PSP Impressor

PSPaudioware.com

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Thanks to all our users around the world for ideas and help in the development of new plug-ins!

PSP Impressor

Overview

The **PSP Impressor** is an incredibly accurate, a fully-featured, high-precision compressor designed to process whole mixes, busses or single tracks. PSP Impressor emulates the magical classic characteristics of compressors with valve and opto-electronic circuits as well as classic modern compressors. PSP Impressor also features a highly configurable side-chain. The side-chain filter has a significant effect on PSPImpressor's sound and operation. Depending on your side-chain filter settings, the filter also lets you use the compressor as a de-esser or to reduce the pumping effects of heavy kick drums and the like.

The PSP Impressor offers many extremely useful features:

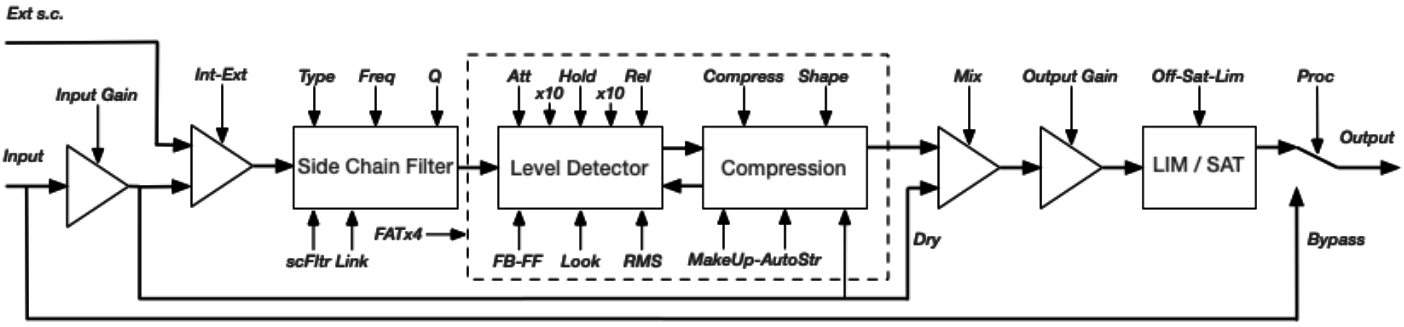
- A wide range of soft knee characteristics
- Peak or RMS level detection
- External or internal side chain signal with a smooth bell-type filter
- An output algorithm which can operate as either a brick wall limiter or soft saturation algorithm to preserve peaks above 0 dBFS.

PSP Impressor also works very well in combination with the [PSP Saturator.](#) With the PSP Impressor before the PSP Saturator in the signal path (think of it like placing a bus compressor before a tape machine), you can employ both plug-ins simultaneously (using subtle settings!) in order to impart a very cohesive sound, like your mix was being produced on expensive analog hardware. Of course, you can also reverse the order, placing PSP Saturator before PSP Impressor—never be afraid to experiment!

Block Diagram

PSP Impressor consist of several processing blocks:

- Input Gain
- Side chain source selector
- Side chain filter
- Level detector
- Compression
- Mix and Output Gain
- Output soft knee Limiter or soft-clip Saturator



Additionally Level detector and Compression blocks are encapsulated within the FATx4 quad oversampling algorithm.

PSP Impressor vs. PSP MixPressor

The most obvious difference between PSP Impressor and it's predecessor, PSP MixPressor, is its new GUI, but PSP Impressor offers many more enhancements and new goodies than just a new interface. PSP Impressor features many refinements in sound quality that make it a vital tool for the future of audio production.

Here's a list of innovations in PSP Impressor:

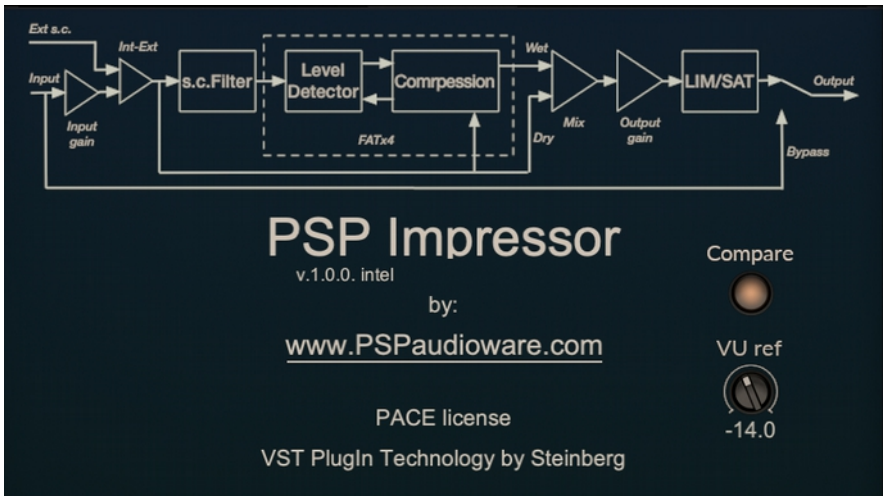
- New high resolution, scalable GUI.
- VST3 support.
- FAT quad sampling option.
- Entire compression engine runs at internal quad sampling for proper compressor modeling when FAT is disengaged and 16x sampling when the FAT is engaged.
- High sample rate support up to 400kHz.
- Feedback and feedback+feedforward compression modes, PSP MixPressor runs only in feedforward mode.
- New side chain filter modes: HPF and Equal Power. PSP MixPressor only had its powerful Bell type side chain filter, however HPF gives PSP Impressor a more classic touch, and the Equal Power results in extremely transparent and well balanced compression of complex tracks, drum groups or even entire mixes.
- Four Auto Attack and four Auto Release modes instead of just two modes for each.
- Refined compression Shapes for added versatility.
- Adjustable Auto Makeup mode.
- Adjustable channel Link.
- Input (Pre) and output (Post) meter modes with optional Compare mode.
- Adjustable VU reference level.
- Adjusted ranges for most of knobs to make them more user-friendly.
- Input - Output knobs' link.
- New output Limiter.
- Other algorithm refinements to make the PSP Impressor the natural tool of a choice for vocals, drums, acoustic guitars and other tracks, groups, and even during the mastering process.

Controls

Label

Click on the PSP Saturator label to open the about box. Click in the about box to close it.
The about box consist of a simplified diagram, information about the version number, license and PSP website link and VU reference trim pot.

About box



VU ref: sets the 0 VU reference level in dBFS. Typically, the plug-in is set to -14dBFS but other usual values like -12, -18 or -20 can be typed or dialed in. VU reference is a global preference. It is stored on your computer independently of your session and is loaded automatically into all instances of PSP Saturator.

Compare: sets the meters into input – output compare mode. When the Compare mode is engaged and IN mode is engaged the solid set of needles indicates the input level and the semi transparent set of needles indicates the output level, when OUT mode is engaged the solid set of needles indicates the output level and the semi transparent set of needles indicates the intput level. Otherwise, when the Compare mode is disengaged there is only a solid set of needles indicating a level accordingly to the IN/GR/OUT switch position. The Compare mode doesn't influence the meters operation in the Gain Reduction mode.

Meters

Each meter offers two needles. The red needle indicates digital peak levels, and should correspond with your DAW's peak meter. The dark gray needle is the VU ("Volume Unit") meter. The VU meter offers a representation of an averaging of the audio level with an integration time of 300ms, which is similar to how we perceive the signal's loudness. A default reference level for 0VU is -14dBFS, which means that when the peak level of a sine wave at -14dBFS is measured, the VU meter displays a value of 0VU.
The VU reference level is adjustable with VU reference trim pot on the rear panel.
The meters also include two text indicators for the peak and VU level under the meter scale.

The VU value is displayed under the scale on the far left, and the peak value on the far right, of each meter. The highest value for each indicator is held for one second. The text will appear red for values equal to or above 0VU and/or 0dBFS. The LEDs will also light whenever the signal reaches values equal to or over 0VU and/or 0dBFS.

PRE/G.R./POST [1]

- PRE: When this button is engaged, the meters show the input signal level.
- G.R.: When engaged, the meters display the gain reduction of the compressor excluding output limiter or saturator.
- POST: When this button is engaged, the meters show the output signal level.

The simultaneous Pre and Post levels can be observed by engaging the Compare Meters options engaged on the rear panel.

[1] Preset independent parameter

Dynamics Controls

ATTack: This knob sets the attack time. You can choose to set the attack to one of four Auto Attack modes from the slowest mode, Auto 1, to the fastest mode, Auto 4. All of the Auto Attack modes adjust the attack time based on the transient content of the track. If you would like to manually set the attack time, you can also adjust the attack time between .10ms and 100ms.

attack x10: multiplies the selected attack time by 10, changing the level detector's attack range to 1ms to 1s.

HOLD: This knob adjusts how long the compressor will hold (sustain) the gain reduction change after the level drops. If no new peaks happen during the hold phase, the signal is then passed into the release phase. You can set the hold time between 0ms and 1s.

RELease: This knob sets the release time. You can choose to set the release to "Slow," which offers an automatically adjusted slow release time, or "fast," which offers an automatically adjusted quick release time. If you would like to manually set the release time, you can also adjust the release time between 10ms and 1s.

release x10: multiplies the selected attack time by 10, changing the level detector's attack range to 100ms to 10s.

TIP: If the release time is very short, the compressor will operate more like a limiter, and its sound will resemble a highly saturated or overdriven analog tape. Long release times will cause the compressor to operate as a leveler, which is used for general leveling of the dynamics of particular parts of a mix.

FB-FF: This tri state rotary selector switches between feedback, feedforward or feedback-feedforward[2] mode to adjust the time-based behavior of the level detector to the needs of the processed audio.

[2]In the feedback compression topology the control level for the level detector is taken post compression which corresponds to how most classic compressor designs like opto, FET or tube based operate. Due to its response characteristics, compressors using feedback topology are often described as warm sounding. Feedforward compression is usually implemented on modern VCA compressors. Compressors using this topology have lower distortion and are usually more transparent than feedback compressors. The combined feedback-feedforward combines the best of both topologies and is used in some buss compressors.

LOOK: This button engages and disengages the lookahead processor by applying a small delay between the side-chain and processed signal. When the button is lit, LOOK mode is active. Look ahead covers the early attack (transient) phase of selected attack and RMS mode.

TIP: Lookahead mode uses those extra samples to more thoroughly analyze the side-chain signal and apply more precise gain reduction in advance of processing the signal content. This is accomplished by a simple delay line on the processed signal with an automatically adjusted time difference between the side chain and processed signal based on the current ATTACK setting. You can use the lookahead feature to (at least partially) protect against occasional overloads on transients when you combine medium attack times with deep gain reductions.

RMS: This button engages root mean square (RMS) level detection in the side-chain signal. This mode is more musical than Peak detection mode; however the fast response of Peak level detection mode might be better for material with a lot of transient information, such as percussive sounds and drums. When the button is lit, RMS mode is active.

TIP: When trying to emulate classic compressors, the RMS level detector is good for emulations of analog compressors with softer responses, while the Peak level detector has a more modern sound, like transistor-based compressors with fast and time-accurate response for precise level control.

TIP: You can completely alter the effect of PSPImpressor significantly by experimenting with attack and release times. For example, you can artificially produce a sharper attack with a pad or string track by using a prolonged attack-time setting (300 to 1000ms), short release time (10ms). Alternatively, you can create a pumping effect by setting the attack time to around 30ms and release time around 350ms. Be sure to experiment with the other settings, such as the compression depth (COMPRESS) as well!

Compression Configuration Controls

COMPRESS: This knob lets you adjust the depth of the compression algorithm (in other compressors, this would be the equivalent of a reversed threshold control, with a high COMPRESS setting equivalent to a very low threshold, and a low COMPRESS setting equalling a very high threshold). You can adjust this knob from 0% to 100%.

SHAPE: This dial lets you select from among six different compression curve characteristics and no compression. This setting affects the ratio and "knee" of the compression algorithm. You can select the following shapes:

- None—no compression
- Soft—a very gentle knee over a very wide dynamic range, with a 2:1 ratio at the end
- Wide—a wide knee ending with a 3:1 ratio
- Medium—a wide knee ending with a oo:1 ratio
- Short—a shorter knee ending with a oo:1 ratio
- Sharp—a narrow knee with with a oo:1 ratio, typical of soft knee limiters
- Hard—almost no knee, the processor operates as a hard knee limiter

Side Chain and Filter Controls

MAKE-UP: You can use this knob to manually restore the level of your signal after compression occurs (in other words, to "make up" any gain loss), or to control the exact level when the automatic make-up gain is applied. You can adjust the make-up gain between -12.0dB and +48.0dB.

FREQ: This knob sets the center frequency of the side-chain bell filter that drives the compressor's level detector. You can adjust the frequency between 20Hz and 20000Hz.

Q: The Q knob determines how steep the filter is. The higher the Q factor, the steeper the slope of the filter, the lower the Q factor, the wider the frequency range that gets through the filter. You can adjust the Q factor between .001 and 10.0.

s.c.FltrType: selects the side chain filter type. The three available filter types are:

 High pass filter

 Bell type filter

 Equal power filter

The default is Bell with both Frequency and Q controls enabled.
When HPF is selected the Q knob is disabled.
The equal power filter doesn't need any tuning so both Freq and Q knobs are disabled.

TIP: The effect of the bell type side-chain filter increases with increasing Q values. If you want to use PSPImpressor as a de-esser, for example, set the Q value high and the FREQ value between 5 and 10 kHz.

TIP: The Equal Power filter contains top and bottom shelves, and the entire midrange is gradually boosted towards high frequencies. This allows it to equalize the power of a typical track or a group which usually lowers with the increase in frequency. The result is a natural response of the compressor across entire spectrum or on complex groups like a drum group.

AUTO: This button activates and deactivates the automatic make-up gain function of PSPImpressor. With this engaged, PSP Impressor will automatically adjust the output gain depending on the Shape, Compress and the AutoStrength trimpot on the hidden panel. When the button is lit, AUTO is active.

TIP: When you want the compressor to only operate at certain frequencies (for example, when using PSPImpressor as a de-esser), don't engage the AUTO button. In these cases, the automatic make-up gain feature might cause the limiter/saturator to work too deeply, or produce dangerous levels at the plug-in's output.

SCExt: This button activates the external side-chain functionality of PSP Impressor. This allows you to use an external signal as the side-chain signal that PSP Impressor responds to. When the button is lit, SCExt is active.

Using External Side-Chain Signals with PSPImpressor

The exact steps to configure your host DAW to send an external side chain signal to an insert slot depends on the specific application. Typically, you'll need to configure the insert slot to be able to get three or four channels of input, then route processed signal to channels 1 and 2 and control signal to input 3 and if possible and needed to input 4 and then insert the PSPImpressor into the configured slot. See your host DAW's documentation for details. if your host DAW doesn't allow external side chain inputs for insert slots, the SCExt button will be disengaged.

SCL: This button activates the internal side-chain listening function of PSP Impressor. When the button is lit, SCL is active.

Global Controls

PROC-BYPASS[1]**:** This button activates or bypasses all PSP Impressor processes. The process is active when the button is lit.

GAINS LINK: The chain – like buttons before INPUT and OUTPUT labels engage the linked gains mode. When engaged, input and output gain knobs react with opposite movement. This allows you to turn up or down the drive of the compression chain with adequate compensation on the output gain.

The linked gains mode can be temporarily engaged or disengaged by pressing and holding Command (on Mac) or Control (on PC).

Using linked gains can help to accurately scale the overall processing depth of the plug-in (except the output limiter / soft saturator).

INPUT: Use this knob to adjust the level of the signal coming into PSP Impressor. You can adjust the input signal by -20.0dB to +20.0dB.

MIX: This sets the balance between the original (dry) audio signal and the processed (wet) signal. You can adjust the MIX between 0% (only dry signal) to 100% (only wet signal).

TIP: You can get some very creative compression effects by using the MIX knob. Setting this parameter below 100% decreases the influence of compressor operation on the signal peaks by moving the point of its operation to the middle of the dynamic range. This way, you can use PSPImpressor in complete mixes without destroying transients, which will result in simultaneous dynamic modulation and a general increase in the average level.
This technique of mixing a deeply-compressed signal with its non-compressed original is known as *parallel compression* or the "New York Drum Sound."

OUTPUT: This knob controls the master PSPImpressor output level. You can adjust the output signal by -20.0dB to +20.0dB.

LIM: This activates or deactivates the brick-wall limiter algorithm. It can be used to ensure no signal over 0dBFS is output, and it also generates a gentle saturation, perfect for bass, pads, and strings. When the button is lit, LIM is active.

THRU: This button, when active, disengages either limiter algorithm.

SAT: This button activates or deactivates the saturation limiter function. This mode accentuates transients, which is useful for adding brightness to loops or percussion instruments. Like the limiter, it also ensures no signal over 0dBFS is output from PSP Impressor. When the button is lit, SAT is active.

Internal Board Controls[2]

OPEN [1]: Opens the internal panel which contains the fine adjustment controls of the plug-in. Click on the Open button to close the internal panel.

FAT [1] : Engages our unique Frequency Authentication Technique, which is quadruple oversampling for the High algorithm and the main Saturation algorithm. Engaging the FAT mode strongly reduces the aliasing effect, which results in more pleasant saturation results. The FAT mode is recommended for major tracks in the mix or whenever strong saturation settings are used – especially in low sample rates. Engaging the FAT mode increases the CPU usage.

The processing latency is sample rate dependent. Numerical values next to the FAT switch indicate the latency in samples and milliseconds reported to the audio host application. The version number of the plug-in is displayed underneath latency values.

Link: Controls the side chain channels' link.

AutoStr: Sets the strength of the auto makeup. The default setting is the estimated RMS make up needed for given Shape and Compression settings. The maximum setting is the estimated makeup gain referring to 0dBFS. The Auto makeup mode doesn't include settings for attack, hold, release and side chain filter's settings, which can influence the exact amount of makeup needed. Our advice is to switch off Auto mode when strong side chain filter's settings are used. This control affects the processing only when the Auto mode is engaged.

MakeUp: Sets up the percentage of optimal post saturation makeup based on current Shape and Saturation settings. While the default value is 50, a higher (up to 1200 – strong makeup) or lower (down to 0 – no makeup) can be set.

HPF: Sets up the HPF filters' cut-off frequency. This control can be used to deal with sub harmonic content in the input signal or introduced by the processing within the plug-in.

HPF mode: Sets the HPF filter into pre, pre and post or post processing modes.

[2] All resistors are shown for entertainment purposes only and cannot be treated as controls

Side Chain Bar

We provide PSP plugins, especially processors, with the PSPaudioware standard SIDE CHAIN BAR. You can access this bar at the bottom of the plug-in interface. Here you can select (mix) the side chain source and switch the plugin into side chain listening (cue) mode.

INTERNAL/EXTERNAL Mix
The INTERNAL/EXTERNAL Mix slider sets the balance of internal to external signal in the final side chain audio channel.
If your DAW does not provide an external side chain source or is turned off for the plugin, the side chain audio channel is always set to internal, independent of the mix slider position.

INTERNAL
Click to quickly set the mix to 100% of internal source.

EXTERNAL
Click to quickly set the mix to 100% of external source.

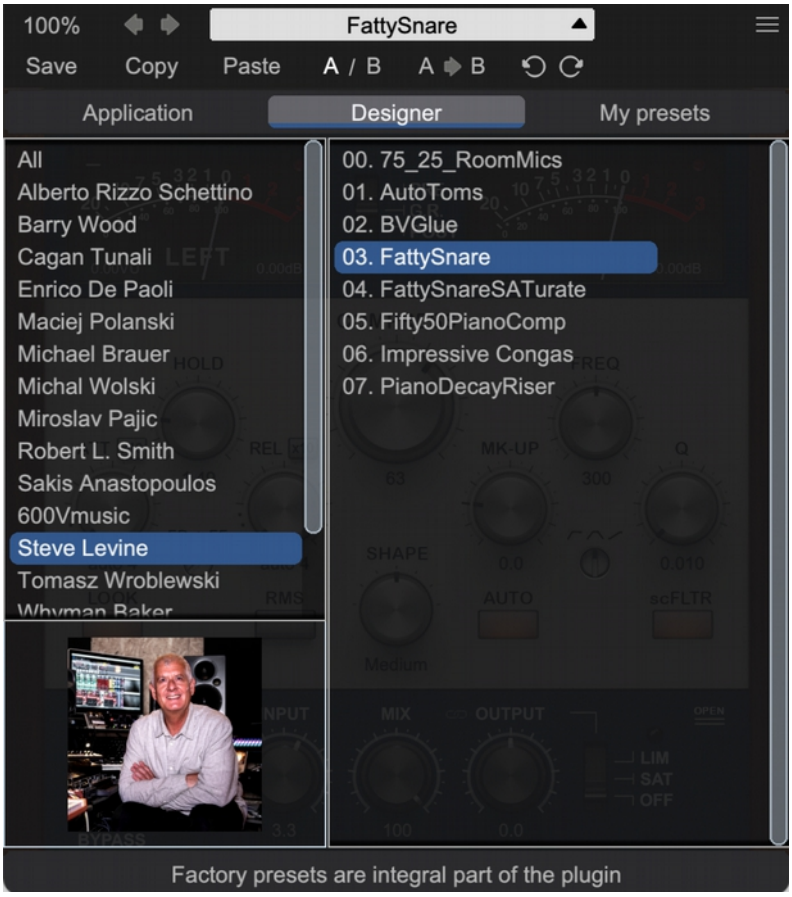
MONITOR
Click to switch the plugin into side chain listening (cue) mode.
If this mode is on, you'll hear the side chain audio which is being processed. To make it even more, clear that the side chain input is being monitored, the entire plugin GUI is tinted amber.

SIDE CHAIN LEVEL METER
Shows the signal level of the side chain input.

Preset Handling & View Options

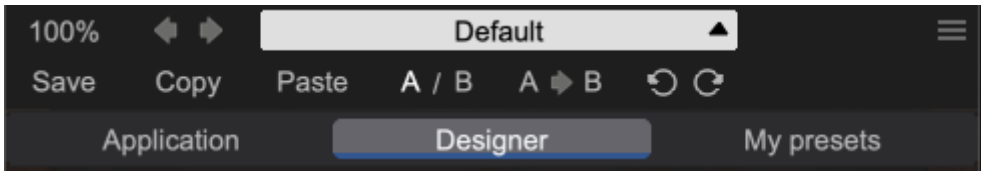
We provided all PSP plugins with a bunch of factory presets. These presets can be used as a starting point for further adjustments, or for quick fixes.

You access presets from the PSPaudioware standard PRESET BAR at the top of the plug-in interface.



Preset Browser

The PSP Impressor edition features a comprehensive new preset management and browser system. To access the preset browser, you click the preset name window at the top of the plug-in (which displays 'Default' when the plug-in loads).



The new preset manage has three main categories which can be accessed with the tabs at the top of the preset browser: **Application**, **Designer**, and **My presets**.

Application - shows all factory built-in presets grouping by application.

< Factory presets are built into the plugin and cannot be directly edited!
You can adjust them and save separately as user presets >

Designer - shows all factory built-in presets grouping by designer.

My presets - shows only user presets.
This view shows all of the presets you have created and saved, or downloaded and added to your custom presets for PSP Impressor.
To add categories to the preset list, you can create new subfolders in the preset directory.

For Windows users, this is located at:

C:\Users\Username\Documents\PSPaudioware.com\User Presets\PSP Impressor

For Mac users, this is located at:

~/Documents/PSPaudioware.com/User Presets/PSP Impressor

< You can always check the exact path by clicking on the "Show file in Finder" tab
at the bottom of the preset browser window. >

To select a preset, you can click a preset name in the right window. When clicked, the preset will be applied so that you can audition it. To confirm the preset choice, you can click the preset name once more to load it.

Each preset has own picture, click on it for open a designer www site.

Copy / Paste

Use the **Copy/Paste** feature to quickly transfer settings between instances of the plug-in.

To use this feature, you can click '**Copy**' at the top of the plugin below the preset browser window. Then, open a new instance of the plug-in on another track (or, on the same track) and click '**Paste**' to paste the settings to the new instance of the plug-in.

This feature can be particularly useful for processing similar instruments or sounds when only a few minor tweaks are needed.

A/B System

The **A/B system** for quickly checking and auditioning changes to the plugin settings.

The **A/B Button** at the top of the interface below the preset browser window allows you to A/B between the current and previous setting of the plug-in. This can be used to audition changes made to your mix, or to audition between two presets.

The **A>B Button** quickly copies the settings of the **A** setting to the **B** setting. This allows you to save your place and apply further tweaks and the audition them with the **A/B Button**.

Undo / Redo

Use the **Undo/Redo** feature to quickly navigate between setting changes.

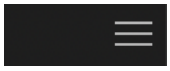
To use this feature, you can use the undo/redo buttons (CCW and CW arrows, respectively) located below the preset browser window. These buttons will undo changes to the current plug-in settings, or allow you to undo a preset change depending on the last action in the plug-in.

100% GUI resizing

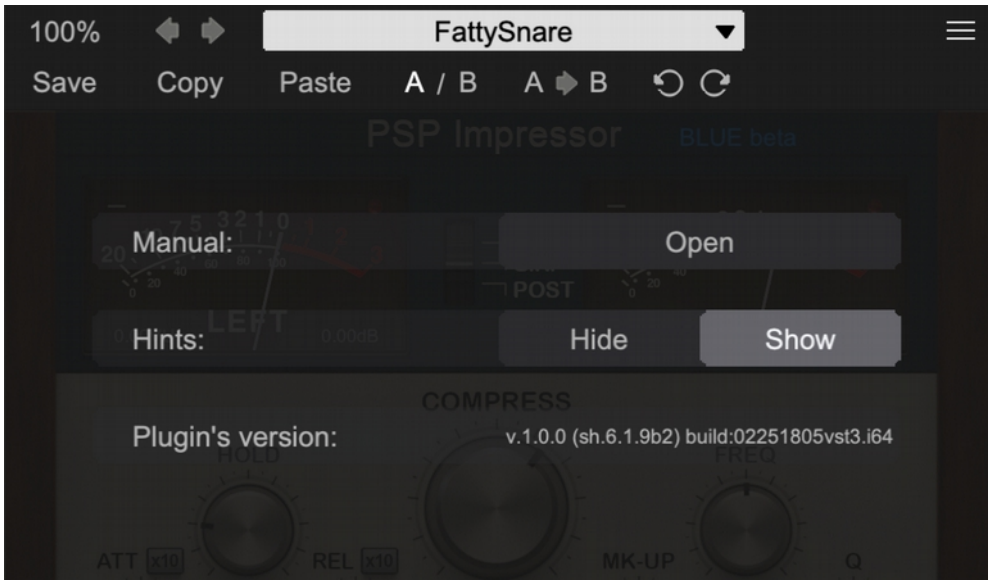
Scroll (using the mouse wheel or using two finger tap on the touchpad) this percentage up or down to change the GUIs zoom factor. Double click to reset its state to the default size (100%).

You can resize the plug-in interface by dragging the right bottom corner of the plug-in to any size you like.

CONFIG section



When clicking three parallel lines – the **CONFIG** menu will open and it allows you to open the manual, check the current plugin version number and turn on/off hints.



Minimum System Requirements

PC

VST3
- Windows 7 – Windows 11
- 64-bit VST3 compatible application

VST
- Windows 7 – Windows 11
- 64-bit VST compatible application

AAX
- Windows 7 – Windows 11
- 64-bit Pro Tools 11, 12 or Pro Tools HD 11, 12 or Pro Tools Ultimate

All DAWs
- The latest iLok License Manager application installed (an iLok dongle not required)

Mac (macIntel or arm AppleSilicon processors)

AudioUnit
- macOS 10.12 – macOS 12.01 Monterey
- 64-bit AudioUnit compatible host application

VST
- macOS 10.12 – macOS 12.01 Monterey
- 64-bit VST compatible host application

VST3
- macOS 10.12 – macOS 12.01 Monterey
- 64-bit VST3 compatible host application

AAX
- macOS 10.12 – macOS 12.01 Monterey
- 64-bit Pro Tools 11, 12 or Pro Tools HD 11, 12 or Pro Tools Ultimate

All DAWs
- Up to date iLok License Manager application installed (an iLok dongle not required)

Processing

- All internal processing done with 64 bit double precision floats.
- 32 and 64 bit floating point audio streams supported.
- Sample rates up to 384 kHz supported.

Limitations of the demo version

We offer a 30 day evaluation period without any audio interruption or control limitations. To get access to the plug-in and your unique authorization details simply log-in to your account at our user area.

Enjoy !

PSP team

Support

If you have any questions about any of our plug-ins, please visit our website:
www.PSPaudioware.com

Where you can find the latest product information, free software updates, online support forum and answers to the most frequently asked questions.

Problems with the installation, activation or authorisation?
Please watch our troubleshooting video tutorials on our YouTube channel.

You can also contact us by e-mail: support@PSPaudioware.com.
We will gladly answer all of your questions. As a rule we respond within 24 hours.

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