Pulsar 1178
User Manual

Table of contents

Introduction

This manual describes the features and operation of the Pulsar 1178 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

FET compressors

FET compressors take their name from an electronic component called a Field Effect Transistor (FET), which is at the heart of the gain reduction system. A FET compressor generally offers a much faster action than other compression technologies (vari-mu, for example) and lends a rather warm and rich sonic character to an audio track.

FET compressors do not usually have a threshold control, as the threshold is fixed. The compression level is adjusted using the input gain of the compressor.

The 1178, released in 1980, is a stereo version of the legendary 1176, which is present in studios all over the world. The 1178 does not have an input transformer and the class-A circuitry present in the early versions of the 1176 compression circuit. It is instead based on a 1176 Rev H, which gives it less warmth, but more transparency and therefore more versatility, while keeping the legendary punch.

This versatility will enable you to use our 1178 as a character compressor on percussive elements (bass drum, snare drum), instruments such as guitar or bass, vocals (where it excels), busses (drums, guitars) or even on your master channel.

Note that, as on the 1176, it integrates the famous "All-buttons" mode or "British" mode, which enables compression ratio close to a limiter, and most importantly, the characteristic musical distortion that goes with it.

The Pulsar 1178 offers:

- The sound of a legendary machine, present in studios all over the world

- Musical compression, whether placed on an instrumental track or on a bus

- Unparalleled warmth and presence

- Advanced sidechain options (external sidechain, lookahead, EQ, etc)

- An output stage allowing addition of a color ranging from a slight warm-up to a destructive clip

- The advantages of "organic" analog sound in conjunction with the advantages of the digital world (presets, parameter automation, etc)

Quick start

Installation

Pulsar 1178 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:

- 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)

- 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 Pulsar 1178 on a track of your choice in your DAW. A good starting point is to load a basic preset corresponding to the type of the track (voice, guitar, bass, drums...). From there:

- Start playback and observe the gain reduction being applied, looking at either the analog VU meters or the modern red GR indicators in the upper-right corner

- Adjust the input gain to get the level of gain reduction suggested in the preset name. Hold down the Shift key on your keyboard while adjusting the Input Gain, and the Output Gain will automatically compensate

- Adjust the Attack and Release speeds using the respective knobs. The faster the Attack, the more the transients will be attenuated, and the faster the Release, the more the sound between the transients will be emphasized. As you play back, observe the effects of these settings on the red gain reduction curve in the upper screen

- Adjust the output gain so that the volume at the output of the compressor is the same as at the input. This can be done by ear, using the "Power" switch to temporarily disable the effect of the compressor, or by matching the input and output level indicators located in the top right of the upper screen. After performing this step, it will be possible to evaluate the effect of the compressor using the "Power" switch, without being misled by volume differences

- If necessary, adjust the amount of compressed and uncompressed sound using the "Mix" knob

This way you can quickly review many of the available factory presets for inspiration without getting into technical considerations.

The user interface

The stereo 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 two racks: the compressor control rack, and the visualization and advanced detection functions rack

Note that in all Pulsar Audio plug-ins, you will find a resizing control in the bottom right corner of the plugin interface.

If the 1178 is inserted on a mono track (in DAWs that support it), it will appear in a more compact form:

The mono user interface

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%):
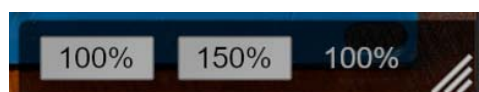
Resizing window

The Toolbar

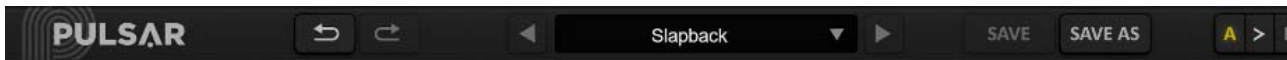Located at the top of the plugin interface, it contains all the functions relating to parameters, presets, communication with Pulsar Audio, etc.

The toolbar

Undo / Redo

The 2 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
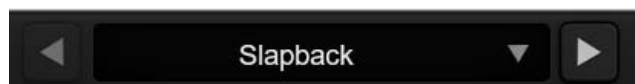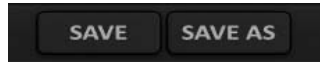

The preset selection area

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.
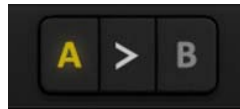
A, B and Copy buttons

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.

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

Rack visibility button

This button enables you to resize the plugin's interface, hiding some parts of it. It is thus possible to display only the main rack, if additional features such as EQ and GR visualization are not required, or on the contrary to display only the visualization rack, for example to keep an eye on the plugin action when the settings are not going to change.

The rack visibility button

The control panel

The 1178's control panel

The control panel of the 1178 is inspired by a famous hardware machine, with a few additions to broaden the sound palette.

This panel consists of two racks:

- The compressor's control rack
- The rack with the visualization tools and advanced detection functions

The control rack

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

In the stereo version, the Input, Output, Attack and Release parameters are duplicated. They can be set independently per channel, i.e. it is possible to have different settings between the left and right channels (in L/R mode), or between the "mid" and "side" channels (in M/S mode).

Power

The switch on the right side of the interface is used to activate (ON position) and deactivate (OFF/bypass position) the effect.

Power switch

Input gain

This knob is used to adjust the input gain of the signal in one of the channels of the compressor. The higher this gain, the greater the compression will be.

Note: If the Shift key is pressed when adjusting this knob, the output gain knob will be adjusted in the opposite direction.

Input Gain knob

Output gain

This knob is used to adjust the output level of one of the compressor channels, in order to compensate the effect of the input gain and the gain reduction due to compression.

Note: If the Shift key is pressed when adjusting this knob, the input gain knob will be adjusted in the opposite direction.

Output Gain knob

Attack

This knob is used to set the attack speed of the compression. The faster the attack is, the more the compressor is triggered on the percussive elements. The attack time ranges from about 800 microseconds (left, slow attack) to 20 microseconds (right, fast attack).

Note: Contrary to the vast majority of compressors on the market, the position to the left of the control (0) corresponds here to the longest attack time, and the position to the right (7) to the shortest attack time.

Attack knob

Release

This knob is used to set the compression release speed, i.e. the speed at which the compressor stops reducing the gain after the input volume has fallen below the trigger threshold. The lower the release time, the more audible the pumping effect will be. This release time ranges from about 1.1 seconds (left, slow release) to 50 milliseconds (right, fast release).

Note: Contrary to the vast majority of compressors on the market, the position to the left of the control (0) corresponds here to the longest release time, and the position to the right (7) to the shortest release time.

Release knob

Controls Link switch

Only visible on a stereo track.

This switch allows you to link the settings of the two channels. If it is activated, each setting that is made on channel 1 is also made on channel 2 (and vice versa).

Note: the relative offset between the two channels is maintained when the channels are linked. To reset the parameters of both channels to the same settings, it is therefore necessary to disable the link beforehand. You can reset any parameter to its default value by double-clicking or ALT-clicking.

Controls link switch

Listen
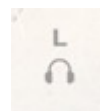
Only visible on a stereo track.

These headphone-shaped switches enable you to listen to only one channel, which can come in handy, especially in M/S mode, to fine-tune the compression of the channels individually.

Left channel listen switch

Sidechain Link

Only visible on a stereo track.

This selector switch is used to link the "sidechains" (detection circuits) of the two channels.

If set to "unlink" (center), the two detection channels are separated, and the gain reduction is set independently per channel. On a stereo bus, the level will be more homogeneous, and the dynamic movement more musical, but the stereo image may be affected for high GR levels.

If set to "link" (up), the two detection channels are mixed, making the gain reduction the same for both channels. This allows, for example, to keep the stereo coherence of a master bus, in case of strong compression.

If set to "invert" (down), the two detection channels are separated, and the gain reduction is independent for the both channels, however, the Left (or Mid) channel is compressed according to the audio of the Right (or Side) channel. The resulting dynamic movement can be very interesting and musical on stereo busses with independent panned sources (guitar busses, with a guitar on the left and a guitar on the right).

Sidechain Link switch

L-R / M-S Switch

Only visible on a stereo track.

This switch allows you to select the stereo routing mode:

- L/R corresponds to the traditional stereo mode (left / right channels)

- M/S corresponds to Mid / Side mode. In this mode, the Mid (or "center") channel contains only the audio that's common between left and right channels; and the Side (or "stereo") channel contains audio that's different between left and right channels)

The M/S mode allows you to compress with different settings the sounds that are panned more at the center and those that are panned more at the sides. It is therefore recommended to set the "Sidechain Link" switch to the "Unlink" position, as well as to disable the link of the controls.

L-R / M-S switch

Mix

This knob is used to blend between the amount of compressed signal (full at position 100) and the original, dry signal (full at position 0). This enables to do parallel compression, which can be used, for example, to give a track the color of the compression, while preserving the transients.

Mix knob

Ratio

The selection buttons located on the left of the VU meters allow you to select the compression ratio (amount of compression to be applied to the signal that exceeds the threshold).

The ALL mode corresponds to the famous "all-buttons-in" or "British mode", characterized by a high ratio, a more saturated sound, and a change in attack and release behavior.

Ratio buttons

Saturation

In order to make the 1178 even more versatile we have added an extra saturation stage, which complements the very slight saturation present in the original.

This stage enables a slight coloration at the compressor's output. This color will be more or less perceptible depending on the sound source and the different settings (input gain, attack and release speeds, ratio).

In the "Dry" position, this additional saturation stage is disabled. The other four positions reproduce various saturation circuits, set to have a subtle but interesting effect that widens the sound palette of the compressor.
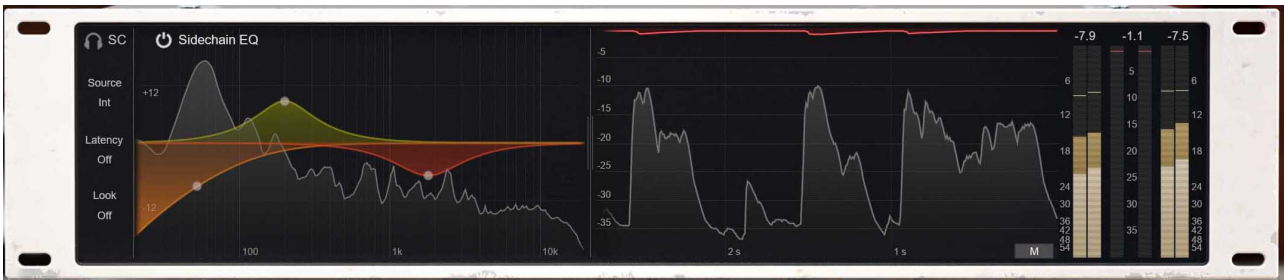

Saturation buttons

Saturation calibration screw

This small screw-shaped knob is a calibration control for the saturation stage, that enables to just slightly color the sound and catch the transients, or at the opposite to obtain a very audible or even destructive saturation.

Saturation Calibration screw

The visualization and advanced sidechain rack

Advanced detection functions

In the top rack of the plugin, you can visualize important information such as the GR (gain reduction) metering; and make various adjustments concerning the detection signal or sidechain, which is the (non-audible) signal used to calculate the gain reduction.

Pulsar 1178 is equipped with comprehensive features for tailoring the response of its compression.

Note that while the word "Sidechain" is often used to refer to an external signal driving a compressor, the default, internal detection is also known as the sidechain. It's more helpful to think of the word "sidechain" meaning "detection circuit", in most cases.

Sidechain Listen

This headphone-shaped button allows you to listen to the sidechain signal, for example to be able to check the presence of the correct signal in case of external sidechain, or to help refine the sidechain EQ settings by ear.
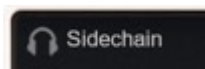

Sidechain Listen Button

Source (INT / EXT)

This button is used to select the detection source. In "INT" mode, the compressor will operate in the usual way, i.e. using the audio input as the detection source.

If "EXT" is selected, an external audio signal will be used as detection source. It will then be necessary to assign a sidechain (external detection) channel in your DAW – please refer to your DAW's manual for more information on routing.

In the case of an external sidechain, it is possible to adjust the input gain (and thus the amount of compression) using the "Ext. SC Gain" slider.
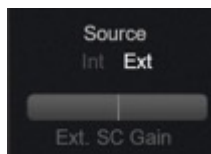
*The sidechain source switch and gain slider*

Look (look-ahead, look-behind)

Even though Pulsar 1178's FET-style attack times are very quick to begin with, you can increase the speed of detection using Lookahead. This shifts the entire detection signal earlier in time, so it's possible for the compressor to even react to transients before they actually happen.

The Lookahead Latency slider can introduce a positive or negative delay to the detection channel. Setting the slider to the left enters look-ahead mode, meaning the compressor will activate earlier – in anticipation of the signal. This is useful for compressing or limiting loud and fast transients, and can be set from 20ms to 0ms. On the other hand, moving the slider to the right of its central position activates look-behind mode, delaying the compression and allowing transients to pass through unprocessed. Look-behind can be set from 0ms to 10ms.

The switch above the slider allows to deactivate look-ahead or look-behind completely. This may be desirable when zero-latency operation is required. Indeed, when this switch is set to "On", look-ahead becomes possible but the plugin introduces exactly 20ms of latency. This latency is automatically compensated by your DAW, so you will not have to worry about it in most cases. However, it may be necessary to remove it in tracking or live situations.

Thus, when this switch is turned off, the left part of the "Look" knob will be disabled. The right part, corresponding to the look behind function, remains usable because it does not need latency to work.
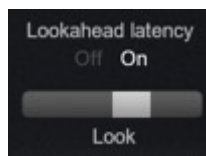
Look-ahead / Look-behind switch and slider

Sidechain EQ

This section enables the detection signal to be filtered using a parametric equalizer that can be edited on the curve. A display of the sidechain signal spectrum is also present in the background, in order to facilitate the visualization of important frequencies.

The switch at the top left enables / disables the EQ. The band parameters (frequency / Q / gain / filter shape) can be controlled directly on the curve, or from the dialog box that appears when the mouse passes over a band.
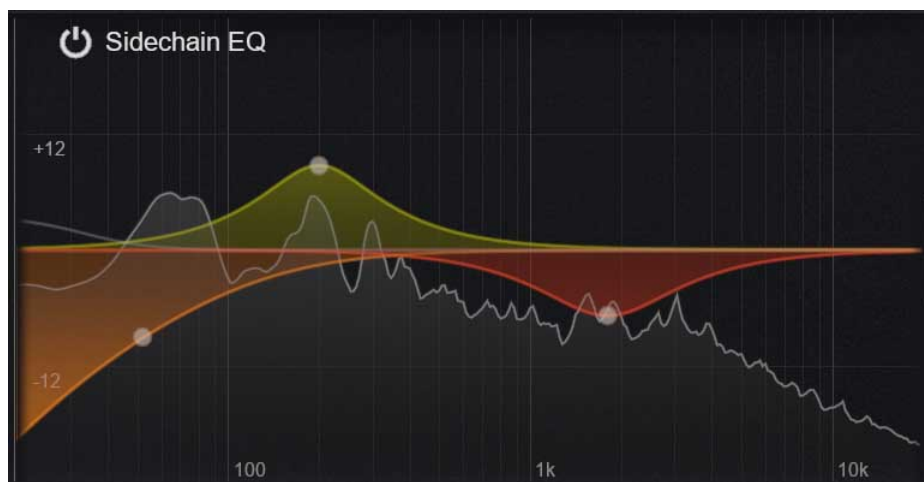
Sidechain EQ screen

To add a band, move the mouse over the EQ surface. A gray line then allows you to preview the band that will be created when you click. Depending on the position of the mouse at the time of the click, different types of bands are created:

- In the upper left corner, a low-shelf filter

- In the bottom left corner, a high-pass filter

- In the middle, a bell filter

- On the top right, a high-shelf filter

- In the bottom right corner, a low-pass filter

Different actions can then be performed using the mouse:

- In the same way as for all other controls, hold down the **Ctrl** key or use the **right mouse button** to make fine adjustments

- Click on a band while holding down the **Alt** key on your keyboard to delete a band

- Hold the **Shift** key on your keyboard while moving a band to enable the Band Solo feature, which allows you to listen to the effect of that band alone on the sidechain signal. In Band Solo mode, the "Sidechain Listen" mode is temporarily activated, i.e. you no longer hear the audio processed by the compressor, but only the detection signal filtered by the EQ band being listened to. The filter type is also temporarily changed, so that you will only hear the frequencies of the spectrum affected by the band

- Use the mouse wheel when the mouse is over a band to change the Q of the band (in the case of bell and shelving filters) or to change the slope of the band (in the case of high-pass and low-pass filters)
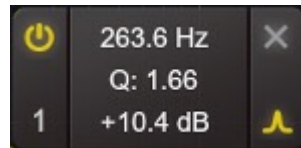
The EQ band parameters window

The band settings window appears when a band is hovered with the mouse.

- The top left button enables or disables the band

- The number at the bottom left indicates the band number (from 1 to 4), which can be useful to identify the band parameters, in case of automation

- The cross at the top right enables to delete the band

- The bottom right button allows you to change the shape of the filter

- It is possible to click on the frequency, gain, Q and slope values to enter values manually

Filtering the sidechain signal can be useful to adjust the compression according to the content:

- Cutting the low-end on a master bus will enable not to trigger the gain reduction on bass drum hits, and thus to have a more discrete pumping

- Lowering the upper frequencies on an overhead track or a drum bus allows less pumping on each crash cymbal shot

- A bell filter will allow you to selectively choose to more or less trigger the compressor on an element of the spectrum (a snare drum or a voice for example)

Metering

The right part of the rack is occupied by the metering section, which provides the input, output, and GR (Gain Reduction) levels, the latter also appearing in the form of a "GR versus time" graph (scrolling view).

Metering screen

Gain Reduction (GR) levels

The vertical red indicators ranging from 0 dB to -40 dB measure the instantaneous level of Gain Reduction applied by the compressor to each channel (left/right, or mid/side in MS mode). A "GR Hold" bar, whose corresponding dB value is displayed above the indicator, indicates a "peak" GR level that has been reached in a short period of time.

Input and output levels

The vertical indicators to the left and right of the GR indicators are used to measure the input and output levels of each channel (left/right, or mid/side in MS mode). They include RMS average level meters (lighter), peak level meters (darker), a peak hold bar that holds the highest peak level for a few seconds, and a text that displays the peak hold level in dBFS.

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

Scrolling curves

Most of the metering section is dedicated to the temporal display of the gain reduction level (in red) and the input volume envelope (in gray). The simultaneous display of these two scrolling curves allows you to observe the relationship between the peaks that trigger the compressor, and the resulting gain reduction, as well as the effect of the attack and release settings.

The scale of the gain reduction, which ranges from 0 dB to -40 dB, is displayed on the left. The scale of the volume envelope ranges from 0 dB to -60 dB and the corresponding graduations are not displayed, however the scale corresponds approximately to that of the input level meters.

Stereo Mode button

The small button in the lower right corner, visible only on stereo instances, allows you to change the way the volume envelope and gain reduction of each channel are displayed in the scrolling views:

- In "M" mode, which is the default mode, the volume and gain reduction curves displayed are those of the mix between the left and right channel

- In "L" and "R" modes (or "M" and "S" in M/S mode), only the volume and gain reduction curves of the corresponding channel are displayed

- In "LR-1" mode (or "MS-1" in M/S mode), the volume and gain reduction curves of each channel are displayed separately in the upper and lower part of the screen respectively

- In "LR-2" mode (or "MS-2" in M/S mode), only one volume envelope curve is visible, corresponding to the mix between the left and right channels. On the other hand, the gain reduction curves of each channel are displayed simultaneously with different colors: the first channel (L or M) in red and the second channel (R or S) in green

Minimum Configuration

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

Available formats:

- VST 2.4 (Windows: 32/64-bit, Mac: 64-bit only)

- VST 3 (Windows: 32/64-bit, Mac: 64-bit only)

- AAX (Windows: 32/64-bit, Mac: 64-bit only)

- Audio Unit (64-bit).

Windows

- CPU: Intel Core i3 / i5 / i7 / Xeon

- Memory: 4 GB RAM / 1 GB free disk space

- Operating system: Windows 7 and higher

- GPU: OpenGL 2.0 compatible GPU with up-to-date drivers

- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

MacOS

- CPU: Intel Core i3 / i5 / i7 / Xeon / Apple Silicon (M1)

- Memory: 4 GB RAM / 1 GB free disk space

- Operating system: 10.9 and higher

- GPU: OpenGL 2.0 compatible GPU with up-to-date drivers

- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

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