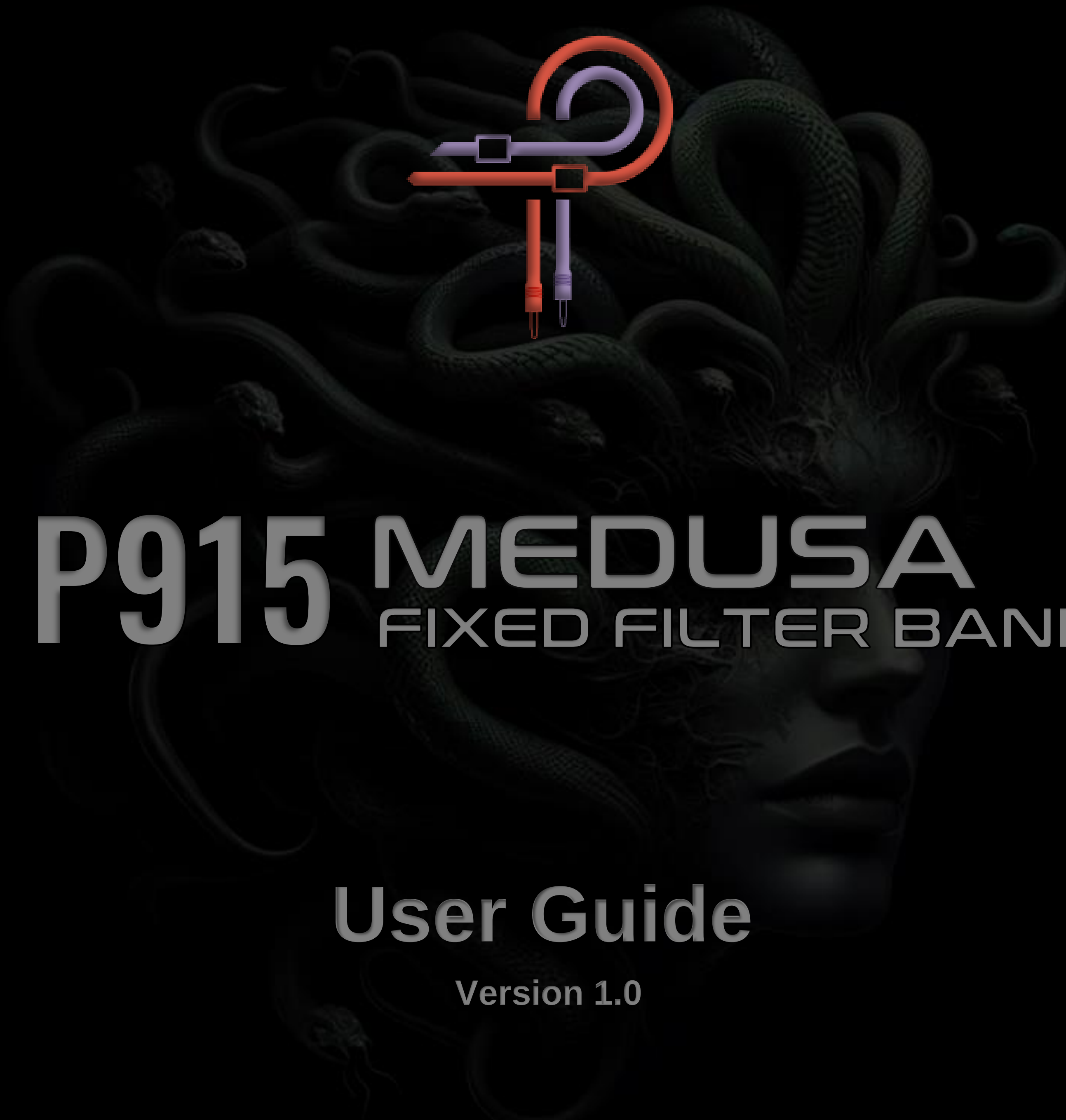


P915 MEDUSA FIXED FILTER BANK

User Guide

Version 1.0

Foreword

At a time when Bob Moog was at the forefront of pioneering voltage control for oscillators, filters and amplifiers, he also conceived a fixed filter bank (FFB) that broke the mold by requiring no voltage control. It aimed to enhance the primary signal by adding or removing harmonics at specific frequencies. The FFB could mimic the natural resonances of acoustic instruments and offer creative sound shaping possibilities. Whether for traditional or avant-garde music, the FFB was a valuable addition.

Thanks to its 14 fixed frequency points, the original 914 FFB was a tweaker's dream allowing for control over how much of each frequency band was added to the signal. Over time, several clones of the 914 emerged based on active filter designs, but there was a longing for a design faithful to the original. Gone was the use of simple inductors, capacitors, and resistors for the filter cells. Gone were the resulting subtle interactions, lost to the sterile world of semiconductors. A design based on the original structure was begging to be pursued.

After sourcing inductors and refining schematics, a faithful hardware replica emerged. This was followed by a meticulous Pulsar Modular software model dubbed the P914 FFB that faithfully matched the original two-stage design, with added features like channel selection, enhancing the filter's charm and character. This development could not have been achieved without the detailed input, direction and mentoring of David Ingebretsen (https://modularsynthesis.com).

P915 Medusa is the next generation evolution of the P914 FFB. It further modernizes this original design, targeting live performance and advanced creative production needs with features like channel-based frequency splitting and delay, a performance bank and the ability to smoothly transition between performance snapshots.

Pulsar Modular – **The sound is unbelievable.**

OS – Oversampling
Enables OS options (OFF, INTEL, VINTAGE or HD).

TX – Transformer Selection

Noise
Adds depth when enabled.

LOW SHELF
Fixed frequency inductor-based low shelf filter.

DRIVE
Meow

Q Variation
Alternate band pass filter Q settings.

Preset Browser

A/B Compare

W. VOL Slider
Wet volume output gain.

Main Out Slider
Final stage clean output gain.

HIGH SHELF
Fixed frequency inductor-based high shelf filter.

SUM
Changes the MIX slider from wet/dry ratio to wet summed to dry.

Options Menu
About / License Status / User Guide / Options and Preferences

Bypass

Polarity

TILT
Tilts all filtered frequencies along the entire spectrum.

High Pass Filter
HPF with resonance. Enable or disable with the associated button.

PERFORMANCE Bank
Stores and recalls snapshots of current settings. Click and hold to store, click to recall. Can be saved to a Preset.

Preset Reload
Resets the originally loaded preset values.

MORPH
Store separate filter settings in the SOURCE and TARGET endpoints by clicking S or T. Use the MORPH slider to transition between them.

Filters
Fixed frequency inductor-based filters. Can be muted individually or in groups with the M button.

Channel Delay
Delays the left or right channels. Enable or disable with L and R buttons.

SPLIT
Assigns the upper row to the left channel and the lower row to the right channel.

Low Pass Filter
LPF with resonance. Enable or disable with the associated button.

MIX – Mix W / D
Ratio of the processed (W) signal to the input (D) signal.

Save
Saves the size and position of the current instance.

Bypass allows the unaffected audio signal to pass through without being processed.

Polarity inverts the audio signal.

Turns the characteristic analog filter noise off or on. Turn this on for a desirable and nostalgic effect on your audio.

Transformer selection changes the characteristics of the low end. Set to Low for more bottom end, set to High for tighter Bass. Options are Low, Low Mid, Mid, High Mid and High. The default is LM (Low Mid).

Left click cycles forward, right click cycles backward.

Oversampling options allow P915 to optionally operate at a multiple of the host sample rate. With OS off, P915 operates with zero latency at the host sample rate (x1).

When oversampling is on, different options are made available. See the descriptions of VINTAGE mode, INTEL mode and HD mode below.

VINTAGE mode operates at double the host sample rate (x2). It applies smooth filters to upper frequencies to maintain a classic rolled-off characteristic and allows any aliasing signals to remain unfiltered. This provides the ability to creatively combine a smooth, vintage top end with modern inharmonic distortion. This is most effective when oversampling at a 44.1 kHz or 48 kHz host sample rate.

INTEL (intelligent) mode operates at double the host sample rate (x2). It scans the full frequency spectrum and attenuates any aliasing signals. The amount of processing applied by this advanced filtering is highly dependent on the signal and the degree to which P915 is being pushed.

HD mode operates at an internal sample rate of 384 kHz. It utilizes the same full frequency scan filtering strategy as INTEL mode. The high sample rate and filtering mechanism make this a pristinely high-quality option at a surprisingly efficient CPU load.

To achieve HD oversampling, P915 applies the following logic:

- 44.1 and 48 kHz oversamples at x8.
- 88.2 and 96 kHz oversamples at x4.
- 176.4 and 192 kHz oversamples at x2, thereby enabling INTEL and VINTAGE options.
- 384 kHz disables oversampling options.

Q Variations are slight differences in Q that are applied to each of the band pass filters to simulate analog component tolerance, leading to distinct resonant characteristics.

Browse, load and save presets and performances using the Preset Browser.

Clicking the left save icon saves only the stored performances to the currently selected preset. Holding the CTRL (Windows) or CMD (Mac) key while clicking the left save icon saves the performances and overwrites the currently selected preset. To create a new preset with performances, click the right save icon. A red asterisk* will show up next to the left save icon to indicate the current preset has been modified.

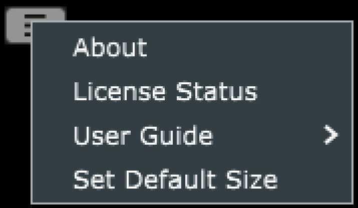
Tip: Modified factory presets will be preserved when updating the software if the install presets option is not selected. This is the default update version installer setting.

A/B allows for temporary storage of different settings for quick comparison. The arrow button allows for copying the active side to the inactive side.

Tip: When comparing settings, clicking the A/B button will perform the toggle. This is a single button, so it is not necessary to move the mouse to alternate back and forth. This makes it easy to compare without knowing which one is selected. We recommend doing this with your eyes closed for maximum focus.

The W. VOL slider features -12 dB to 12 dB of gain applied to the wet signal for increased creative control over the wet level.

The MAIN OUT slider features -12 dB to 12 dB of clean gain, applied at the final output stage for overall control.



Options Menu

About – Check the version number or demo expiration date.

License Status – Manage your license.

User Guide – Open the user guide.

Set Default Size – This option can be selected to apply the size of the current P915 instance as the default size for all instances of P915.

The TILT filter is a seesaw-like curve that can be used to boost one end of the spectrum while at the same time cutting at the other. You can brighten or darken sounds while maintaining the basic spectral balance.

Balance is represented by the ratio of the dark area (L) to the light (H) area. For example, when the TILT filter is used to increase high frequencies, the light area will cover a larger area than the dark area.

The LOW SHELF is a specially designed passive fixed Q boost filter with a center point of 60 Hz and a maximum gain of 8 dB.

The DRIVE knob pushes the tolerance limits of the internal circuitry, resulting in a thick, assertive, enlarged tone.

The HPF (High Pass Filter) features a 12 dB/oct stopband roll off with a separate resonant peak control (RES).

Enable or disable with the associated button.

The PERFORMANCE Bank stores and recalls up to 6 variations of filter and DRIVE settings (snapshots).

Click and hold a button to store the current settings to that button, thereby creating a PERFORMANCE snapshot. The light will flash twice indicating that the snapshot has been stored. Click the button any time to recall the snapshot as it was saved.

All PERFORMANCE snapshots are stored along with a saved Preset.

Tip: CTRL+ALT (Windows) or CMD+OPTION (Mac) + Left Mouse Click on a PERFORMANCE snapshot to store it in the SOURCE position of the MORPH slider. Do the same with a Right Mouse Click to store it in the TARGET position.

The Preset Reload button reloads the state of the originally loaded Preset, not including the snapshots stored in the PERFORMANCE Bank or the MORPH slider.

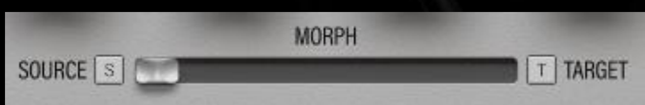
This is very useful for situations where new PERFORMANCE snapshots or MORPH endpoints are stored and are to be saved with the Preset, but the Preset should remain in the originally loaded state otherwise.

In use, after loading a Preset and storing a new PERFORMANCE snapshot or MORPH endpoints, press the Preset Reload button before saving the changes to the Preset if keeping the originally loaded Preset starting state is desired.

Vintage styled fixed frequency inductor-based band pass attenuation filters. Each filter is set at a musically complementary half octave interval from the adjacent filter (filters are arranged top to bottom, left to right).

The M button will mute the filter. Use CTRL+ALT (Windows) or CMD+OPTION (Mac) + Left Mouse Click with the M button to mute or unmute a full row. Use the same keys with a Right Mouse Click to mute or unmute both rows. By further adding the SHIFT key to these combinations, the state of each mute button in the row or both rows will alternate, meaning whatever was enabled will become disabled and vice versa.

*Note: You may notice the half octave interval points are not the arithmetic halfway point. While the calculation to arrive at a full octave is arrived at by multiplying by 2, arriving at a half octave is not so straightforward due to the logarithmic scale. To calculate a half octave, multiply by 1.414 (e.g. 250 * 1.414 =~ 350; 350 * 1.414 =~ 500 - please note the fixed frequencies are rounded visually for usability).*



The MORPH slider allows for smooth transitions between two sets of filter settings.

To set the starting point, simply click the S button (SOURCE) once all filters and the DRIVE control are set as desired. Likewise, to set the ending point, simply click the T button (TARGET).

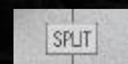
These settings are stored along with a saved Preset.

Tip: CTRL+ALT (Windows) or CMD+OPTION (Mac) + Left Mouse Click on a PERFORMANCE snapshot to store it in the SOURCE position of the MORPH slider. Do the same with a Right Mouse Click to store it in the TARGET position.

Note: Morphing between filter settings using MIX SUM and filter settings using non-MIX SUM will generate an unavoidable short noise or click at the 50% mark of the MORPH slider.

Each channel can be delayed up to 99 ms. The L button activates the left channel delay while the R button activates the right channel delay.

This can be used either in combination with or separately from the SPLIT functionality.



Splits the band pass filters into two banks of six bands each, with the top row of filters applied to the left channel and the bottom row of filters applied to the right channel.

Splitting the banks in this manner results in very musical and complementary filtering since each bank features octave spacing between filters and the two banks are offset with each other by a half octave.

Note: The HPF, LPF and shelving filters still apply to both channels when SPLIT is enabled.

The HIGH SHELF is a specially designed passive fixed Q boost filter with a center point of 9 kHz and a maximum gain of 8 dB.

The LPF (Low Pass Filter) features a 12 dB/oct stopband roll off with a separate resonant peak control (RES).

Enable or disable with the associated button.

Engaging the SUM button sums the wet signal into the dry signal. When the MIX slider is at 0, the fully unprocessed sound is heard and as it is increased, the wet signal is increasingly applied to the dry.

Disengaging the SUM button alters the ratio (or balance) of wet to dry signal. When the MIX slider is at 0, it is fully dry, but unlike when SUM is engaged, when it is at 100, it is fully wet with no dry signal.

Tip: Experiment with the W. VOL slider while having MIX at 100 and switching between having SUM engaged and disengaged to get a sense of how the W. VOL slider affects these MIX modes differently.

The MIX W / D (wet / dry) slider blends unprocessed dry signal in with the processed wet signal. Its behavior is dependent on the state of the SUM button. See the SUM button description for more information.

Balance is represented by the ratio of the dark area (D if in blend mode, empty if in SUM mode) to the light (W) area. For example, when MIX is used to increase the amount of wet signal, the light area will cover a larger area than the dark area.

Tip: Try this fun and easy workflow. Select the 'All Zero' Preset. Turn SUM off and move the MIX slider to the fully wet position. You should hear nothing. Experiment with the filters to find something that alters the sound in an interesting way. Now enable SUM (optional, try leaving it disabled as well), move the MIX slider to the fully dry position and increase the wet signal amount as desired.

Managing Presets

Basics

If the option to install presets is selected during installation, updates will overwrite the original presets but custom named presets will remain untouched. Be sure to save your own presets with different names using the save as option (to the right of the preset browser), or alternatively, ensure the preset installation option is not selected when updating the software.

Backing Up Presets

Presets can be backed up and restored using your operating system file manager. Simply perform a copy/paste of either individual preset files or the full presets folder to a backup location of your choosing. The presets folder can be found in the following locations:

FOR WINDOWS

‘C:\Users\Public\Documents\Pulsar Modular\P915 Medusa\Presets’

FOR MAC OS X

‘/Users/Shared/Pulsar Modular/P915 Medusa/Presets’

General

Fine Tuning Mode

Press and hold the modifier key (in macOS: “control, option or command”, in Windows: CTRL) while left clicking to adjust the knobs or sliders. Alternatively, right click when adjusting knobs or sliders without the need for a modifier key.

Uninstalling P915 Medusa

FOR WINDOWS

- In ‘C:\Program Files\Common Files\VST3’, locate the ‘P915 Medusa.vst3’ file and delete it.
- In ‘C:\Program Files\Common Files\Avid\Audio\Plug-Ins’, locate the ‘P915 Medusa.aaxplugin’ folder and delete it.
- In ‘C:\Users\Public\Documents\Pulsar Modular’, locate the ‘P915 Medusa’ folder and delete it. This folder contains the user guide and presets. If no other folders exist under ‘Pulsar Modular’, this can be deleted as well.

FOR MAC OS X

- In ‘/Library/Audio/Plug-Ins/Components’, locate the ‘P915 Medusa.component’ file and delete it.
- In ‘/Library/Audio/Plug-Ins/VST3’, locate the ‘P915 Medusa.vst3’ file and delete it.
- In ‘/Library/Application Support/Avid/Audio/Plug-Ins’, locate the ‘P915 Medusa.aaxplugin’ folder and delete it.
- In ‘/Users/Shared/Pulsar Modular’, locate the ‘P915 Medusa’ folder and delete it. This folder contains the user guide and presets. If no other folders exist under ‘Pulsar Modular’, this can be deleted as well.

Restrictions

The USER may not reverse engineer, disassemble, re-sample, create Impulse Response profiles or re-record, decompile, modify, alter in whole or in part PULSAR NOVATION LTD audio plugins for the intent of renting, leasing, distributing, repackaging (whether for profit or not).

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Please kindly report any errors or omissions in this user guide to psupport@pulsarmodular.com.

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P/N: 29223, Rev. 1.0

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