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Introduction

This manual describes the features and operation of the Pulsar IPA 25 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

A Legendary Compressor, Redesigned for Today

The **Pulsar IPA 25** draws inspiration from a stereo hardware compressor that has become an essential reference in professional studios worldwide since the 2000s.

Originally equipped with a particularly flexible VCA topology, this unit is renowned for its punch, its ability to beef up a mix without smothering it, and its precision handling of both individual signals and bus groups such as drums, guitars, or even mastering buses. It offers a unique approach to dynamics through a combination of fine tuning, intelligent detection curves, and a customizable internal processing chain.

With the Pulsar IPA 25, we wanted to go further:

- **Faithful modeling as the foundation**—capturing the original unit’s compression behavior and unique character, delivering musical compression for both tracks and buses.
- **Expanded harmonics control**—adjust the VCA drive from clean to saturated, or even swap the gain reduction circuit for variable-bias and diode-bridge emulations, unlocking new sonic possibilities.
- **Advanced sidechain flexibility**—with full EQ, look-ahead, and external sidechain support for precise dynamic control.
- **Clipping/limiting with creative routing**—place it pre-compression, post-compression, or post-mix, transforming the plugin into a complete transient shaping tool.

All housed in a modern, intuitive interface built for today’s workflows.

The Pulsar IPA 25 is both a tribute and an extension. A powerful, versatile, and musical mixing tool designed for demanding producers, musicians, and engineers.

Quick start

Installation

Pulsar IPA 25 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 .

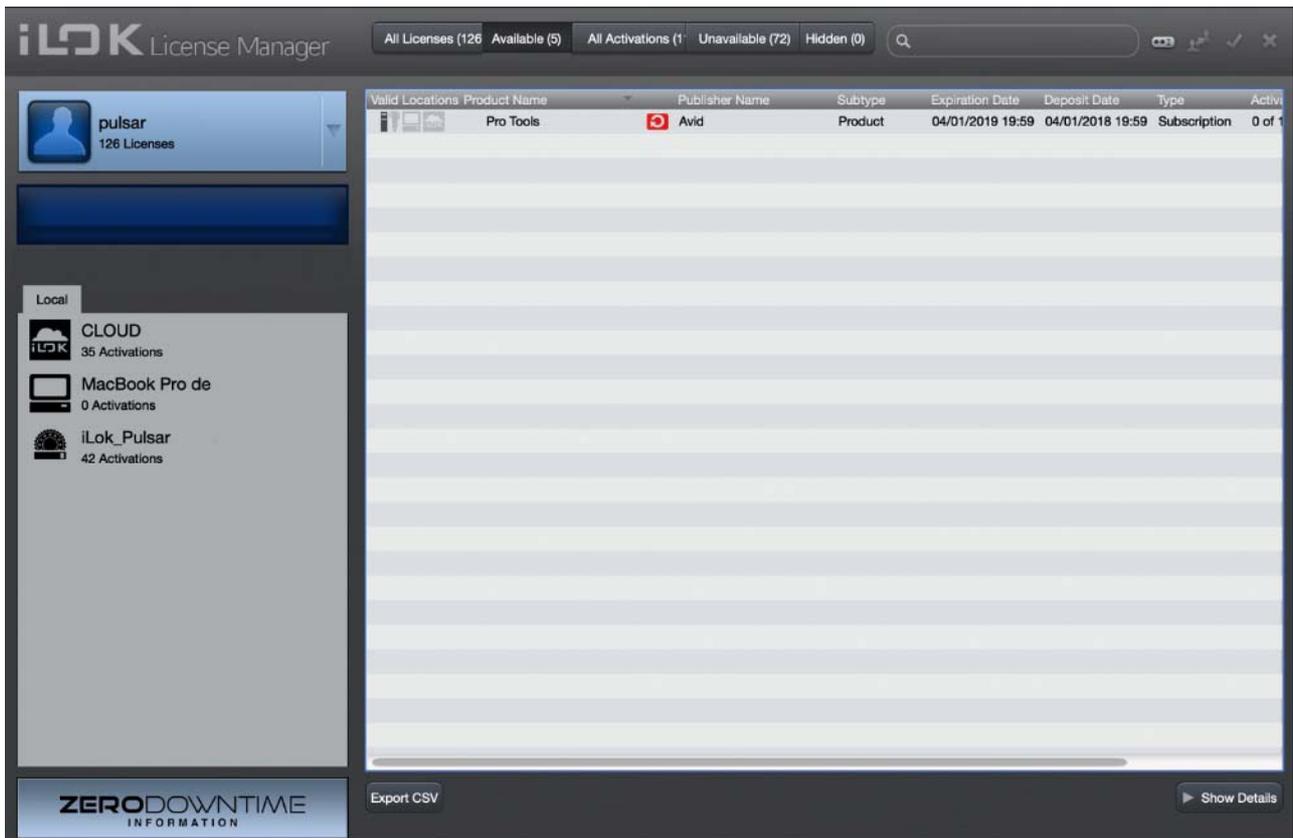
You can choose between three activation methods:

- 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

Getting Started

Load the Pulsar IPA 25 on a track of your choice in your DAW. A good starting point is to load a basic preset matching the track type (vocals, guitar, bass, drums...). From there:

- Start playback and observe the gain reduction applied, by watching either the analog VU meters or the modern red indicators at the top right
- Adjust the Threshold to reach the gain reduction level suggested in the preset name.
- Adjust the attack and release speeds using the Attack and Release knobs. The faster the attack, the more transients are reduced, and the faster the release, the more the sound between transients is emphasized. While playing, observe the effect of these settings on the red gain reduction curve in the top display.
- Adjust the Makeup Gain so that the compressor's output volume matches the input. If the AUTO button is active, an automatically calculated make-up gain is applied instead, which is a quick and easier alternative. After this step, you can evaluate the compressor's effect using the Power switch without being misled by volume differences.
- If needed, adjust the amount of compressed and uncompressed signal using the "Mix" knob.
- Experiment with Ratio, Knee and Punch controls, which can quickly alter the compression response.
- Try adjusting the Clip/Limiter section's Ceiling setting to control transients or add bite.

The plugin comes with numerous presets created by professionals. Feel free to explore them to quickly discover its possibilities, even without deep technical knowledge.

User Interface



IPA 25 user interface

The user interface consists of 2 distinct panels:

- The toolbar, common to all Pulsar Audio plug-ins (at the top).
- The control panel, specific to the IPA 25 plug-in. This panel consists of 2 racks: the compressor control rack, and the visualization and advanced sidechain features rack.

Note that a resize control is available in the interface's bottom right corner, as with all Pulsar Audio plug-ins.

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

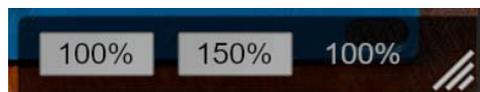
Located at the bottom right of the interface of all Pulsar Audio plugins, this control allows you to resize the plugin's interface to your liking. It comes in the form of three lines, like a classic resizing handle:



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.

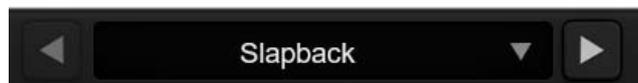


Undo / redo buttons

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

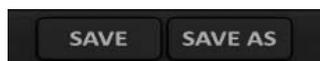


The preset selection area

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

Rack Visibility Button

This button allows you to resize the plugin interface by hiding certain parts. You can choose to display only the main rack if additional features like the EQ and GR visualization are not needed, or conversely display only the visualization rack, for example, to keep an eye on the plugin's activity when no changes to settings are planned.



The rack visibility button.

Menu Button

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



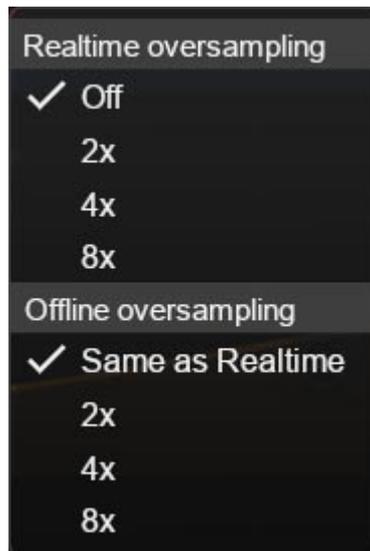
The Menu button

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.

Latency

The IPA 25 introduces 35 ms of latency by default (plus a few additional ms if Oversampling is active), enabling key features such as:

- Look-ahead detection (for precise transient shaping by analyzing the signal in advance)
- Transparent limiting in the Clip/Limiter section (requiring latency for optimal performance)

The plugin automatically reports its latency to your DAW, which compensates for it during playback and rendering. This ensures that the processed audio remains in sync with the rest of your project.

In certain situations, such as real-time monitoring (e.g., when processing an input signal from an audio interface), DAWs cannot compensate for latency, causing undesirable delay.

For these cases, IPA 25 provides a Zero-Latency Mode (0 ms), where:

- Realtime Oversampling is disabled (though rendering oversampling remains available)
- Look-ahead is disabled (but look-behind compression is still possible)
- The Limiter mode of the Clip/Limiter module is unavailable (though Clipping remains active)

Note: Avoid switching latency modes while playback is active in your DAW, as this may disrupt latency compensation and cause phasing or delay issues.

Stepped Knobs

These options allow limiting the rotary knobs in the interface to a set number of positions. In "Full Steps" mode, the number of steps corresponds to the number of yellow dots around the knobs. In "Half Steps" mode, some intermediate steps are also added. These stepped modes recreate the ergonomics of certain hardware units (particularly from the mastering world), where settings can be easily reproduced identically.

Note: even in stepped mode, it's still possible to access intermediate knob values by using "fine" edit mode, meaning by using the right mouse button or holding down the Ctrl key (Windows) or Cmd key (Mac) during adjustment.

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

General Layout of the Interface



The IPA 25 control panel

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



Control Rack

This rack contains the main controls used to set up the compressor.

It is organized into 6 sections: COMPRESSOR - CLIP/LIMITER - OUTPUT - OPTIONS - SIDECHAIN - Metering and master bypass section.

COMPRESSOR Section



Compressor Section

Compressor On/Off

You can activate or deactivate the compression module alone using an On/Off button located to the left of the section title.

⏻ COMPRESSOR

Compressor On button

THRESH

This knob sets the compression threshold. Signals above this level (in dBFS) will be compressed, while signals below will stay unaffected. Turning the knob clockwise lowers the threshold, increasing the amount of compression.



Threshold knob

RATIO

This knob controls how much compression is applied when the signal exceeds the threshold. A higher ratio means more aggressive compression.

The extreme values range from **1:1** (no compression, where the signal passes completely unaffected) to **infinity:1** (hard limiting, which completely prevents any signal from exceeding the threshold). Between these extremes, typical ratios (2:1, 4:1, etc.) offer varying degrees of dynamic control, from subtle smoothing to punchy leveling.

Note: Higher ratios require careful threshold adjustment to avoid unnatural over-compression.



Ratio knob

KNEE

This button selects the knee, or the transition curve into compression.

- **Hard:** compression starts abruptly as soon as the threshold is exceeded, ideal for a clear and punchy compression effect.
- **Mid:** a good balance between precision and smoothness. Suitable for versatile compression.
- **Soft:** compression starts softly before reaching the threshold. Ideal for vocals and natural dynamic control.



Knee selection button

ATTACK

This knob sets the compression attack time. The shorter the attack time, the more the compressor reacts to percussive elements. The attack time ranges approximately from 30 microseconds (0.03ms) to 30 milliseconds.



Attack knob

RELEASE

This knob sets the compression release time. The release time ranges approximately from 50 milliseconds (0.05s) to 3 seconds.

Note on Release Behavior: The original hardware featured two release controls: a stepped knob with 6 fixed time values (0.05s, 0.1s, 0.2s, 0.5s, 1s, 2s) and a separate variable mode for continuous adjustment. In Pulsar IPA 25, we've unified these into a single knob that defaults to variable control while preserving access to the original stepped values. Simply click on the time labels around the knob to snap to these precise, hardware-authentic release times.



Release knob

MAKEUP / AUTO

The MAKEUP knob controls the output level of the compressor to compensate for the gain reduction caused by compression.

If the AUTO button is engaged, the MAKEUP knob is automatically adjusted to provide automatic compensation, calculated based on THRESHOLD and RATIO. This compensation is optimal for a source calibrated at -12 dBFS.



Makeup knob and Auto button

MIX

This knob adjusts the balance between the compressed signal (100%) and the original signal (0%). It enables parallel compression for enhanced punch while preserving dynamics. The exact behavior depends on the Clip/Limiter routing setting:

When ROUTING is set to **PRE MIX** or **PRE COMP** : MIX blends between the unprocessed signal (dry) and the fully processed signal (including compression and clipping/limiting effects).

When ROUTING is set to **POST MIX** : MIX blends between the unprocessed signal (dry) and compressed signal (wet), with the resulting mix then being processed by the Clip/Limiter module.

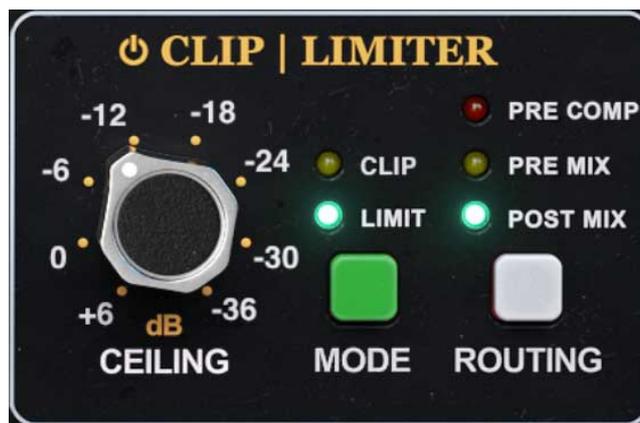


Mix knob

CLIP/LIMITER Section

We added a Clip/Limiter module to the IPA 25. This module controls the fastest transients. It offers two modes: Clipping for soft and musical saturation, and Limiting for transparent but firm peak control.

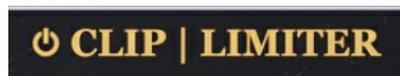
Depending on the hardware or music type, the Clipper may add slight saturation and "fatness", while the Limiter is better suited for invisible limiting, especially in mastering or bus contexts.



Clip/Limiter Section

Clip/Limiter Module On/Off

You can activate or deactivate the Clip/Limiter module alone using an On/Off button located to the left of the section title.



Clip/Limiter On button

CEILING

This knob sets the ceiling of the Clipper or Limiter. This parameter defines the maximum level the processed signal can reach at the output of the Clip/Limiter section. It acts as an upper barrier: any signal exceeding this level will be either clipped (Clip mode) or limited more transparently (Limiter mode).

This setting is particularly useful to control output headroom, avoid unwanted distortion, or match

the final level to a target standard (e.g. -1 dBFS for digital distribution).



Ceiling knob

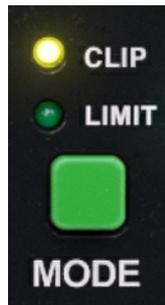
MODE

This button selects the operating mode of the section: Clipper or Limiter.

- In Clipper mode, peaks above the threshold are clipped in a controlled manner, introducing slight saturation that can add character and energy to the signal.
- In Limiter mode, a limiter/maximizer reduces peaks while minimizing audible distortion. This mode is ideal to preserve sound integrity while increasing perceived loudness.

Note: *The limiter requires some latency to operate. If the zero-latency mode is enabled in the options ("...") menu, then the limiter cannot be used and only the clipper is available.*

The choice between the two modes depends on the source type and desired result — creative saturation or subtle level control.

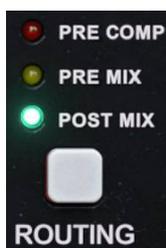


Mode selection button

ROUTING

This selector chooses the routing of the Clip/Limiter module, which can be placed at three different points in the processing chain:

- Before the compressor (PRE COMP): useful to tame very fast transients before they trigger excessive compression, resulting in more stable and less "pumping" compression.
- After the compressor, but before the dry/wet mix (PRE MIX): ideal to control peaks generated by the compressor itself while maintaining a natural parallel mix.
- After the compressor and dry/wet mix (POST MIX): the clip/limiter acts as a final safety net at the output to prevent level overshoot or to precisely shape the overall processed signal.



Routing selection button

OUTPUT Section

This section only contains the master output volume knob of the plugin.



Output volume knob

OPTIONS Section



Options Section

EDIT Button

Selects which channel is currently being edited:

- In L/R mode: toggles between Left and Right channels.
- In M/S mode: toggles between Mid and Side channels.

This lets you adjust each channel independently, using the same control set. The selected channel is displayed in the interface, and all changes apply only to it until changed again.

Note: the EDIT button intentionally doesn't toggle back to linked (L+R or M+S) mode because returning to linked processing would discard any independent channel settings. To go back to linked mode, click the "L+R" or "M+S" label or LED: the interface switches to linked mode, the currently visible settings will be applied to both channels, and any different settings made to the other channel will be lost.



EDIT button

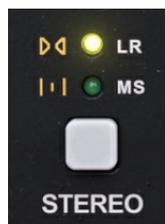
STEREO Button

Switches between the two primary stereo processing modes:

- **L/R Mode** (Left/Right):
 - Controls visible when EDIT is set to L process the left signal
 - Controls visible when EDIT is set to R process the right signal
- **M/S Mode** (Mid/Side):
 - Controls visible when EDIT is set to M process the mono (center) component
 - Controls visible when EDIT is set to S process the stereo width/difference component

Notes:

- *Only the Compressor section parameters can have independent channel values - all other sections remain linked between channels*
- *This control is not available on mono tracks*
- *Channel-specific adjustments are preserved when switching modes*

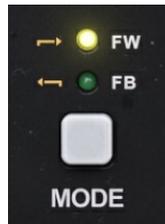


STEREO button

MODE Button

Chooses between Feedforward and Feedback detection topologies:

- **Feedforward:** the input signal is used to control gain reduction. This mode is generally more responsive and aggressive.
- **Feedback:** the compressed output signal is used as the detection signal, producing slower, softer, and more musical compression.



MODE button

VCA Button and DRIVE Control

The VCA button lets you choose between different VCA circuits, expanding the compressor's sonic palette. VCA means "Voltage Controlled Amplifier" and it is the electronic component handling gain reduction in the compressor.

- **V-Mu** : Reproduces a "Push-Pull" amplifier circuit, delivering sound characteristics similar to variable-bias compressors.
- **IPA 25** : Reproduces the original hardware's circuitry.
- **N-DIODE** : Diode-bridge VCA that produces more harmonics than the original version, adding extra presence.

Note: *The VCA selection only affects how gain reduction is applied to the audio signal. All other emulated circuits (including timings, ratio, and knee) remain faithful to the original hardware's design, maintaining their original flexibility.*

The DRIVE knob adjusts the VCA calibration, controlling its harmonic distortion characteristics.



VCA and DRIVE controls

SIDECCHAIN Section

The Sidechain section enables precise customization of the compression behavior by modulating the sidechain signal through several key parameters.

Note: *Additional sidechain parameters (including EQ, look-ahead, and external sidechain) are available in the left portion of the upper rack.*



Sidechain Section

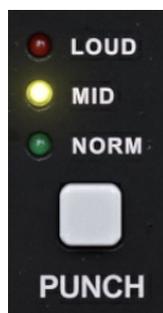
PUNCH Button

Selects between fixed sidechain filters that emphasize high frequencies, making compression more reactive to bright transients while preserving low-end dynamics. Particularly useful for tightening mixes, controlling cymbals, or adding punch.

- **NORM** : Flat response (no filtering) - standard compression behavior
- **MID** : Subtle high-end sensitivity (slight bass attenuation + treble boost) for gentle transient control
- **LOUD** : Aggressive bright signal response (~3dB/octave slope) for punchy, dynamic compression

The PUNCH filter's unique 3dB/octave slope naturally guides compression toward a pink noise response - closer to real-world mix spectra than flat sidechain processing. While the full sidechain EQ (top rack) allows arbitrary adjustments, this fixed filter provides instant spectral shaping optimized for musical results.

Note: *The chosen filter's frequency response is visible in the sidechain EQ module, in the top rack. The Punch filter, however, cannot be heard in Sidechain Listen mode, as in FB mode it is processed inside the feedback loop.*



PUNCH button

LINK Knob

This knob adjusts the stereo linking amount between channels. At 0% (IND), each channel's gain reduction is calculated independently. At 100%, both channels receive identical gain reduction, ensuring consistent stereo balance. Intermediate values blend these behaviors for optimal dynamic control.

Practical example: With a fully panned hi-hat and 0% linking, only the hat's channel would compress. At 100% linking, both channels would compress equally - preventing stereo image shifts while potentially over-compressing the silent channel. A middle setting would provide a compromise: the hat's channel would be fully compressed while the other channel would be less compressed.

Note: Even at 100% linking, if a link filter is active (selected via the FILTER button), the gain reduction may be different across channels, depending on their spectral content.

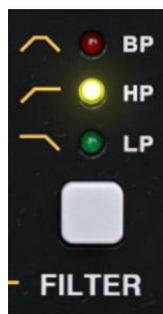


LINK knob

FILTER Button

Selects the frequency filter applied to the control signal used for channel linking (controlled by the LINK knob):

- **HP (High Pass)** : Removes low frequencies from the link control signal, so channels link primarily on transient information in the mid/high range.
- **LP (Low Pass)** : Removes high frequencies, making the link respond mainly to sustained low-frequency content.
- **BP (Band Pass)** : Engages both the high-pass and low-pass filters, allowing channel linking to react only to a selected mid-frequency band while ignoring low and high extremes.



FILTER button

Analog Metering Section

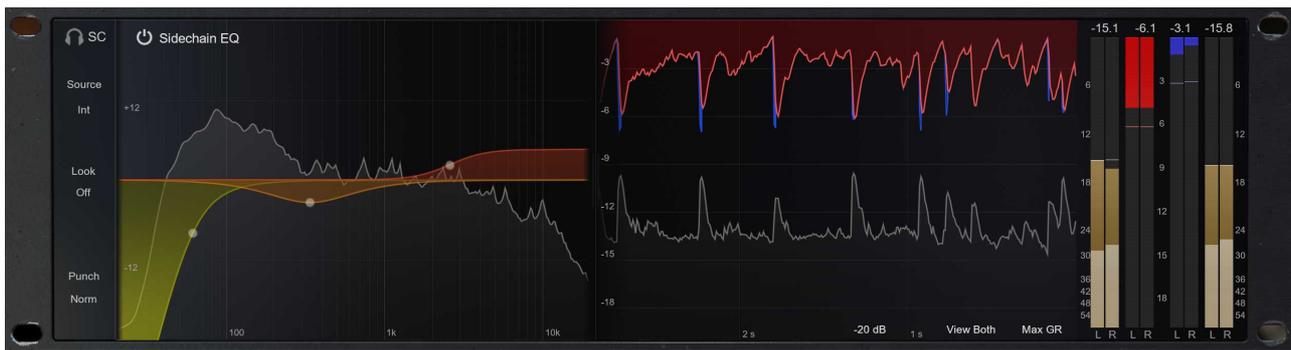
This section displays the compressor's gain reduction in L/R or M/S (depending on the selected mode).

The LEDs at the top of the section light up to indicate Clip/Limiter gain reduction (in L/R).



Analog metering section

Visualization and Advanced Sidechain Features Rack



Visualization rack

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

Pulsar IPA 25 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 by ear.

Note: While the Punch feature is essentially a fixed sidechain EQ, its effect cannot be heard in Sidechain Listen mode. This is because the Punch module is placed differently in the signal path than the 4-band sidechain EQ: in feedback mode, Punch is processed within the feedback loop itself. As a result, its action cannot be monitored on the sidechain signal independently of the compression.



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.

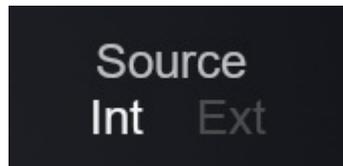
If "EXT" is selected, an external audio signal is used as sidechain source. It is then necessary to assign a sidechain (external sidechain) channel in your DAW — please refer to your DAW manual for routing details.

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

This button selects the sidechain signal source for compression detection:

- **INT** : Uses the plugin's audio input as sidechain source (normal operation)
- **EXT** : Uses an external audio track from your DAW (like a kick drum track) routed to the plugin's sidechain input. This allows the compressor to react to a different signal than the one being processed (for example, for making bass duck when kick hits). *Note: Routing methods vary by DAW - consult your software's manual for sidechain setup instructions.*

When using external sidechain, the **Ext. SC Gain** slider lets you adjust the incoming sidechain signal level, since external sources often have different volumes than your processed signal (either too loud or too quiet to properly trigger compression).



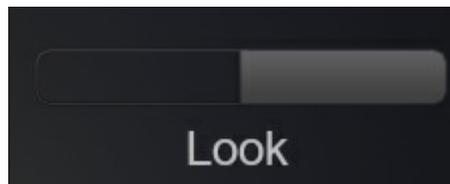
Sidechain source selector

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 compressed 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 example to be limited later by the 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.*

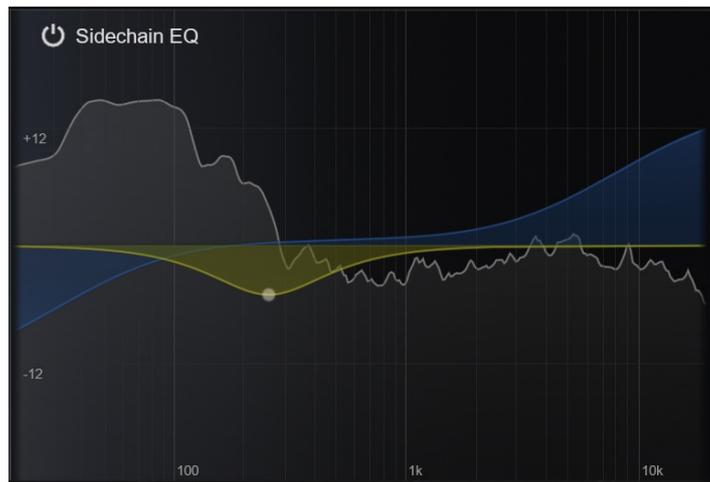


Look-ahead / look-behind switch and slider

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.



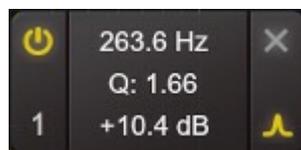
Sidechain EQ display

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. In Band Solo, "Sidechain Listen" is temporarily enabled so you hear only the filtered detection signal, and filter type is temporarily changed to hear only 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 occupied by the metering section, featuring a large scrolling graph showing both the input volume envelope and gain reduction over time, along with vertical meters for input, output and GR (Gain Reduction) levels.



Metering display

Scrolling Curves

The prominent left-side display shows the time-based input volume envelope (in gray) alongside gain reduction curves (in red/blue), allowing you to observe the relationship between audio peaks and compression response, including the effect of attack and release settings.

Three configuration buttons at the bottom right of the display let you customize the metering view:

- **GR Meter Range** : Adjusts the vertical scale of GR meters. The default 40dB setting offers the most versatile range for general use, while 20dB/10dB/5dB options provide progressive zoom for detailed observation of compression activity (particularly useful in mastering).
- **GR Meter Source** : Selects which gain reduction to display:
 - "Compressor": Red curve showing only compression GR
 - "Clip/Limiter": Blue curve showing only clipping/limiting GR
 - "View Both" (default): Stacked red+blue curves showing total GR from both modules
- **Displayed GR Channel** : Chooses channel visualization:
 - "Max GR" (default): In this mode, the display shows a single curve representing the highest reduction value across all channels at any given moment. This view is compact and easy to interpret, making it suitable for most common scenarios where a quick overview of the overall reduction is sufficient.
 - "Dual GR": This mode provides a split display, showing the gain reduction occurring on each channel independently. The exact channel pair depends on context:
 - When the compressor is in Left/Right mode, or when you are viewing the clip/limiter GR, the two curves represent the Left and Right channels.
 - When viewing the compressor's GR and the compressor is in Mid/Side mode, the two curves represent the Mid and Side channels.

This view is particularly useful for stereo analysis, as it allows you to monitor how the compressor behaves on each channel.

Note: *The Dual GR option is not available in M/S mode when both compressor and clip/limiter gain reduction are displayed at the same time.*

Input and Output Levels

The vertical indicators on the right show input and output levels for each channel. These include RMS average level meters (lighter), peak level meters (darker), a "Peak Hold" bar that holds the highest peak for several seconds, and text showing the dBFS level of the "Peak Hold".

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

Gain Reduction Levels

The plug-in provides four vertical meters for gain reduction: two meters (one per channel) for the compressor, and two meters (one per channel) for the clip/limiter. The compressor meters are displayed in red, while the clip/limiter meters are displayed in blue. A "GR Hold" bar, with the corresponding dB value displayed above, shows the highest GR level reached over a short period for each processing stage.

Note: *the compressor GR meters show the left and right channels when the compressor is in L/R mode, and the mid and side channels when it is in M/S mode. The clip/limiter GR meters always show the left and right channels, since this module never operates in M/S mode.*

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: OpenGL 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

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