical vertical slices. These control the settings of each of the five equalization filters (Gain, Width (Q), Frequency), corresponding to the bands of the original machine.

Control strip of a band

On the lower part, there are five additional bands per channel, an addition compared to the original machine: high-pass and low-pass filters, plus Air, Sub, and Tilt bands.

Additional bands

Power

The switch in the centre of the interface allows the effect to be switched on and off (ie, bypassed).

Power button

Gain

This knob, absent from the original machine, is used to adjust the output gain of a channel, and thus possibly compensate manually for the gain lost or gained after filtering.

Output Gain knob

Auto-gain

When Auto-Gain is on, this mode applies a compensatory gain derived from the settings of the different bands, and will help you to keep the sound power constant while adjusting the EQ controls.

Note: the algorithm used does not perform real-time sound power measurements but estimates based on the EQ settings only. This means it will work on a larger range of musical sources. Depending on the source, you may need to make small adjustments with the output gain knob.

Auto-Gain button

Mid/Side

This switch selects the stereo management mode – what signal Pulsar 8200's two sets of controls are actually processing.

- L/R (switch off) corresponds to the traditional stereo mode (left / right channels)
- M/S (switch on) stands for Mid/Side mode. In this mode, the left-hand set of controls (previously the left stereo channel) now controls the Mid information – the sound information that's present in both left/right stereo channels at the same time. The right-hand set of controls (previously the right stereo channel) now controls the Side channel – the sound information that's different between the left and right stereo channels.

The M/S mode allows you to equalise the sounds that are panned more in the centre and those that are panned more to the sides with different settings. It is therefore recommended to deactivate the "Link" switch when working in M/S. In fact, this happens automatically when Mid/Side mode is activated.

You could use Mid/Side mode to collapse the stereo width of low frequencies to mono by setting up a high-pass filter only on the Side channel. There are plenty of creative and corrective uses for mid/side too.

Mid/Side switch

LR Link

This switch links the settings of both channels. If it is activated, every setting that is made on the left side is automatically mirrored in the right channel (and vice versa).

Note: The relative distance between the two channels is maintained when the channels are linked. Attempting to reset a knob on one channel to default will cause the other to move by the same amount, not necessarily to the same destination setting.

LR Link switch

Gain Scale

This control acts as a "Master" gain knob, intended for attenuation only. It applies the same attenuation factor to the gains of all the filters in the Pulsar 8200. For example, if a bell filter is active and has a gain of 8dB, setting the Gain Scale knob to 50% will result in a bell filter with a gain of 4 dB. You can use this control to back off a signal's overall EQ processing.

Gain Scale knob

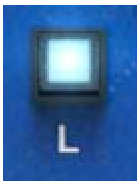
Listen

These headphone-shaped buttons allow you to listen to only one channel. This is particularly useful in M/S mode for fine-tuning the EQ for one of these channels in isolation.


Listen switch

Channel bypass

These switches are located on each channel (on the outside of the interface) and allow each channel to be deactivated independently.



Left channel bypass switch

De-esser Freq

Allows you to adjust the frequency of intervention of the De-esser.

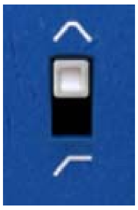
- If the de-esser is in "band-pass" mode (central switch in high position), the de-esser will only operate on a frequency band centred on the selected frequency
- If the de-esser is in "high-pass" mode (central switch in low position), the de-esser will only intervene at the top of the spectrum, starting at the selected frequency

Note: by holding down the **Shift** key while adjusting this parameter, you will be able to listen solo to the relevant frequency band, which can be useful for adjusting the bandpass filter on a sibilant for example.

De-esser Frequency

De-esser filter type

This switch is a selector for the type of filter used to attenuate transients. Choose between band-pass (Top) and high-pass (bottom) types.


De-esser filter type

De-esser Amount

Adjust the strength of the De-esser's transient attenuation. The higher this parameter is set, the more transients are attenuated. This parameter should be adjusted to achieve sufficient attenuation without audible artefacts.

Note: by holding down the **Shift** key while adjusting this parameter, you will be able to audition the transients attenuated by the De-esser.

De-esser Amount knob

De-esser Bypass

Activate / deactivate the De-esser.

De-esser bypass button

De-esser Delta

Clicking on this button will allows you to audition only the transients attenuated by the De-esser.

De-esser delta button

Band bypass

This switch simply activates or deactivates the band.

Band bypass switch

Band name field

An editable text field is present above each band for naming. This allows the user to note the role they have determined for that band during the equalisation process, e.g. "clarity", "air", etc. These are often made use of within presets to help guide their use.

Editable band name field

Gain of an equalizer band

This knob adjusts the filter gain of the relevant EQ band.

Note: It is possible that the actual gain is slightly different from the gain displayed on the screen print. This is because on the original machine, small errors in the component values cause slight biases.

Filter gain adjustment knob

Freq of an equalization band

This knob is used to adjust the filter frequency of the relevant EQ band.

- Bell filter: the adjusted frequency will be the centre frequency of the filter (at the top)
- Shelf filter: the adjusted frequency will be the lower corner frequency for band 1, or the upper corner frequency for band 5

Note: by holding down the **Shift** key while adjusting this parameter, you will activate the Band Solo function, which allows you to audition the effect of only that band on the input signal.

Knob for adjusting the frequency of the EQ band filter.

Width of an EQ band

This knob adjusts the bandwidth of the filters. The higher the setting, the narrower the bandwidth. On bands 1 and 5, setting this knob to the minimum value will turn the Bell filter into a Shelf filter - low-shelf for band 1, and high-shelf for band 5.

Filter Frequency adjustment knob

HP

This knob allows you to activate the HIgh-pass filter and adjust its frequency. It's also possible to adjust its slope (from 6 dB/octave to 48 dB/octave) via a pop-up menu, by right-clicking on the knob.

High-pass filter frequency

Note: it's possible to adjust the slope of this filter in a pop-up menu displayed when right clicking on the knob.

Sub

Sub band gain control. This band, inspired by the famous boost/cut combo of the Pultec EQP-1A passive equalizer, allows you to add energy to the lower end of the spectrum (the so-called "sub" frequencies) without the muddy edge brought by classic shelf filters.

Sub filter gain

Tilt

This knob is used to adjust the slope of the "Tilt" filter (explained in the introduction to this manual). Above 0 dB/Octave, you will add brightness and clarity while curtailing bass frequencies; below 0 dB/Octave, you can take the edge off harsh highs while increasing power in the lows.

Tilt filter slope (in dB/octave)

Air

"Air" Band Gain Control. This band, inspired by the famous Boost/Cut of the Pultec EQP-1A passive equalizer, uses a pair of passive EQ filters to add energy at the very top of the spectrum (so-called "air" frequencies) without the aggressiveness of conventional shelving filters.

Air filter

LP

This knob allows you to activate the Low-pass filter and adjust its frequency. It's also possible to adjust its slope (from 6 dB/octave to 48 dB/octave) via a pop-up menu, by right-clicking on the knob.

Low-pass filter frequency adjustment knob

The curve display rack

Curve display rack

The top half of Pulsar 8200's window is taken up by the curve display rack. This section allows you to:

- View and edit the frequency response of the equalizer on both channels
- Visualize the spectrum of the output signal on several time scales
- Analyze the input and output levels, as well as the level difference between input and output, plus the De-esser's response

Equalization using the curve display

In the upper rack, the curve display shows the frequency response of the signal selected at the bottom of this rack (Left or Right, Mid or Side), as well as the frequency responses of the EQ individual bands as part of the overall EQ curve.

- 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 of your QWERTY keyboard to control the active channel.
- The chain button is used to activate or deactivate the link between channels. It is a repeat of the control 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 QWERTY 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 / Cmd** 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 audition the effect of only that band on the input signal.
- Use the mousewheel when hovering your cursor over a band to change its Q, or its slope on the HPF/LPF

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
- Click on any central field (name, frequency, gain, Q) to type 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 customize what is displayed:

- 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 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.

If the De-esser is enabled and reduces the gain of a portion of the bandwidth, the gain reduction will be displayed in red.

While in Infinite Spectrum mode, changes made in the EQ can be seen immediately, despite the long time window of analysis. This is thanks to FFT measurements for the visualization being taken before the EQ, and the EQ curve being applied to that measurement.

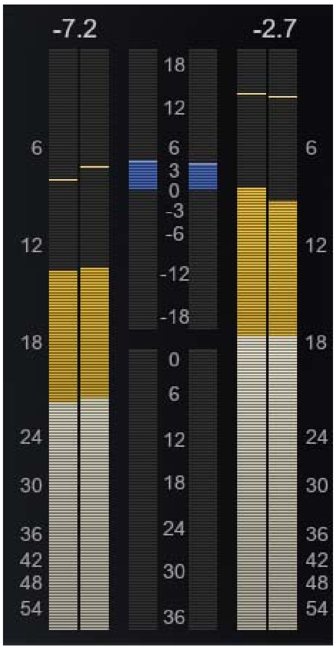
Level visualization

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 (white), 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 top 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.

The bottom central indicator (red) displays the instantaneous gain reduction level of the De-esser.



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)
- VST 3 (Windows: 32/64-bit, Mac: 64-bit)
- AAX (Windows: 64-bit, Mac: 64-bit)
- Audio Unit (Mac: 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
- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

MacOS

- CPU: Intel Core i3 / i5 / i7 / Xeon / Apple Silicon (M1, M2, etc.)
- Memory: 4 GB RAM / 1 GB free disk space
- Operating system: 10.9 and higher
- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

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