Pulsar Vocal Studio

User Manual

www.pulsar.audio

Table of contents

Introduction

This manual describes the features and operation of the Pulsar Vocal Studio 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

Voice Processing in Mixing

Vocal mixing is a crucial step in music production, where the voice, a central element of many tracks, is shaped to blend harmoniously into the overall mix. Whether you are working on a pop song, a hip-hop track, or an acoustic project, vocal mixing requires both a keen ear and technical mastery of audio tools. A good vocal mix enhances the emotion and intent of the performer while ensuring optimal clarity and presence within the track's context.

This process requires a wide range of tools, including:

- Equalization (EQ): to adjust frequencies and avoid conflicts with other elements in the mix.
- Compression: to control the dynamics of the voice and ensure a consistent presence.
- Effects (reverb, delay, etc.): to add depth and space to the vocals.
- And many more...

And, of course, the skills to use them: vocal mixing is both an art and a science, requiring an understanding of the tools, musical sensitivity, and meticulous attention to detail.

Pulsar Vocal Studio simplifies the technical aspect of this process by reducing the numerous parameters of all these tools to just a few easy-to-understand and adjustable settings, allowing users to achieve a flawless result quickly while maintaining high sound quality.

Quick start

Installation

Pulsar Vocal Studio 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 .

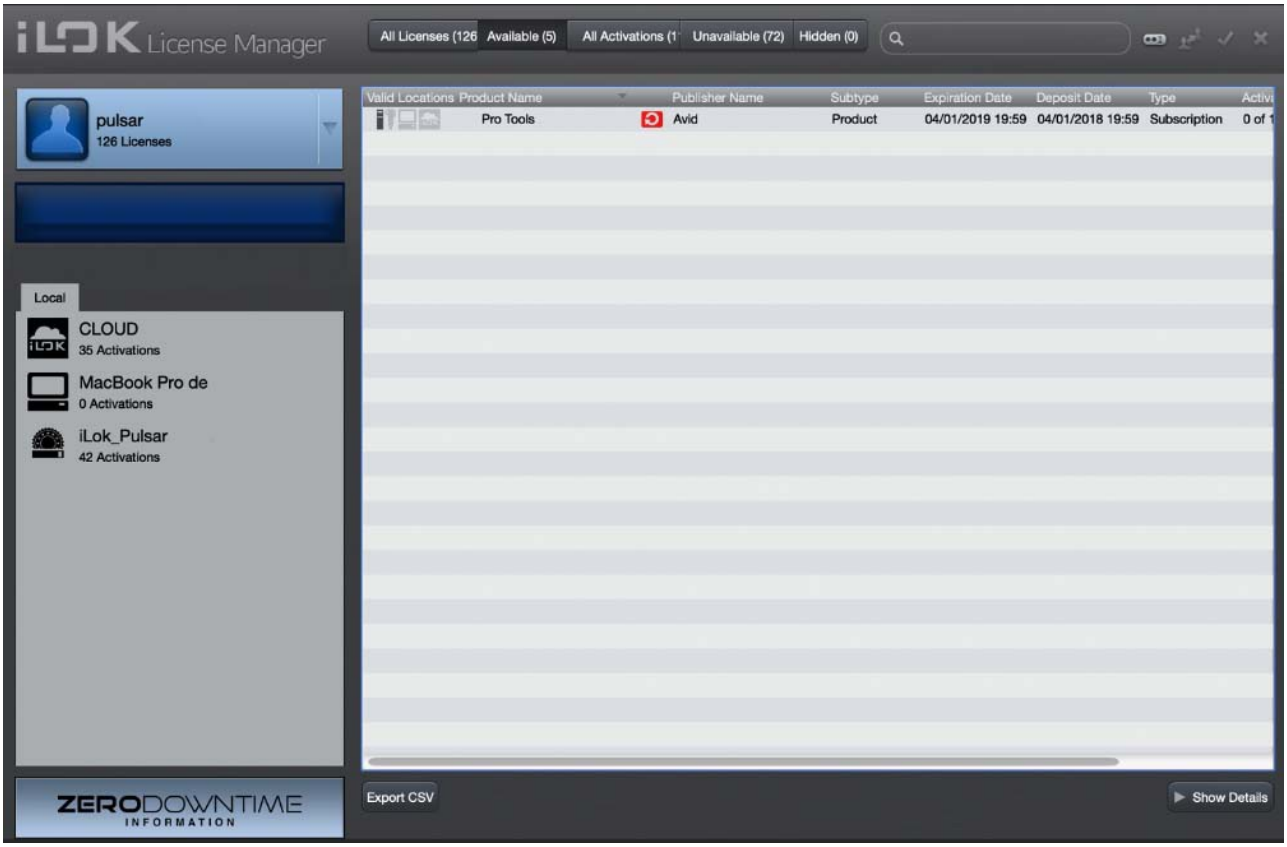
You can choose between three activation methods:

- 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 **Pulsar Vocal Studio** onto a vocal track in your DAW. A good starting point is to load a basic preset that matches the aesthetic you're aiming for. From there:

- Adjust the **sensitivity** knob during playback to bring the LED ring gauge to "GOOD"
- Adjust the **noise gate** so that it closes (red LED) during non-sung passages
- Adjust the **de-esser** to reduce harsh sibilance
- Adjust the **compressor** until you achieve a more controlled and "produced" sound
- Select an **EQ preset** and adjust the **Amount** knob until you achieve a balanced result—not too muddy, not too aggressive, and in line with the track's aesthetic.
- Add **delay** , **reverb** , and possibly a **special effect** to enhance the vocal track and position it in the mix

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

The User Interface

The User Interface

The user interface consists of four distinct sections:

- The toolbar, common to all Pulsar Audio plug-ins (at the top)
- The control rack, specific to Pulsar Vocal Studio, which itself consists of three racks:
  - The top rack, featuring calibration controls (*Sensitivity*), *Drive* section, and dynamic processing tools: compressor, noise gate, de-esser.
  - The middle rack, containing the EQ and spectral correction tools (high-pass, low-pass, focus band, etc).
  - The bottom rack, with special effects and time-based effects (delay, reverb).

Note that in the bottom right corner of the plugin interface, you will find a resize control, common to 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.

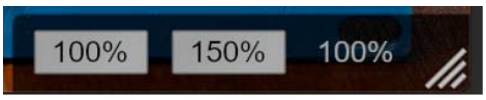
Using the GUI resize control

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.

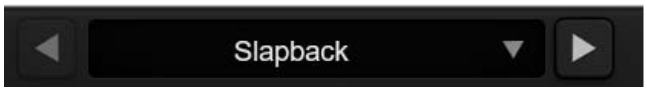
Note: right-clicking on any of these buttons provides access to the list of stored operations, allowing you to undo or redo multiple actions at once.

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.

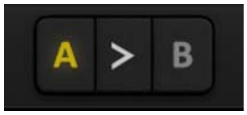
Save and Save As buttons

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

Menu Button

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

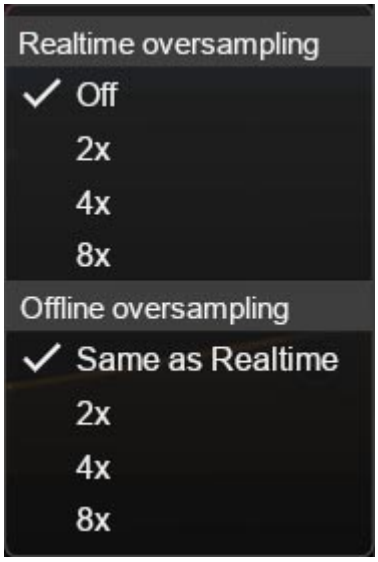
The Menu button

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.

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 Vocal Studio Control Panel

The Vocal Studio control panel consists of three racks:

- The **top rack** with calibration controls (*sensitivity*), input saturation, and dynamic processing: compressor, noise gate, de-esser.
- The **middle rack** with EQ and spectral correction tools (high-pass, low-pass, focus band, etc).
- The **bottom rack** with special effects and time-based effects (delay, reverb).

The Top Rack

This rack provides access to general controls and dynamic processing controls.

Top Rack

Sensitivity Section

This section is used to calibrate the plugin to:

- Achieve the ideal level for the various Vocal Studio modules to sound optimal.
- Maintain a consistent level regardless of the chosen compression level. This allows comparisons to be made with/without compression, or even with/without Vocal Studio, eliminating volume bias.

To properly calibrate the plugin, turn the *Sensitivity* knob until the circular LED ring around it mostly shows **GOOD** . You can also perform this operation automatically by pressing the **LEARN** button during playback of a typical passage from your vocal track, which will execute an automatic calibration. At the end of this process, the display should mostly show **GOOD** .

Note: This parameter is not saved with presets to ensure they remain independent of input levels.

"Sensitivity" Section

Drive Section

This section allows you to adjust the input saturation of the plugin. Saturation enhances the vocal track by adding harmonics, which can thicken the tone or even push it into noticeable distortion, similar to what a microphone preamp would do.

Three saturation colorations are available:

- **Solid-state** : Recreates the signature sound of a classic British Class A transistor preamp.
- **Tube** : Emulates a pentode-style tube preamplifier.
- **Germanium** : Captures the distinct coloration of a germanium transistor preamp, similar to those found at the input stage of iconic spring reverbs.

The *Drive* control lets you set the amount of saturation applied. You can subtly color a flat vocal take or add warmth and body to a thin-sounding recording. At higher gain levels, the saturation turns into clipping, resulting in a more aggressive and edgy effect.

Drive Section

Gate Knob

This control adjusts the noise gate level. A noise gate is an audio processing effect designed to block unwanted noise from passing through the audio circuit, ensuring a cleaner sound output. This effect typically mutes background noise when the singer is not performing. Above the chosen threshold level, the sound passes through; below this level, the gate is closed, resulting in silence.

Rather than a real noise gate, this module actually behaves like a dynamic expander — instead of cutting the signal completely, it gently reduces the level of sounds below the threshold, preserving a more natural and musical feel.

While a noise gate is usually a complex effect with many adjustable parameters, Vocal Studio features an advanced algorithm that requires adjusting only one parameter, allowing for quicker processing and letting you focus on the music rather than technical settings.

To set it correctly, ideally, place it in a passage where the singer is silent and adjust the knob until the noise gate is closed (no sound - red LED). Make sure the chosen level is not too high by checking passages where the vocal volume is at its lowest to ensure the noise gate remains open (you can hear the singer - green LED). An incorrect noise gate threshold setting (too high) can cause unwanted audio artifacts.

"Gate" Section

De-ess Knob

This knob adjusts the de-esser level. A de-esser is a dynamic tool that reduces or eliminates sibilance, those overly pronounced "s" sounds that can make a vocal track unpleasant to listen to.

Like the noise gate, the Vocal Studio's de-esser uses an advanced algorithm that only requires adjusting a single parameter, making it easier to achieve a smooth vocal sound.

To set it correctly, identify a passage where sibilance is unpleasant and turn the knob up until an acceptable reduction is achieved (you can observe the gain reduction of sibilance on the indicator around the knob). Ensure that the chosen level is not too high by checking passages without sibilance to confirm that the de-esser is not working unnecessarily (the gain reduction indicator should barely light up). An excessive de-esser setting can cause an undesirable "muffled" effect on the vocal track.

You can get visual feedback of the de-esser's effect in two ways:

- On the LED ring around the knob, which shows the gain reduction from 0 dB to -24 dB.
- On the frequency response curve in the EQ's spectrum analyzer.

Note: This de-esser is a multi-stage process applied both before and after the equalization stage.

"De-essing" Section

Compress Knob

This control adjusts the dynamic compression. A compressor is a tool that reduces the dynamic range of a signal, meaning it balances the difference in volume between loud and quiet moments in a vocal track. Proper compression helps bring a vocal forward in the mix and can create a more "produced" effect on a vocal track.

Compression is usually a complex process with many adjustable parameters, but Vocal Studio features an advanced algorithm, consisting of several compressors (applied before and after the EQ process), that only requires setting a single knob, making it faster and easier to achieve the desired effect.

Simply increase the compression level until the desired compression effect is achieved. It can be useful to toggle the compressor on/off while adjusting it to ensure the chosen compression level genuinely enhances the vocal track and its place in the mix.

You can get visual feedback of the compressor's effect via the LED ring around the knob, which indicates the gain reduction from 0 dB to -24 dB.

Compression Knob

Compression *Type* Selection Buttons

This section allows you to choose between three different compression styles:

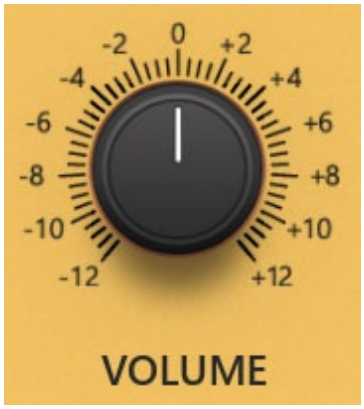
- **NATURAL** : This compression algorithm is based on a FET compressor, providing a natural and relatively transparent compression
- **MODERN** : This mode offers a compression style more characteristic of modern productions; it is faster and more colorful than "NATURAL" mode, giving a more "produced" effect. However, be careful not to push it too far, as it may result in an overly noticeable "pumping" effect
- **GRITTY** : This algorithm is a bit more radical than the other two; it adds what might be described as "grit", meaning a slightly aggressive color that helps the vocal cut through a dense mix

Compression Type Selectors

Volume Knob

This knob adjusts the plugin's output volume within a range of +/- 12 dB.

The outer ring displays the RMS level difference between the plugin’s input and output, ranging from +12 to -12 dB, to help adjust the output level. Note that since the human ear does not have a uniform sensitivity curve, a perceived volume difference may still exist even when the displayed level is close to 0 dB.



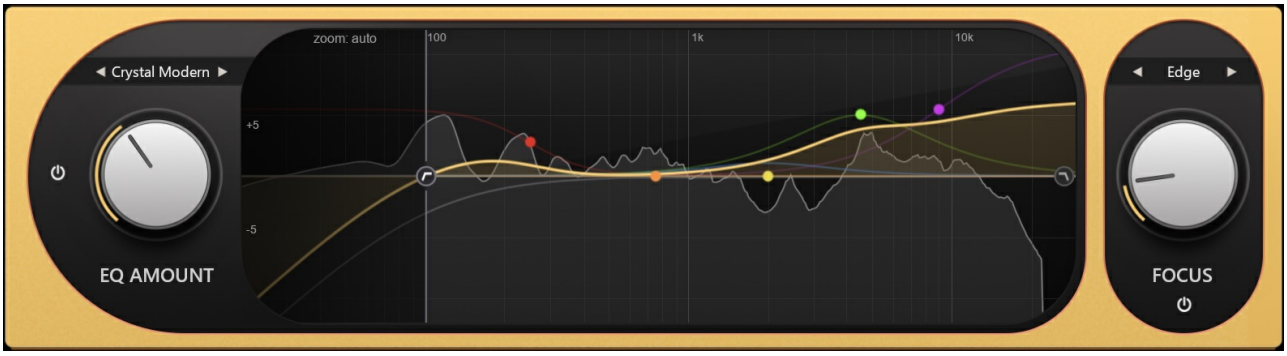
"Volume" Knob

The Middle Rack

This rack is used for spectral processing, such as high-pass/low-pass filtering and equalization.

Equalization is simply a process that allows boosting or attenuating certain frequencies of an audio signal. For example, an equalizer can increase the amount of bass in an audio track. An equalizer also includes high-pass and low-pass filters (also called low-cut and high-cut), which eliminate the lower spectrum below a certain frequency and the upper spectrum above a certain frequency.

This processing helps improve the intelligibility of a vocal track, prevent frequency masking with other elements in the mix (thus enhancing vocal clarity in the context of the song), and provide an aesthetic "coloration" to the vocal track.



Middle Rack

Equalization Section

This section controls the equalizer, which consists of seven parametric bands:

- Band 1: A low-shelf filter, which boosts or attenuates frequencies below a certain point.
- Bands 2, 3, 4: Traditional bell filters, used to boost or cut the volume around a specific frequency.
- Band 5: A high-shelf filter, which boosts or attenuates frequencies above a certain point.
- A high-pass filter, which removes all signals below a certain frequency.
- A low-pass filter, which removes all signals above a certain frequency.

Additionally, it includes an **EQ AMOUNT** knob, which acts as a master gain control, applying the same factor of attenuation or boost across all five bands. For example, if a bell filter has a gain of

+8 dB, setting this control to 50% will result in a +4 dB gain.

A presets section is also available, allowing users to quickly achieve a satisfying result without manually adjusting each filter.

Interaction Modes:

- Clicking and dragging a band with the mouse allows you to adjust the filter's frequency and gain.
- You can change the filter's bandwidth using the mouse wheel.
- Right-clicking on a band disables it.

Equalization Section

Focus Section

This section is a dynamic equalizer band that enhances "density" in a specific frequency range. It features three modes and a knob to control the intensity of the effect.

The three modes are:

- **Air** : Adds density above 10 kHz to create an airy sensation and emphasize breathiness.
- **Edge** : Adds density around 1-2 kHz to enhance presence in a voice that sounds too "hollow".
- **Presence** : Adds density around 4-5 kHz to improve vocal clarity and articulation.

The **FOCUS** knob adjusts the intensity of the effect.

"Focus" Section

The Bottom Rack

This rack contains three subsections:

- The **Special FX** section: Allows adding special effects (octave, saturation, etc.).
- The **Delay** section: Allows adding and configuring a delay effect.
- The **Reverb** section: Allows adding and configuring a reverb effect.

Bottom Rack

Special FX Section

This section allows you to select an effect preset from the dropdown menu and adjust its intensity using the amount knob.

The following effect presets are available:

- **Wide Stereo** : Creates an artificial stereo image from a mono source, for a wider and more immersive sound.
- **Thick Octave** : Adds a voice one octave lower to enhance the low end.
- **Wide Octave** : Combines a lower octave voice with a stereo effect for a richer and more spacious sound.
- **Soft Breath** : Adds a whispered voice effect, subtle and airy.
- **HF Whisper** : Injects an airy effect into the high frequencies, bringing clarity and intimacy to the vocal.
- **Wide Breathy** : Blends a whispered voice effect with stereo widening.
- **Classic Chorus** : Applies a classic stereo chorus effect.
- **Phasy Motion** : Produces a mono phaser effect for subtle modulation.
- **Wide Movement** : Combines a mono phaser effect with stereo spatialization.
- **Breathy Octave** : Mixes a lower octave voice with a breathy, whispered effect.
- **Ultra Large** : Includes a lower octave voice, a whispered effect, and maximum stereo widening.

Adjust the intensity of each preset easily with the dedicated control knob.

It is also possible to apply a high-pass and low-pass filter to limit the frequency range of the effect, using the filters at the bottom of the section.

"Special FX" Section

The *Delay* Section

This section adds a delay effect to your vocal track (delay repeats the sound to simulate an acoustic echo).

The bypass button controls whether the signal is sent to the effect, allowing you to apply it only to part of a phrase or a single word. The knob adjusts the amount of effect applied.

Available delay types:

- **Slapback** : A short, punchy delay inspired by vintage tape delays, with little to no repeats.
- **Ping-pong** : A stereo delay with repeats bouncing between the left and right channels at a constant interval.
- **Wide** : A delay with different repeat times for the left and right channels, resulting in a wider stereo image.

Delay color types:

- **Modern** : A clean, modern-sounding delay, ideal for precise and polished results.
- **Tape** : A warmer-sounding delay, inspired by analog tape echo units.
- **Disk** : A delay with a distinctive tone, inspired by a famous magnetic disk echo unit like the Binson Echorec.

Several parameters allow you to further fine-tune the effect:

- **Tap** : Sets the delay time by clicking in rhythm. If the SYNC mode is activated, it will be turned off. Note that the Tap button is not available in Slapback presets.
- **Sync** : Delay time is typically set as an absolute value in milliseconds. However, enabling the SYNC button makes the delay time relative to the project's BPM.
- **Time** : Manually adjusts the interval between repeats. Not available when the SYNC button is on.
- **Sig.** : Available only in SYNC mode, this parameter adjusts the delay time proportionally to the project's tempo. The setting 1/4 corresponds to the duration of a quarter note, 1/8 to an eighth note, 1/16 to a sixteenth note, and so on. Positions marked 'T' represent triplet durations, while positions marked 'D' correspond to dotted note durations.
- **Feedback** : Controls the number of repeats.
- **Width** : Adjusts the stereo width of the delay. At 0%, the signal is mono; at 100%, fully stereo.
- **Drift** : Adjusts the delay time difference between the left and right channels.
- **Ducking** : Lowers the delay volume while the voice is present, improving intelligibility while keeping the repeats at the end of phrases.
- **HP / LP** : High-pass and low-pass filters to limit the frequency range of the delay, helping it blend better into the mix.

"Delay" Section

The *Reverb* Section

This section adds a reverb effect to your vocal track (reverb simulates an acoustic space to create depth).

The bypass button controls whether the signal is sent to the effect, allowing you to apply it only to part of a phrase or a single word. The knob adjusts the amount of effect applied.

Several types of reverb are available, some being available in several sizes/lengths:

- **Halls** : Simulates a large space, offering natural, long, and immersive reverb.
- **Plates** : Simulates a vintage plate reverb unit, producing a warm and rich sound.
- **Springs** : Simulates a spring reverb, for a colorful and vintage effect.
- **Room** : Simulates a small/medium-sized room, useful for adding depth without overwhelming the mix.
- **Chamber** : Simulates a reverb chamber (a physical room used in the past to create natural reverb).
- **Gated** : Combines a reverb with a noise gate to retain only the beginning of the reverb tail.

Several parameters allow further adjustment of the effect:

- **Pre-delay:** The time between the initial audio source and the start of the reverberated signal. This parameter helps increase the perceived size of the reverb.
- **Brightness:** A filter that enhances or reduces the brightness of the reverb.
- **Ducking:** Reduces the amount of reverb present while the original signal is active. This improves vocal intelligibility while preserving reverb tails.
- **Width:** Adjusts the stereo width of the reverb. At 0%, the signal is mono; at 100%, it is fully stereo.
- **HP / LP:** A dual filter that adjusts the high and low frequency ranges to shape the reverb color and enhance clarity in certain frequency zones.

"Reverb" Section

Minimum Configuration

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

Available formats:

- VST 2.4 (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

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