Pulsar VM-Comp
User Manual

www.pulsar.audio

Table of contents

Table of contents

Introduction

This manual describes the features and operation of the Pulsar VM-Comp effect processor. To be sure you understand how to use your plugin and appreciate 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

« Variable-bias » compressors

The term « variable-bias » refers to a dynamic compression technology that appeared in the 1950s (for example, the legendary Fairchild 670, originally designed for the vinyl mastering process, but very quickly adopted in the studio), which consists in varying the polarization of a tube to control the gain. The compression ratio is variable, and also depends on these parameters.

The result is compression with a slower attack than a FET or VCA compressor, but an extremely musical behavior, making it an exceptional bus compressor, both for a drum bus and for an entire mix. This type of compressor is also present in the vast majority of mastering studios.

In addition, its design based on transformers and tubes generates a slight harmonic distortion that brings subtle presence and warmth, and this famous « glue » effect that makes all the sounds sit musically in the mix.

The Pulsar VM-Comp will offer you:

- The sound of a legendary machine, present in studios all over the world
- A musical bus compression
- Unparalleled warmth and presence
- The advantages of « organic » analog sound as well as the advantages of digital sound (presets, parameter automation, etc.)

Quick start

Installation

Pulsar VM-Comp 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 .

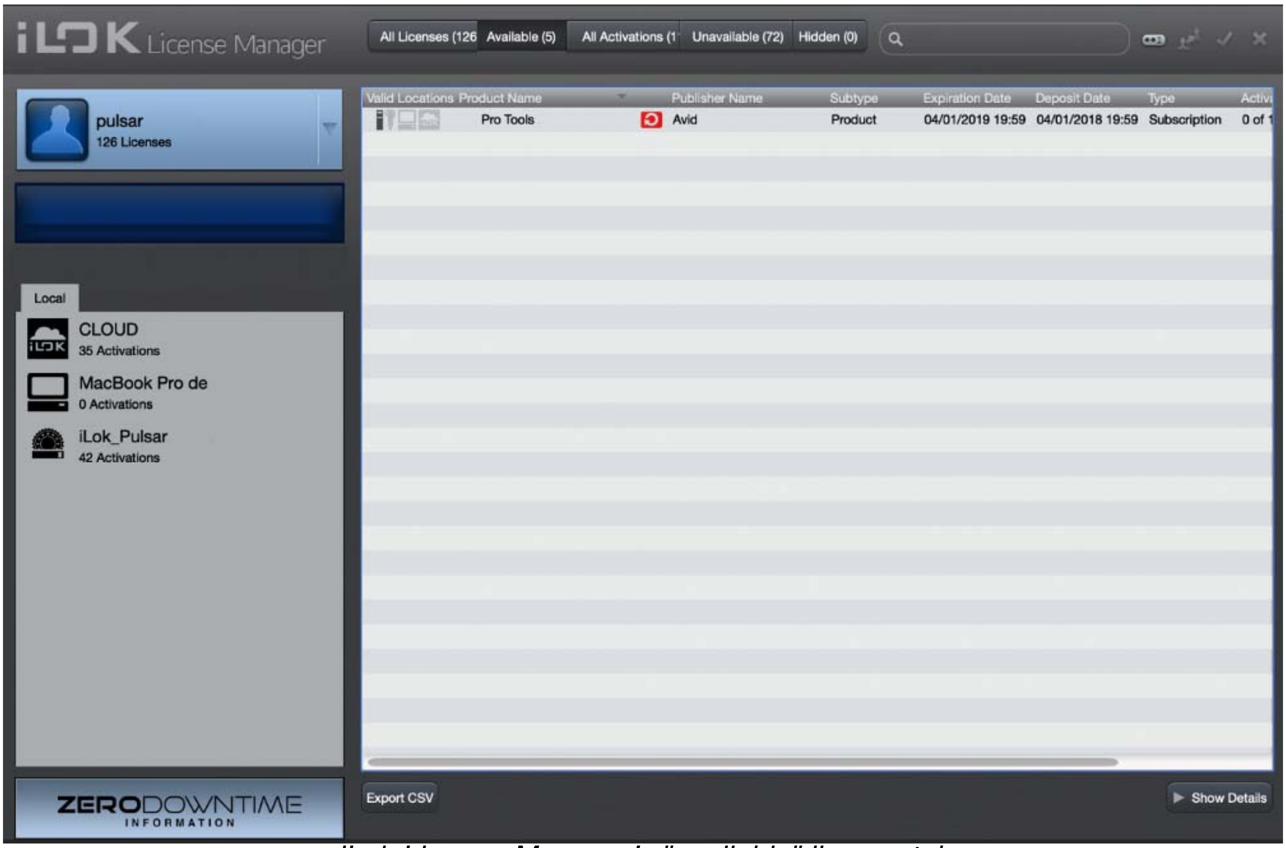
You can choose between three activation methods:

- 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 the VM-Comp on the track of your choice in your DAW. The base preset is a good starting point. You can now:

- Adjust the « Dual Input » gain in order to obtain the Gain Reduction suggested in the preset, for example -3 dB for the base preset. The amount of Gain Reduction is displayed in the visualization rack at the top.
- Adjust the amount of compressed sound with the « Mix » knob.
- Adjust the release and attack times using the « Attack » and « Release » knobs

You can also review the available factory presets to find inspiration quickly, without getting into technical considerations.

The user interface

The user interface

The user interface consists of 2 separate panels:

- The toolbar, common to all Pulsar Audio plug-ins (top)
- The control panel, specific to the plug-in. This panel consists of 3 racks: the visualization rack, the compressor control rack and the advanced detection functions rack

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.

Parameter locking

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.

Control surface and multi-channel parameter edition

If you are using a control surface, such as AVID S1/S6 or Mackie HUI, to control your Pulsar plug-in, and the plug-in allows independent control of L/R or M/S channels:

- When the "link" option for the controls of the two channels is disabled, automation reading/writing and parameter control via the control surface operate as expected, with each control functioning independently.
- When the link option is enabled, only channel A parameters are utilized, corresponding to either the Left or Mid channel based on the selected stereo mode. Adjusting channel A parameters via the control surface or reading automations from channel A automatically synchronizes the parameters of channel B. **Automations for channel B parameters are ignored, as are changes to channel B parameters made via the control surface.** In addition, channel B automations are not recorded.
- **Note :** In Pro Tools, contrary to the information stated above, automations are written to both channels A and B. Even with the link option enabled, playing an automation on channel A does not synchronize channel B. This behavior occurs because, when recording an automation for a linked parameter in Pro Tools, both channels are recorded separately. During playback, the automation tracks for channels A and B are read independently, without interacting with the link feature.

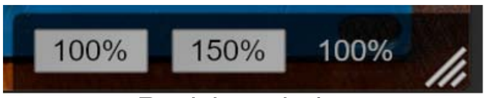
Using the GUI resize control

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: right-clicking on any of these buttons provides access to the list of stored operations, allowing you to undo or redo multiple actions at once.

Undo / redo buttons

Preset Selection

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

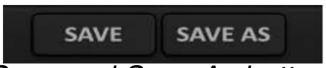
- Read the name of the current preset. If an asterisk appears after the preset name, it means that the state of the plugin no longer matches the saved preset
- Select a preset from the list of available presets, arranged in sub-banks
- Delete the current preset (« Delete Preset » option)
- Rename or 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.



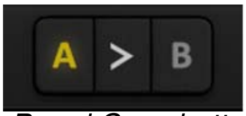
Save and Save As buttons

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

Menu Button

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

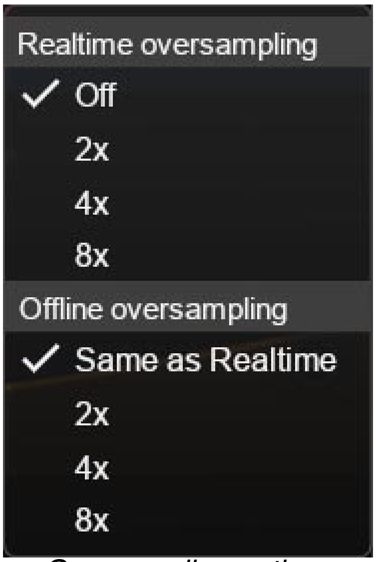

The Menu button

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.

Disable Static Noise option

All analog equipment introduces a hiss, mainly caused by thermal noise in the electronic components, the amplitude of which differs from one model to another.

In some Pulsar plugins, we assumed that modeling this noise was appropriate, although at a lower level than in real life (often around -90 dBFS), because it contributes slightly to the character of the original device.

In some cases (if the output of the plugin is strongly amplified), this noise can become audible and undesirable, so it is possible to deactivate it using the "Disable Static Noise" option.

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 VM-Comp's control panel

The VM-Comp's control panel is inspired by a famous piece of hardware, with some additions to expand the sound palette.

The control panel consists of 2 racks:

- The compressor's "analog" control rack (at the bottom)
- The visualization and advanced sidechain features rack (at the top)

Control rack

This rack contains the main controls (switches and knobs) used to set up dynamic compression.

Control rack

Bypass

This switch allows you to activate and deactivate the effect.

Bypass switch

Dual Input

This knob is used to adjust the amount of signal at the compressor's input. The higher the input level, the more audible the color of the tubes will be.

Dual Input knob

Output

This knob is used to adjust the output level after compression (to compensate for the input gain and the gain reduction due to compression).

Output knob

Threshold

This knob sets the compression trigger threshold. The lower it is, the more the compressor will compress the signal.

Threshold knob

Note: the compression ratio also depends on this setting. The lower the Threshold, the higher the ratio.

Attack

This is the knob for adjusting the compressor attack time. The shorter the attack time, the more the compressor will be triggered on percussive elements. This time is between about 2 milliseconds (Fast) and 100 milliseconds (Slow).

The white positions reproduce the behavior and timing of the original hardware design, while the orange positions extend the range by offering faster attack times. These additional steps allow you to achieve a tighter, more modern response when needed, without altering the familiar feel of the classic settings.

Attack knob

Note: the actual attack time also depends on the Release setting.

Release

This knob is used to adjust the compressor release time. The shorter the time, the more audible the pumping effect will be. This time is between about 70 milliseconds (Fast) and 1.8 seconds (Slow).

This parameter can be set to any of the 1-2-3-4-5-6-7 positions, or to any value in between using the fine edit mode (by holding down the Ctrl or Cmd key).

The grey positions (1 to 5) follow the response characteristics of the original circuitry, whereas the orange positions (5 to 7) provide extra-fast release times. These added steps give you greater flexibility for punchier, more reactive compression, while still preserving the musicality of the traditional settings.

Release knob

Note: The actual release time also depends on the Attack setting.

HI RATIO switch

This switch is used to adjust the compressor ratio (« quantity » of compression):

- HI RATIO switch Off: ratio of about 1.5:1 – light compression
- HI RATIO switch On : ratio of about 4:1 – limiting mode

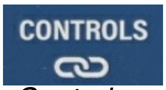
The effective ratio can grow to much higher levels in case of large gain reduction.

HI RATIO switch

It should be noted that due to the electronic structure of this compressor, the ratio is not fixed; it depends on the position of this switch, but also on the input gain and threshold levels.

Link Controls Switch

This switch links the settings of the two channels. If enabled (the links of the icon are connected) each setting that is adjusted on channel 1 is also adjusted on channel 2 (and vice versa).



Link Controls switch

Link Sidechain Switch

This switch is used to connect the sidechains (detection circuits) of the 2 channels. If it is activated ("Linked" is selected), the two detection channels are averaged, applying identical gain reduction to both channels. This enables, for example, the stereo coherence of a master bus to be kept in case of high compression.

Link Sidechain switch

ST MODE Switch

This switch selects the stereo management mode: traditional Left/Right (L - R) stereo, or Mid/Side (M - S) stereo. In the latter case, the left channel is the Mid (Center), and the right channel is the Side (Sides).

ST MODE Switch

Mix

This knob is used to crossfade between the compressed signal (MAX) and the original signal (MIN).

Mix knob

Channel Listen buttons

These buttons allow you to listen to one of the two channels:

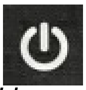
- The L or M channel (depending on stereo mode)
- The R or S channel (depending on stereo mode)


Channel Listen Button

Channel bypass Buttons

These buttons allow to bypass one of the two channels:

- The L or M channel (depending on stereo mode)
- The R or S channel (depending on stereo mode)


Channel bypass Button

Visualization and Advanced Sidechain Rack

Visualization and Advanced Sidechain Rack

This control and visualization panel shows important levels such as Gain Reduction (GR) over time, as well as allows various settings related to the sidechain signal, which is the inaudible signal used to calculate gain reduction.

Pulsar VM-Comp is equipped with comprehensive features allowing you to tailor the compression response.

Sidechain Listen

This headphone-shaped button lets you listen to the sidechain signal, for example to verify the presence of the correct signal in case of an external sidechain, or to hear the effect of the EQ on the sidechain signal when adjusting its filters.

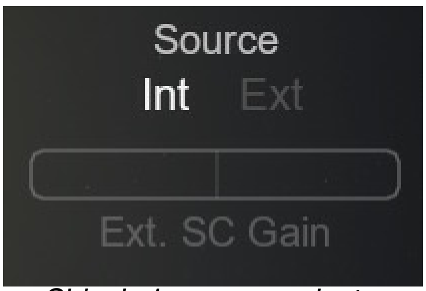
Sidechain Listen button

Source (INT / EXT)

This button selects the sidechain source. In "INT" mode, the compressor operates normally, using the audio input as sidechain source.

If "EXT" is selected, an external audio signal is used as sidechain source. It is then necessary to assign a sidechain (external sidechain) channel in your DAW — please refer to your DAW manual for routing details.

In the case of external sidechain, the input gain (and thus compression amount) can be adjusted via the "Ext. SC Gain" slider.



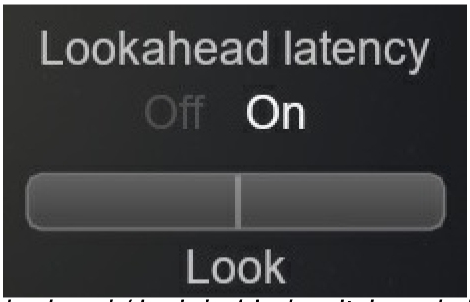
Sidechain source selector

Look (look-ahead, look-behind)

This slider adds positive or negative delay in the sidechain path. Moving the slider to the left anticipates the sidechain signal relative to the audio signal (look-ahead mode, from -20 ms to 0 ms), allowing compression of very fast transients. Moving right allows transients to pass more freely, for instance to be limited later by a Clip/Limiter module (look-behind mode, from 0 ms to 10 ms).

When zero-latency mode is enabled in the Options menu, look-ahead is disabled, and only the right side of the slider (look-behind) remains available.

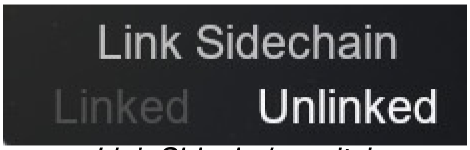
Note: *the effect of look-ahead / look-behind can be visualized in the scrolling GR curve, where a slight offset will be visible between the input volume envelope and the GR curve.*



Look-ahead / look-behind switch and slider

Link Sidechain

This switch is used to connect the sidechains (detection circuits) of the 2 channels. If it is activated ("Linked" is selected), the two detection channels are averaged, applying identical gain reduction to both channels. This enables, for example, the stereo coherence of a master bus to be kept in case of high compression.

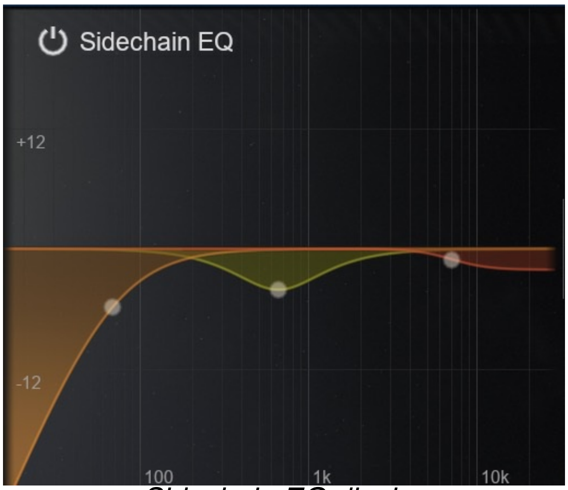


Link Sidechain switch

Sidechain Equalizer

This section filters the sidechain signal using a parametric EQ controlled via the curve. A spectrum display of the sidechain signal is also present to facilitate visualization of important frequencies.

The top-left switch enables or disables the EQ. Band parameters (frequency / Q / gain / filter shape) can be controlled directly on the curve or via a dialog box that appears when hovering over a band.



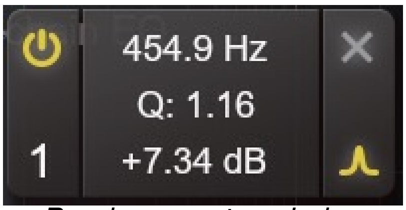
Sidechain EQ display

To add a band, move the mouse over the EQ surface. A gray line previews the band to be created on click. Depending on mouse position at click, different band types are created:

- Top-left: low-shelf filter
- Bottom-left: high-pass filter
- Center: bell filter
- Top-right: high-shelf filter
- Bottom-right: low-pass filter

Various mouse actions can then be performed:

- As with other controls, hold **Ctrl** (on Windows) or **Cmd** (on Mac), or use the **right mouse button** , for fine adjustments
- Click on a band while holding **Alt** to delete it
- Hold **Shift** while moving a band to activate Band Solo mode, which plays only that band's effect on the sidechain signal. Band Solo temporarily overrides the detection signal so you hear only the frequencies affected by that band.
- Use the mouse wheel over a band to adjust Q (for bell or shelf filters) or slope (for low/high-pass filters)



Band parameter window

The band parameter window appears when hovering over a band.

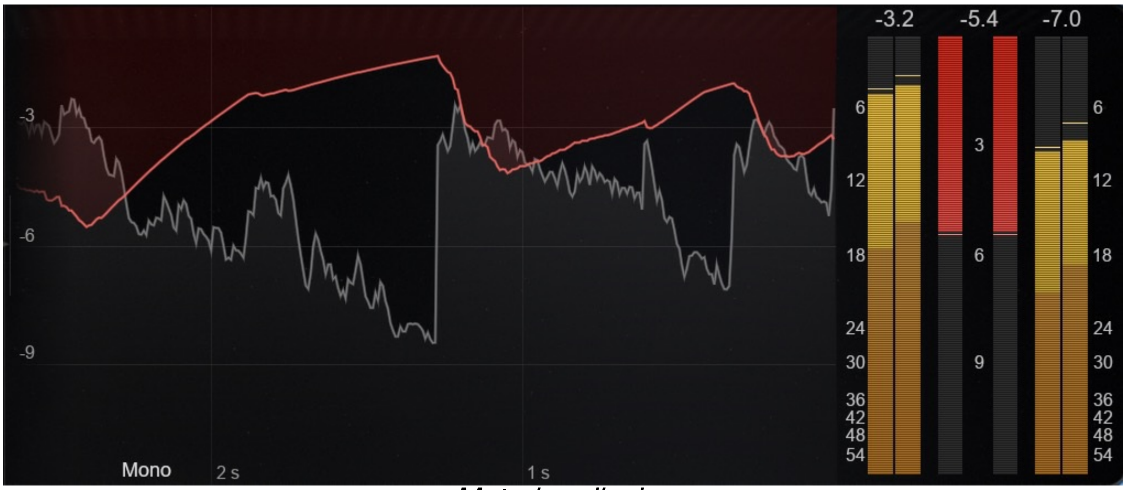
- The top-left button activates or deactivates the band
- The bottom-left number shows the band number (1 to 4), useful for identifying bands in automation
- The top-right cross deletes the band
- The bottom-right button changes the filter shape
- You can click frequency, gain, Q, and slope values to enter them manually

Filtering the sidechain signal is useful to adjust compression based on content:

- Cutting lows on a master bus compression reduces triggering on kick drum hits, resulting in more discreet pumping
- Reducing high frequencies on overheads or drum bus reduces pumping on crash cymbal hits
- A bell filter lets you selectively increase or decrease compressor triggering on a spectral element (e.g. snare or vocal)

Metering

The right side of the rack is dedicated to the metering section. It features a large scrolling graph displaying both the input volume envelope and the gain reduction over time, along with vertical meters for input, output, and GR (Gain Reduction) levels.



Metering display

Scrolling Curves

The main display on the left shows the time-based input volume envelope (in gray) alongside the gain-reduction curve(s) (in red and green), allowing you to observe the relationship between audio peaks and the compressor’s response, including the influence of attack and release settings.

The gain-reduction scale, ranging from 0 dB to –12 dB, is shown on the left. The volume-envelope scale ranges from 0 dB to –60 dB; although its graduations are not displayed, it approximately matches the scale of the input level meters.

The small button in the lower-left corner—visible only in stereo instances lets you change how the volume envelope and gain-reduction curves of each channel are displayed in the scrolling views:

- **Mono** (default): displays a single mixed volume-envelope curve (L + R, or M + S in M/S Mode), and a single mixed gain-reduction curve (GR-L + GR-R or GR-M + GR-S in M/S Mode).
- **Channel-1** : displays only the volume and gain-reduction curves of the L channel (or M channel in M/S mode).
- **Channel-2** : displays only the volume and gain-reduction curves of the R channel (or S channel in M/S mode).
- **Stacked** : displays each channel’s curves separately, one above the other.
- **Overlaid** : shows a single mixed volume-envelope curve (L+R, or M+S), while displaying both gain-reduction curves simultaneously, using different colors: red for the first channel (L or M) and green for the second channel (R or S).

Input and Output Levels

The vertical meters on the right display the input and output levels for each channel. They include RMS (lighter), peak (darker), a “Peak Hold” bar that retains the highest peak for several seconds, and a numeric readout showing the peak value in dBFS.

A red LED lights up when the peak level exceeds 0 dBFS and remains lit until manually cleared by clicking the meters.

Gain Reduction Levels

The vertical red meters ranging from 0 dB to –12 dB show the instantaneous gain reduction applied to each channel (L/R, or M/S in MS mode). A “GR Hold” bar, whose value is displayed above the meter, captures the maximum gain-reduction peak reached over a short period.

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 (64-bit only)
- VST 3 (64-bit only)
- AAX (64-bit only)
- Audio Unit (64-bit only).

Windows

- CPU: Intel Core i3 / i5 / i7 / Xeon
- Memory: 4 GB RAM / 1 GB free disk space
- Graphics card: OpengGL 2.0 compatible GPU
- 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: 10.11 and higher
- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

License agreement

This license agreement concerns and describes your rights and the conditions under which you may use your Pulsar Audio software. We recommend that you read the entire agreement. By accepting the present agreement or by using Pulsar Audio software, you accept all these conditions.

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Updates

This license gives you the right to all minor updates (e. g. 1.1 to 1.2), but excludes major versions (e. g. 1.x to 2.x).

License transfer

You may transfer all your rights to use the Software to another person provided that you transfer this Agreement and the Software to that other person; and that the recipient accepts the terms and conditions of this Agreement and any other provisions pursuant to which you have acquired a valid license to use this Software.

Activation

Pulsar Audio will not be held responsible for any failure to activate PACE's iLok protection system / license.

Trial

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