PSP 285

The ultimate semi-modular delay machine

Operation manual

PSPaudioware.com

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OVERVIEW

PSP 285 is based on our legendary PSP 84 plug-in and its successor PSP 85, and is our latest exploration of the endless possibilities offered by variable sample rate delay lines. It can produce an extremely wide variety of delay-based effects – from simple slapbacks and echoes to complex rhythmic patterns, from faithful magnetic tape delay simulations to twisted out-of-this-world pitch-shifting and resonant filtering effects – all at world-class audio quality.

Features

- Semi-modular architecture
- Up to 10 seconds of delay (pre + main) time per channel, depending on internal sampling frequency
- Pre-delay for each channel
- Continuous delay time control over the delay time
- Cross-channel feedback and independent channel settings
- Channel link mode
- Delay gating for improved control over number and level of echoes
- Stereo (L/R) or Mid-Side (M/S) operating modes
- Wet signal ducking for improved sound clarity
- Modulation section with a tempo-synced LFO and envelope follower that can be mixed in any proportion
- A wide selection of envelope follower control signals in linked or unliked mode
- 6 click-free LFO waveforms as well as variable LFO offset and phase channel spread
- A selection of 17 different filter types, with adjustable cutoff and resonance as well as flexible routing capabilities
- Post filter or independently routed Drive algorithm with hard and soft saturation modes
- Multi algorithm reverberation module including springs, plates, various spaces, and reverse, available in both mono-to-stereo and dual mono modes
- Flexible reverb positioning in the signal chain: dry, wet, or mixed wet+dry delay output
- Mixer parameters can be stored in each preset, or applied to any preset selected when in Global mode
- Wet mode switch for use in sends
- Control signal filtering for smooth, click-free operation
- High quality 64-bit floating point signal path and unique proprietary processing algorithms
- Support for sample rates of up to 384 kHz.

Applications

PSP 285 is primarily designed for use on individual tracks during recording, mixing, or live performances – it's great for experimenting with vocals, guitars, or any other type of track. You can use PSP 285 for delays with classic sound and authentic feel, but its capabilities go far beyond that. Its algorithms are suitable for leading tracks, solos, synthesizers, drum loops, or instrumental patterns – or to create out-of-this-world audio phenomena, with a library of inspiring presets to demonstrate its potential.

PSP 285's independent channel settings and panned feedback let you go beyond simple stationary pan positions, to create lively spatial delay effects that shine through in a mix – including Mid-Side processing. Built-in ducking, which can be controlled by an external side–chain signal, can greatly improve the mix clarity. Resonant filters let you sculpt the tone of your delays, and the built-in envelope detector and tempo-synced low frequency oscillator (LFO) offer enormous modulation possibilities. Delay gating provides extended control over the number and level of echoes, something usually possible only with multi-tap delays.

Thanks to its variable internal sample rate, PSP 285 can create sounds ranging from clean and pristine to dirty and distorted… from pitch-twist effects and flangers to rich vocal doublers and unequaled detune effects. Last but not least, the built-in reverberation unit can add space and warmth whenever required.

ARCHITECTURE: HOW PSP 285 WORKS

Variable sample rate delay lines

The heart of PSP 285 is a pair of delay buffers that run at variable sample rates, ranging from 50% to 200% of the host sample rate. That means that the delay time you hear is controlled by two different factors:

- the number of samples in the delay line (*buffer length*), and
- how quickly those samples are processed (*sample rate*).

The actual buffer length is set by the Up/Down buttons located in the Delay Left and Delay Right section, or by the Invisible Slider between Up and Down (see p. 16). Delay time can be displayed in either milliseconds or as a note value for tempo-synced operation.

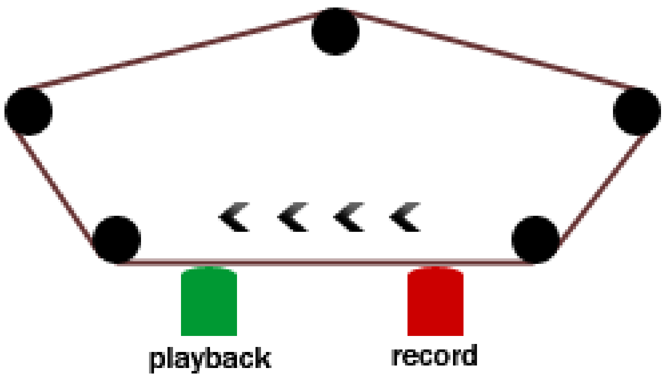
The plug-in's internal sample rate can be adjusted manually (via the MANUAL knob in the Delay section's Control block) or controlled by the signals coming from the Modulation section.

To understand how these two factors affect the delay time you hear, it's useful to think of a classic tape delay, as shown in Picture 1. In this delay, a loop of magnetic tape is in contact with two tape heads, one for recording and one for playback. The delay time depends on two things: the distance between the record and playback heads, and how fast the tape is moving.

These two factors are the exact analog equivalents of PSP 285's *delay buffer length* and *delay line sample rate*:

- tape length ~ delay buffer length, and
- tape speed ~ delay line sample rate

Increasing the buffer length results in a longer delay time, because the "head spacing" is wider. However, increasing the sample rate results in a *shorter* delay time because the "tape" is moving faster.



Picture 1: simple magnetic tape delay

It's obvious that in a mechanical delay, both of these ways to change delay time are needed. Putting the heads closer together means that the tape doesn't have to run dangerously fast to get short delays, and slowing down the tape means you can get longer delays without having to put tape heads far apart in a gigantic box. But why would we still want these two forms of control in a digital plug-in? Because the two methods produce two very different effects when you change the delay time, and both of them are useful and musical.

Changing the buffer length is heard as a sudden jump to another part of the audio stored in the buffer. In our tape analogy, it's kind of like instantly moving the tape head to another place in the loop to play back something else. This would be really cool, but you'd need magic to do it!

On the other hand, when we change the sample rate, the audio playback never stops – it just gets faster or slower. When you shorten the delay time, just like what happens with a tape being sped up, audio that's already being processed (i.e. "already on the tape") plays back at a higher pitch. Conversely, if you lengthen the delay time, the pitch of audio already recorded will drop. Either way, once the buffer fills up at the new delay time (i.e. all the "audio on the tape" has been replaced with audio at the "new tape speed"), then the pitch stabilizes once more.

High quality realtime sample rate conversion

One of the common complaints about digital effects and processors – including delays – is that they introduce artifacts altering the sound in undesired ways. PSP 285 is designed to minimize such artifacts… and since we're proud of how it does that, we're going to take a moment to explain it a little bit.

In very simple terms, digital audio needs at least two samples to define a single sine wave. . That means that you can accurately record audio at any frequency up to half the sample rate, which is called the *Nyquist frequency*. If you try to record frequencies above that, they will be misread by the analog-to-digital converter, creating other frequencies that you can hear, but aren't harmonically related to the audio input. This is called *aliasing*, and it can really ruin your audio… and since PSP 285 relies on sample rate conversion we took care to deal with that problem.

PSP 285 minimize aliasing problems with a set of two 6th order low-pass anti-aliasing filters with a resampling ratio dependent floating cutoff frequency. They are placed before and after sample rate converters to control aliasing. Those filters are tuned to emulate the subtle high-frequency loss typical of high quality delays from the past.

In addition, every system doing sample rate conversion has to figure out how to play back a waveform at the new sample rate without changing it very much. This requires a process called *interpolation*, and the relatively crude techniques found in many commercial products – *decimation* (throwing samples away) and *linear interpolation* (drawing lines between the old samples to make guesses at where the new samples should go) – introduce high harmonic distortion. In contrast, PSP 285 uses high quality multi-point interpolators at both the input and output stages of the delay buffer. These interpolators have been designed to introduce as little distortion as possible, while keeping CPU usage and latency at a manageable level.

When we combine these processes, the result is a pristine sound across the entire internal sample rate range. We think the results are quite special, and we think you will, too.

Filters

PSP 285’s Filter section contains two filters (Filter 1 and Filter 2) and a Drive module. These can be used to shape the spectral content of audio at several points in the plug-in’s audio path.

Filter 1 offers the following filter types:

- low-pass (LPF)
- semi low-pass (LPFs)
- band-pass (BPF) 1
- band-pass (BPF) 2
- notch (Ntch)
- semi high-pass (HPFs)
- high-pass (HPF)
- all-pass (APF)
- state-variable low-pass (sLPF)
- state-variable band-pass (sBPF)
- state-variable high-pass (sHPF)
- 4^{th} order ladder low-pass (lLPF)
- 4^{th} order ladder band-pass (lBPF)
- 2^{nd} order ladder high-pass (lHPF)
- gain low-pass (gLPF)
- gain band-pass (gBPF)
- gain high-pass (gHPF)

The first six filter types have been mathematically derived from analog 2nd order resonant filter prototypes. Their frequency response faithfully emulates that of their analog counterparts over the entire audio frequency range, thanks to oversampling. The Semi types are –18 dB shelving filters.

The next three filter types are digital emulations of state-variable filters, which are widely used in effects and synthesizers (most famously those designed by Tom Oberheim); these are the filters originally used in PSP 85. The ladder filters are based on the circuit design developed by Robert Moog, extended by Tom Oberheim and used in many classic synthesizers.

For most Filter 1 types, you can set the Cutoff Frequency (FREQ) and Resonance (RESO) manually, as well as modulate them both. Modulation depth can be negative or positive, which means the asymmetric LFO waveforms such as sawtooth can be inverted (upward saw becomes downward saw), and the envelope follower can be used to increase or decrease those parameters' values. With the modulation Alternate Routing (AltRt) button engaged, the phase lag between cutoff and resonance modulation can be set by the LFO Spread parameter.

The state-variable and ladder filters contain an internal saturation model, which emulates nonlinear characteristics of their analog counterparts.

In the Gain filter types, the RESO knob controls post-filter gain instead; these filters’ resonances are preset to a default medium value. Gain filters have been specially designed for tremolo, emulation of tape dropouts, or to achieve crazy variable-feedback effects.

The Drive module can be used to emulate a magnetic tape saturation, to control the feedback level, or to achieve more pronounced distortion effects. Drive can be routed and used independently, or as an integral part of Filter 1.

When set to work at Filter 1's insert point, the Drive algorithm is located post-filter. Otherwise, when the Drive is routed to the same insert point as Filter 1, it processes pre-filter.

There are two types of saturation available in the Drive module: Soft and Hard. The Hard saturation type is the same as the legacy Drive mode in PSP 85.

Filter 2 is controlled by a single cutoff frequency knob, and its frequency can't be modulated.

Filter 2's type list consists of six filter types:

- low-pass (LPF)
- semi low-pass (LPFs)
- band-pass (BPF)
- notch (Ntch)
- semi high-pass (HPFs)
- high-pass (HPF)

Each one of these types is available in two different slope modes, soft and sharp. Sharp filters are indicated with a + in the name, e.g. LPF+ or HPFs+.

When using Filter 1 or Filter 2 on feedback it is worth to mention that with low resonance values a gradual loss of frequency can be noticed over every echo. When sharp resonance values are selected a frequency loss is rapid and doesn't change considerably from tap to tap.

WARNING!

Loud feedback may damage your ears or speakers! When the filter is operating in FB mode, the plug-in can easily become unstable. Always start with low feedback and resonance values while in this mode, and increase them gradually to achieve the desired effect. It is a good practice to lower the plug-in output level as well, or enable and lower the ceiling level of the output Limiter.

At the top of each Delay module is the word PANIC. Click the word to instantly set feedback gains to minimum and switch off the entire Filter 1 section. This will stop any runaway feedback or other instability at the source.

Modulation

The Modulation section consists of two main functional blocks:

- low frequency oscillator (LFO)
- envelope follower

The LFO is capable of generating the following waveforms, selected by clicking icons:

- sine
- square
- triangular
- sawtooth
- random with sharp transients per step
- random with smoothed transients

The waveforms with abrupt level changes – square, sawtooth, and stepped random – are low-pass filtered to ensure that the plug-in’s audio output remains click-free. All six waveforms can be synced to host tempo and song start position, with adjustable phase offset.

The LFO rate has a range of 0.01 Hz (one cycle per 100 seconds) to 15 Hz. The phase offset between the left and right LFO channels produces fat chorusing and panning effects when applied to the delay line sample rate or to the resonance (gain) modulation of the Gain filters.

The envelope follower analyses the signal and extracts its temporal envelope. Its value depends on the weighted average of the absolute signal level, calculated for a given time constant. The envelope follower's time constant is adjusted with the SPEED knob, while its sensitivity to input signal level is determined by the Sensitivity (SENS) knob. The envelope followers can be set to unlinked (DUAL MONO) mode to process each channel separately.

The LFO signal can be mixed in any proportion with the envelope follower signal, making up the final modulation signal. This summed signal can then modulate the Filter 1 cutoff frequency, resonance, and/or the Delay sample rate.

Fiilter 1's modulation can operate in two modes. In Normal mode, Left (or Mid) and Right (or Side) modulation channels are connected to corresponding common modulation signal channels. In Alternative Routing mode (AltRt button), channel 1 of the modulation signal controls the Left (or Mid) and Right (or Side) cutoff frequencies, while channel 2 controls the resonances. This allows you to set up some phase lag between those two parameters – for example, when you add LFO Spread, or use an unlinked envelope follower with a stereo control input signal.

The Delays' sample rate modulation always operates in Normal mode: Channel 1 controls the Left (or Mid) resampler and Channel 2 controls the Right (or Side) resampler.

The PSP 285 Ducker and Gate are controlled by independent envelope followers with appropriate Speed settings.

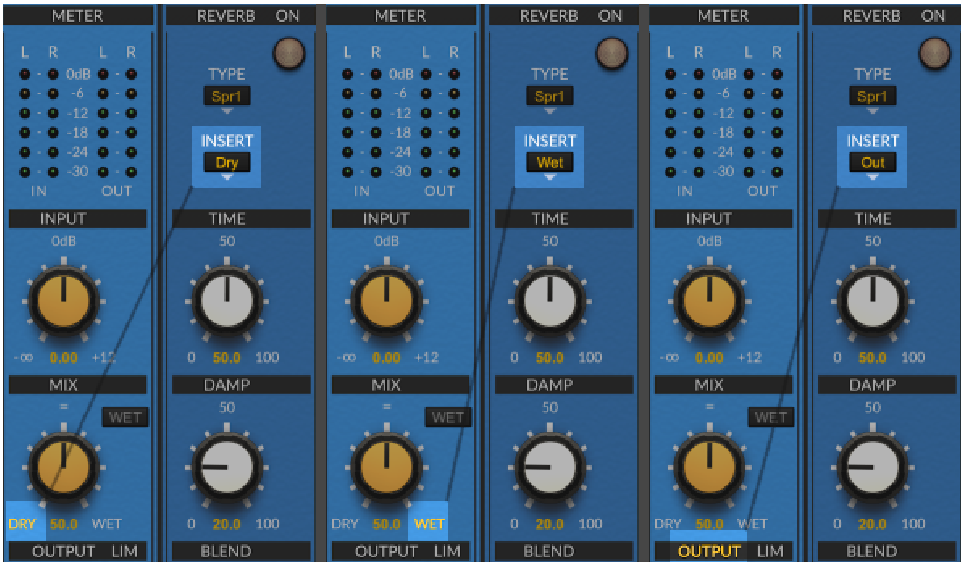
Reverberator

PSP 285 is equipped with a multi-algorithm reverberation module. This module is capable of emulating spring, plate, chamber, room, hall, and ambience reverbs, as well as inverted reverb. We did our best to provide algorithms that best suit the delay effect in each preset; however, the selection of the algorithm should usually be dictated by the specific track and mix where it's used. Each of the algorithms is available in mono-to-stereo and dual mono configurations.

The Reverb module has an adjustable reverberation time, damping, and Blend, which lets you set the internal wet/dry mix. The reverberation time is shown in percentage values relative to the available range for each of the algorithms.

The Reverb module can operate at three insert points – it can process only the wet (delayed) signal, only the dry (undelayed) signal, or the wet+dry overall output signal.

The Dry, Wet, and Output labels in the Mixer section turn yellow depending on the selected reverb's insert point, as shown below:



Master section

The input and output levels can be controlled by the INPUT and OUTPUT knobs. Since PSP 285 contains non-linear elements (saturation) and an envelope follower within its signal path, the INPUT knob position not only changes the volume, but can also affect the sound itself.

Dry and wet signals can be mixed in any proportion; PSP 285 uses the *–6 dB pan law*, meaning that at center position the dry and wet signals are both attenuated by 6 dB prior to summing. Unless a very short delay time is set, this might require some gain boost, at either the input or output, to compensate for the volume loss.

Mixer parameters can work in Preset mode, when each of those parameters is stored within and read from a Preset when it's loaded, or in Global mode when mixer's parameters are independent of presets.

The final stage of the Mixer chain is an optional brick wall limiter with adjustable Ceiling level. This limiter has a soft knee and a very short response time, so it can generate a considerable amount of distortion. Although its main purpose is to protect yourself and your studio gear from excessive signal levels, it can also be used to add an extra pinch of fuzz to your output!

Gate

PSP 285 has a built-in gate algorithm. The gate attenuates the signal whenever it is below the set threshold, and opens up when the signal comes back above above the threshold.

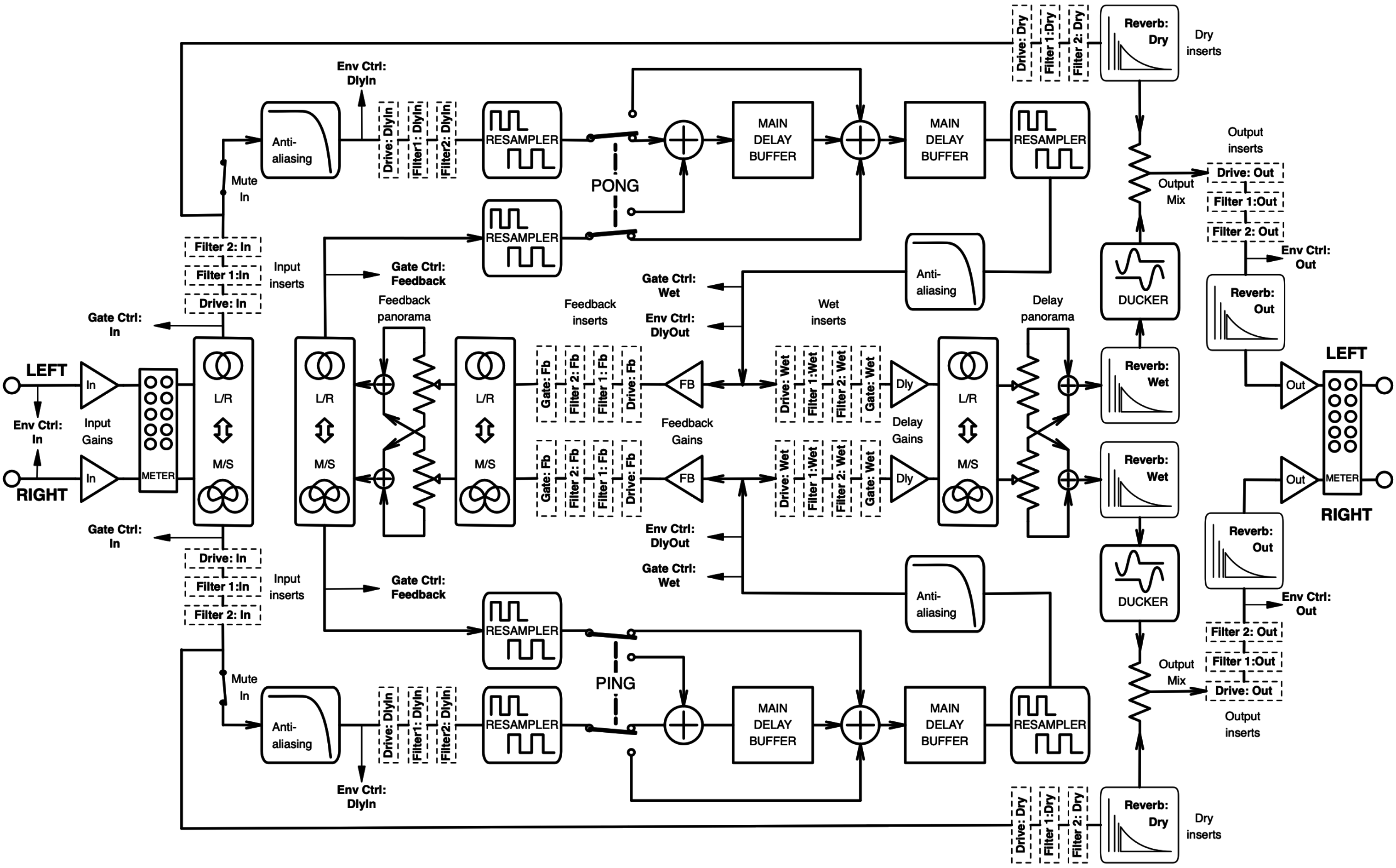
The gate can be placed at various routing points, depending on its intended application. For instance, setting it up on the wet signal lets you control the number of taps of the echo. When set up on feedback with control from the input, the feedback amount can be controlled by the level of the input signal, or by the external sidechain input. One good application for this is to feed a drum track through PSP 285 and set the threshold to only put delay on the downbeats.

Ducker

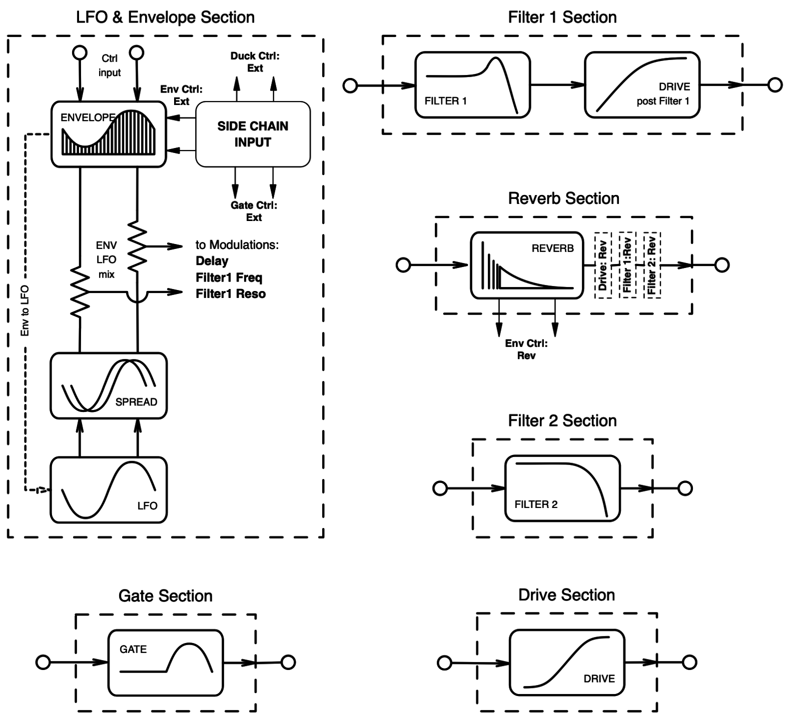
Using dense echo patterns on a track can ruin the clarity of the overall mix. To help prevent this, PSP 285 has a built-in ducker with adjustable time constant and threshold and a wide selection of control signal points.

In simple terms, whenever the control signal (e.g. input signal) is present, the wet signal is attenuated down to –18 dB, and when the control signal disappears, the wet signal comes back to its normal level. This automatically “thins out” the echoes so they don’t become too dense and overwhelm the track or the mix. Since lower-level echoes are barely heard or unheard in a busy mix, this provides clarity and “air” without harming the music.

PSP 285 Block Diagrams



PSP 285 main delay. Alternative module on inserts shown with dash lines.



PSP 285 Alternative control and processing modules.

USER INTERFACE: HOW TO USE PSP 285

NOTE:
The knob or other controls for any module that isn't activated will be greyed out.

Some controls have additional functions, which are accessed via COMMAND-click (macOS) or CTRL-click (Windows).

Main section

BYPASS – bypasses the plug-in. Note that the processing is still performed in the background, even though the results are not heard.
M/S – switches to Mid/Side processing mode.
PSP 285 label – click to show the back panel, which plugin block diagram, accesses this manual and lets you hide or show the tool tips that appear when you hover your mouse on a control.

Mixer section

METERS – show the left and right channel input and output signal levels.
INPUT – adjusts the input gain; this knob either attenuates or boosts the signal prior to any processing.

MIX – sets the proportion between dry (unprocessed) and wet (processed) signals. You can also think of this knob as setting the overall effect depth. Note that PSP 285 uses a -6 dB pan law, which might require input or output boosting to compensate for the gain reduction applied at mixing.

OUTPUT – adjusts the output gain; this knob either attenuates or boosts the output signal.

WET – switches to 100% Wet signal output and no Mix control, for use on effect sends.

LIM – sets the brick wall limiter's output level.

Lim LED – flashes to indicate when the limiter is in operation.

The **LIM** label and the **Lim LED** both act as invisible buttons to toggle the limiter on and off.

PADLOCK – switch between Preset and preset-independent (Global) mode for all mixer parameters:

Padlock open (shown at left) – Preset mode

Padlock closed (shown at right) – Global mode (note all knobs are now red)

Use Command-click (macOS) / Ctrl-click (Windows) to copy visible settings between modes.

Dynamics section

GATE module:

ON – toggles the gate on and off.
THRESH – noise gate threshold. Any signal below this level will be further attenuated. Gating offers additional, more flexible control over the number and level of echoes, closing the gap between standard and multi-tap delays.
Gate LED – indicates that the gate is closed.
SPEED – sets how quickly the gate takes effect, from 50 to 1000 ms.
RT (route) – sets operation points and control signals. The gate can be controlled (triggered) by the input signal (Input ctrl), an external sidechain input signal (Ext ctrl), or the signal at any of several insert points.
Any of these trigger signals can be linked or unlinked (Dual Mono).
The full list of control routings is shown here:

Fb - Feedback
FbIn - Feedback Input ctrl
FbX - Feedback Ext ctrl
Wet
WetIn - Wet Input ctrl
WetX - Wet Ext ctrl
FbD - Feedback Dual Mono
FbInD - Feedback Input ctrl Dual Mono
FbXD - Feedback Ext ctrl Dual Mono
WetD - Wet Dual Mono
WetInD - Wet Input ctrl Dual Mono
WtXD - Wet Ext ctrl Dual Mono

DUCKER module:
ON – toggles the ducker on and off.
DUCKER – ducker threshold. When input signals reach this threshold level, wet (processed) signal will be automatically attenuated.
Ducker LED – indicates when the ducker is active.
SPEED – sets ducker opening speed (time).
CTRL -sets the ducker's control signal. The ducker is always inserted on the wet signal, and can be controlled by the internal audio signal or by an external signal present at the auxiliary sidechain input, if there is one. The Ducker can operate on linked or unlinked (DUAL MONO) channels.
The available CTRL options are shown below:

Wet - Internal ctrl
WetX - External ctrl
WetD - Dual Mono Internal ctrl
WetXd - Dual Mono External ctrl

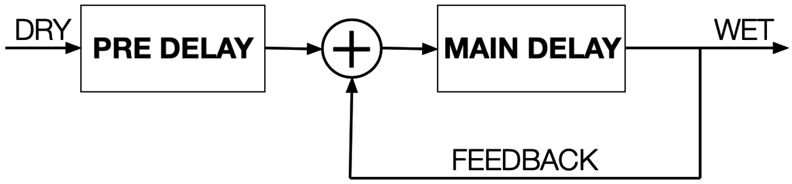
IMPORTANT NOTE: **The Ext Ctrl sidechain option is only available if a sidechain input is connected to PSP 285 from elsewhere in your host application (your DAW software). If Ext Ctrl is selected without such a signal routing present, the routing will default to Input Ctrl.**

Delay section

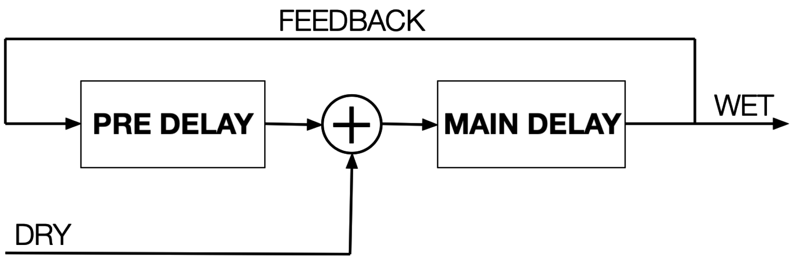
LINK – allows both delay channels to be adjusted at the same time. When engaged, any adjustments will affect both delay channels; note that linked Pan controls mirror one another, so turning one knob to the left turns the other knob to the right.
Holding COMMAND or CTRL causes whatever parameter you're adjusting to momentarily change from linked to unlinked, or unlinked to linked.
L>>R – copies the Left/Mid channel settings to the Right/Side channel.
R>>L – copies the Right/Side channel settings to the Left/Mid channel.
SWAP – When holding COMMAND or CTRL, the L>>R and R>>L buttons became SWAP, which swaps all parameter settings between the two channels.

MUTE IN – mutes the delay line input. On mono tracks, clever muting and feedback panning allows the two delay lines to be used in series.
PANIC – turns off Filter 1 and stops feedback for instant control of unstable "runaway" sounds.

PING and **PONG** – sets up a ping-pong delay effect, by changing the configuration of the Predelay and Main delays' feedback loops as shown:
Let's say you want to set up a ping-pong delay of 125 ms that will start on the left side. Trying to do this with a conventional delay can be tedious and tricky to figure out: you have to set up predelay and main delay on the two sides to produce the appropriate bounce from side to side, without having later echoes turn mono after the initial couple of taps, etc.



Simplified diagram of a single channel Predelay and Maindelay in Normal configuration.

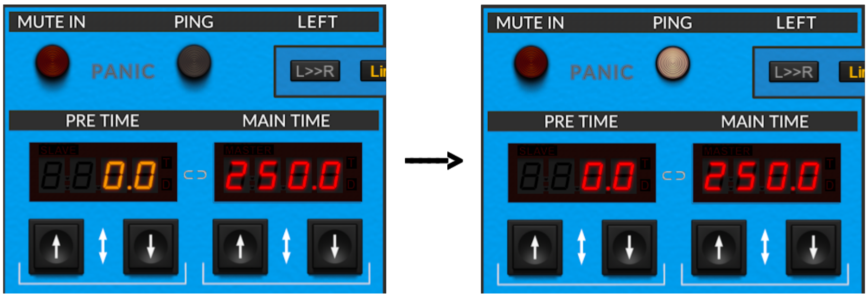


Simplified diagram of a single channel Predelay and Maindelay in Ping or Pong configuration.

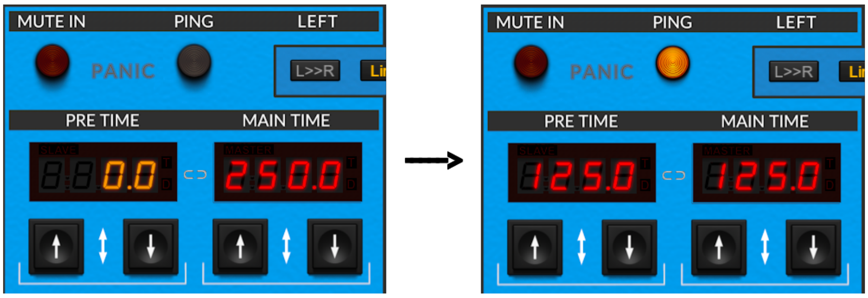
To do this in PSP 285, you can simply use PING on the left channel. This mode changes the left delay's Pre and Main Times *and* reroutes the dry signal to avoid the first predelay. It also resets the right delay's Pre and Main Times, *without* the alternate routing.

NOTE: If you wanted the echo to start on the right side, you would use PONG on the right channel, but the steps are the same.

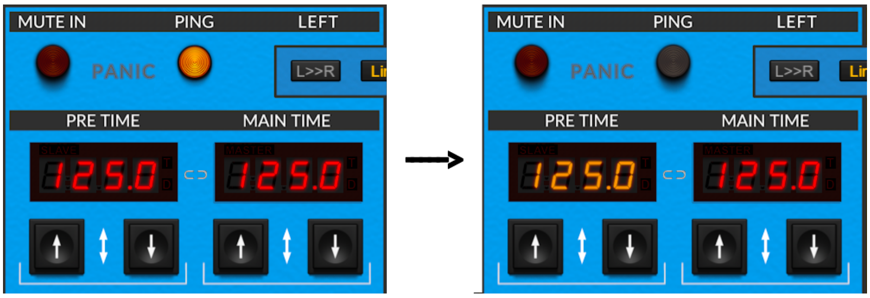
Click on the PING button to engage the Ping-Pong mode. The indicator light is white, and the delay lengths remain unchanged. Note that the Pre Time turns red.



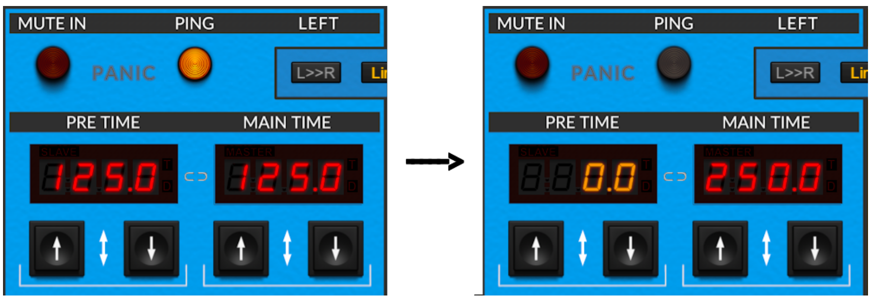
Now, if you COMMAND-click/CTRL-click the button (the indicator light is orange), you can change the configuration while also recalculating the Pre and Main Times to produce the desired effect, as shown:



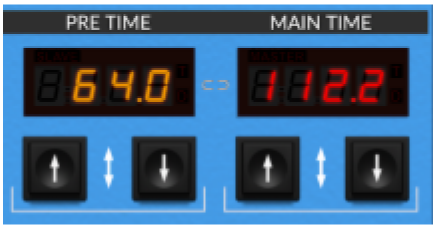
If you switch back to normal mode by clicking on the PING button to turn off the indicator light, the recalculated Pre and Main Times are preserved.



Or, you can COMMAND-click/CTRL-click the button to retrieve your original delay values.



This is a great way to create unusual stereo multi-tap rhythms: just use PING or PONG to set up a basic effect, then individually tweak the Pre and Main Times for the Left and Right delays to create the rhythms you want. Try experimenting with turning PING and PONG off and on in various configurations, to hear how the various echoes change position in the stereo panorama: sometimes on the left, sometimes on the right, sometimes in the center.



DISPLAY – When a delay time parameter is being edited, its value is displayed here as a time in milliseconds or as a note value, with T to indicate triplets or D to indicate dotted notes. Click on it to directly type in a time value in milliseconds, or to pop up a menu with note values to choose from. Pressing the SHIFT key gives you a wider set of note values to scroll through.

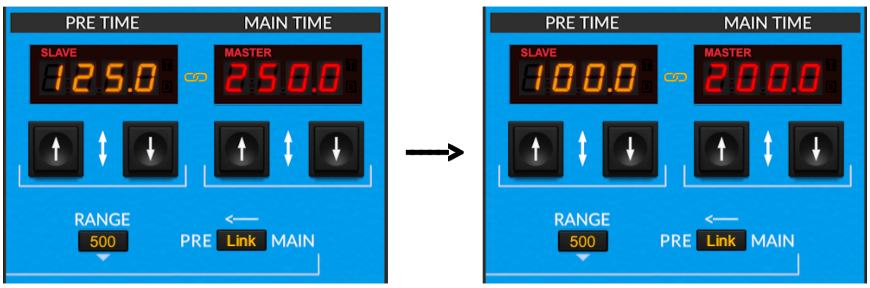
PRE TIME – Predelay time, applied to the input signal before it enters the feedback loop.

MAIN TIME – Main delay time, which is also the length of the feedback loop. With zero feedback, a single echo will be heard after (PRE TIME + MAIN TIME).

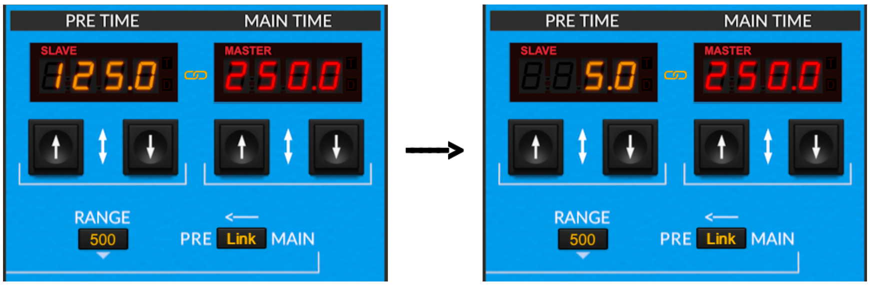
UP/DOWN – These arrows turn delay time up or down in 1 ms increments. Hold down SHIFT to increment by 0.1 ms for finer control. The icon with the up and down arrows between the two buttons is an “invisible slider” that lets you click and drag for very fast coarse time adjustments. You can also hover your mouse over the display and use its scroll wheel.

RANGE – chooses a delay time range (only active in Time mode). You can choose between 50 ms, 500 ms, and 5000 ms (5 seconds). Smaller ranges allow for finer delay time adjustments.
MODE – delay time mode. Two modes are available:
Time – the time is expressed in milliseconds and doesn't follow tempo changes
Note – the time is expressed as a note value (rhythmic figure) related to tempo
The delay time settings in milliseconds (Time mode) and note values (Note mode) are independent of one another.
COPY TO – copies the delay time setting from one mode to the other. Copying a note value to a time in milliseconds will be exact; copying a time in milliseconds to a note value will not, as the value will be quantized to the nearest note value at the current tempo.
LINK – links the Pre Time to the Main Time in a fixed proportion.
The chain link icon between the displays turns linking on/off and is illuminated when turned on. The time displays will say SLAVE and MASTER when linked.

When you change the Main Time, the Pre Time will change proportionally. When you change the Pre Time, the Main time stays unchanged.



EXAMPLE 1: If the Pre Time is set to 1/2 of the Main time, then changing Main Time from 250 to 200 changes the Pre Time from 125 (1/2 of 250) to 100 (1/2 of 200).



EXAMPLE 2: Changing the Pre Time doesn't alter the Main Time; instead, it changes the proportion between Pre and Main. Here, changing the Pre Time from 125 to 5 changes the proportion from 1/2 (125/250) to 1/50 (5/250). Now, if you change the Main Time, the Pre Time will adjust so it remains 1/50 of the Main Time.

FB-PAN – feedback return panning. Can be used to route part or all of the feedback signal to the other delay channel.

FEEDBACK– the gain reduction applied to the feedback signal on each circulation. Controls the number and level of echoes.

PAN (DELAY) – delay output panning. Each delay line (channel) can be panned independently.

In Mid/Side mode, both controls operate as ordinary Left/Right pans.

NOTE: In the Mid/Side mode, both pan controls work as ordinary Left-Right panpots.

GAIN – delay output gain.
Feedback INV – inverts the phase of the feedback return.
Delay INV – inverses the phase of the delay signals.
Both of the INV settings are usually only audible when delay time is short. They are useful for chorus/flanger effects.

Delay Control section

MANUAL – directly changes the delay line's sample rate. This knob not only changes the delay time, but also results in an abrupt pitch shift, which will be audible as long as the audio in the delay buffer is played back at a speed different from the one at which it was captured.

DLY MOD – sets the delay line sample rate modulation depth.

ON (DLY MOD) – turns delay line sample rate modulation on or off.

Modulation (LFO & ENV) section

Low Frequency Oscillator

MODE – LFO operating mode; three different modes are available:

- **Hertz** – free-running; rate set in Hertz
- **Note** – free-running; rate expressed as tempo-related note values (rhythmic)
- **Note+** – like Note mode, but the phase stays locked to the song position.

RATE (display) – LFO rate in Hertz or as a note value, with T to indicate triplets or D to indicate dotted notes. Click on it to directly type in a time value in milliseconds, or to pop up a menu with note values to choose from. Pressing the SHIFT key gives you a wider set of note values to scroll through.

UP/DOWN – These arrows turn delay time up or down in 1 ms increments. Hold down SHIFT to increment by 0.1 ms for finer control. The icon with the up and down arrows between the two buttons is an “invisible slider” that lets you click and drag for very fast coarse time adjustments.

SHAPE – LFO waveform. There are six waveforms available: sine, square, triangle, sawtooth, random with sharp stepping, and random with smoothed transitions between steps.

SPREAD – phase offset between the two LFO channels. When modulating the delay line sample rate, Spread determines the phase offset between the left and right channels. When modulating the filter, Spread sets the phase offset between cutoff and resonance modulation.

OFFSET – Sets the LFO offset (starting point) relative to the start of the song. Only active in Note+ mode.

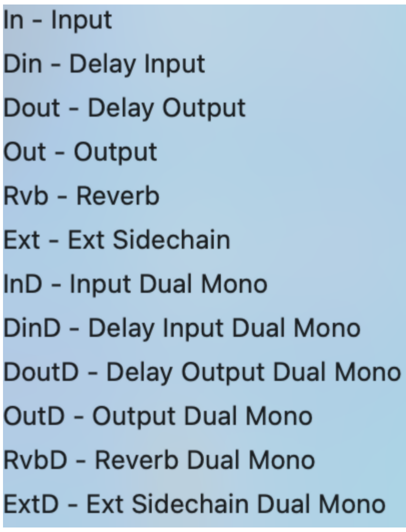
LFO – switches the LFO section on/off.

MOD SOURCE – sets the proportion of LFO vs envelope follower signal in the summed modulation output signal. The LEDs indicate the values of the summed modulated signal for each channel, Left/Mid and Right/Side.

Envelope Follower

SENS – sets the envelope detector sensitivity. This control pre amplifies the signal before envelope extraction.

SPEED – controls the envelope detector’s time constant. This determines the attack and release times of the main envelope detector (used for modulation purposes), as well as the release times of the dedicated envelope detectors for the gate and ducker.



CTRL- sets the signal source for the envelope detector.

- **In** – overall plug-in input
- **Din** – delay buffer input. The signal routing in the plug-in determines the effect, if any, of the anti-aliasing filters and other elements in the signal path.
- **Dout** – pure delay buffer output (including feedback processing)
- **Out** – overall output of the plug-in
- **Rev** – reverb output
- **Ext** – external side-chain signal

These signals are available linked and unlinked (Dual Mono).

AT-RL – sets the ratio between the envelope follower's attack and release times. To the left of the center setting, the attack time is shorter and the release time is longer; to the right of center, the attack time is longer and the release time is shorter.

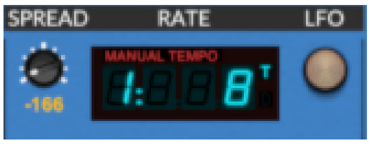
ENV->LFO – Sets how much the envelope follower modulates the LFO speed.

Env – switches the Envelope Follower section on or off.

Manual Tempo Control

Normally, your host application (DAW) will provide a base tempo for PSP 285, and all of the Note values for delay times and LFO rates will be aligned with that tempo. If you hold down COMMAND or CTRL and hover your mouse over the LFO RATE display or arrow buttons, it will turn red and read HOST TEMPO, showing the tempo set in your DAW. You can't manually set this number, only change it in your DAW settings.

However, if your DAW isn't providing tempo data, the RATE display will show the words MANUAL TEMPO. This is the only time when you can set the base tempo manually; it's not available when your DAW is in control.



As stated above, if you hold down COMMAND or CTRL and hover your mouse over the LFO RATE display or arrow buttons, it will turn red and read MANUAL TEMPO. Now, however, you can change the tempo with the Up/Down buttons, the invisible slider, or your mouse scroll wheel, as long as you're holding down COMMAND or CTRL.

Filters section

Filter 1:

ON – toggles Filter 1 on/off.

ROUTE – chooses the filter's signal routing (see **AltRt** below)

TYPE – filter type: selects one of 17 available filter types, as shown in the list on the facing page.

FREQ – sets the cutoff frequency. When band-pass or notch filter type is selected, this knob sets the center frequency.

FREQ MOD – cutoff (center) frequency modulation depth.

RESO – sets the filter resonance. i.e. the level of the response peak around the cutoff frequency. High values will result in self-oscillation, and when combined with delay feedback can lead to dangerously high sound levels. Remember how to find the PANIC button!

MOD – sets the resonance modulation depth.

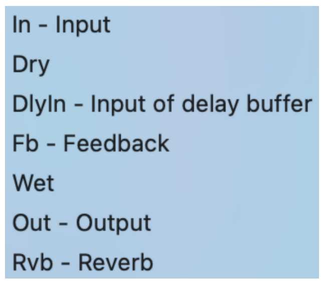
AltRt (alternate modulation routing) – turns alternate modulation routing on/off.
With AltRt turned off, both channels of the modulation signal are directed to their corresponding Filter channels' cutoff frequency and resonance.
With AltRt turned on, Channel Left/Mid modulation controls filter cutoff frequency, and channel Right/Side controls filter resonance. When phase offset is added via LFO Spread control, this can produce dramatic effects.

Filter 2:

ON – toggles Filter 2 on/off.

ROUTE – Sets the location of the filter within the plug-in signal chain:

- IN – filtering is applied to the input signal prior to any further processing.
- Dry – filtering is applied to the dry signal.
- DlyIn – filtering is applied to the delay buffer's input signal.
- Fb – filtering is applied to the feedback signal, just before its return.
- Wet – filtering is applied to the wet signal.
- Rev – filtering is applied to the reverb output signal.



TYPE – selects one of 12 filter types, as shown in the list on the facing page.

FREQ – sets the cutoff frequency. When a band-pass or notch filter type is selected, this knob controls the center frequency.

Filter 1 types:

- **LPF** - Low Pass Filter,
- **LPFs** - semi Low Pass Filer, with 18 dB shelf,
- **BPF1** - Band Pass Filer type 1,
- **BPF2** - Band Pass Filer type 2,
- **Ntch** - Notch Filer,
- **HPFs** - semi Hi Pass Filer, with 18 dB shelf,
- **HPF** - Hi Pass Filter,
- **APF** - All Pass Filter,
- **svLPF** - state variable Low Pass Filter,
- **svBPF** - state variable Band Pass Filter,
- **svHPF** - state variable Hi Pass Filter,
- **lLPF** - ladder 24 dB/oct Low Pass Filter,
- **lBPF** - ladder 24 dB/oct Band Pass Filter,
- **lHPF** - ladder 12 dB/oct Hi Pass Filter,
- **gLPF** - gained Low Pass Filter,
- **gBPF** - gained Band Pass Filter,
- **gHPF** - gained Hi Pass Filter,

Filter 2 types:

- **LPF** - Low Pass Filter,
- **LPF+** - Low Pass sharp Filter,
- **LPFs** - semi Low Pass Filer, with 18 dB shelf,
- **LPFs+** - semi Low Pass sharp Filer, with 18 dB shelf,
- **BPF** - Band Pass Filer,
- **BPF+** - Band Pass sharp Filer,
- **Ntch** - Notch Filer,
- **Ntch+** - Notch sharp Filer,
- **HPFs** - semi Hi Pass Filer, with 18 dB shelf,
- **HPFs+** - semi Hi Pass sharp Filer, with 18 dB shelf,
- **HPF** - Hi Pass Filter,
- **HPF+** - Hi Pass sharp Filter,

DRIVE section

ON – toggles the Drive module on/off.

DRIVE – sets the amount of internal filter saturation. Low values will add harmonics, while higher settings will result in audible distortion.

DRV TYPE – opens a popup menu with a choice of eight different insert points for the Drive module, each with either hard or soft saturation.

The first choice is Post Filter Mode, where the Drive module distorts the Filter 1 output. In all of the other insert modes, the Drive module is inserted before both filters.

Reverb section

ON/OFF – toggles the reverberator on/off.

TYPE – selects the reverb algorithm and signal routing from this popup list:

- Amb - Ambience
- Spr1 - Spring 1
- Spr2 - Spring 2
- Plt1 - Plate 1
- Plt2 - Plate 2
- Chbr - Chamber
- Room
- Hall
- Rvrs - Reverse
- AmbD - Ambience Dual Mono
- Spr1D - Spring 1 Dual Mono
- Spr2D - Spring 2 Dual Mono
- Plt1D - Plate 1 Dual Mono
- Plt2D - Plate 2 Dual Mono
- ChbrD - Chamber Dual Mono
- RmD - Room Dual Mono
- HallD - Hall Dual Mono
- RvrsD - Reverse Dual Mono

Each of the algorithms can process linked channels (mono to stereo) or process the two input channels independently (Dual Mono), a mode which is especially intended for use with ping-pong effects.

DAMP – adjusts the high frequency damping.
Greater values will result in stronger high frequency damping and faster high frequency attenuation for an overall darker sound.

TIME – sets the reverberation decay time.

BLEND – determines the wet/dry mix for the reverb itself as opposed to the delay, filter, etc..

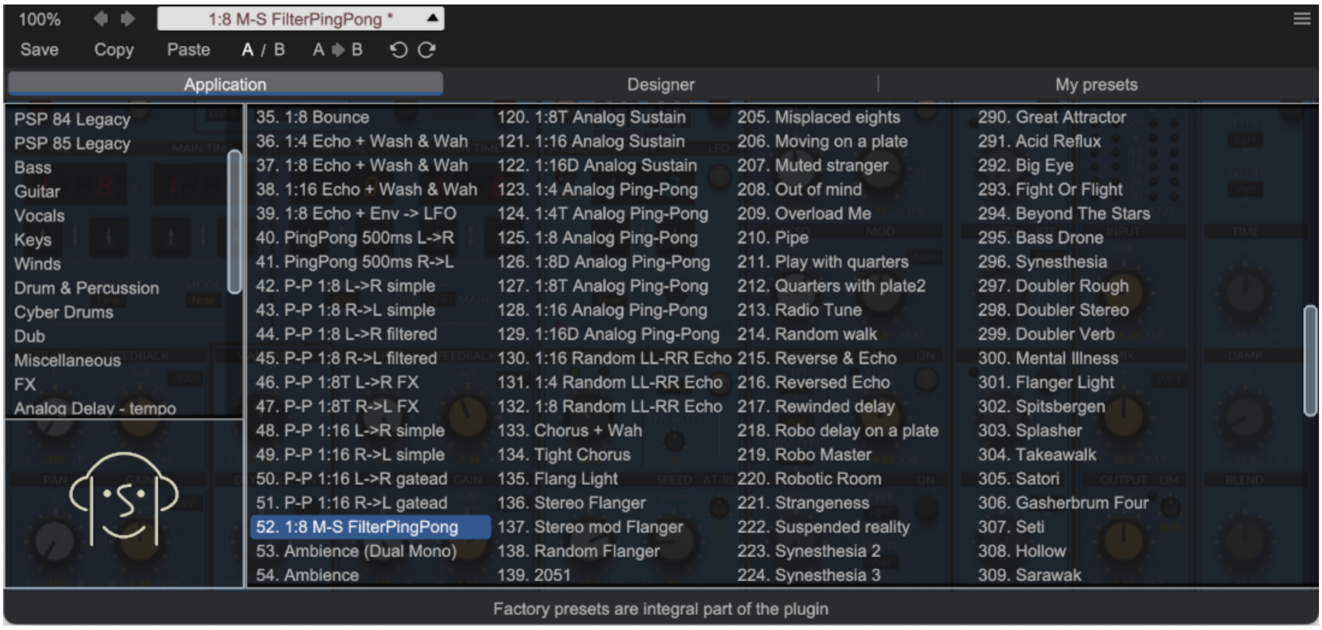
INSERT – sets the reverb insert point. Options include:

- **Dry** – reverb is applied to the dry (undelayed) signal
- **Wet** – reverb is applied as the last processing step just before wet/dry mixing
- **Out** – reverb is applied after the wet (delayed) and dry (undelayed) output mixing

PRESET HANDLING AND VIEW OPTIONS

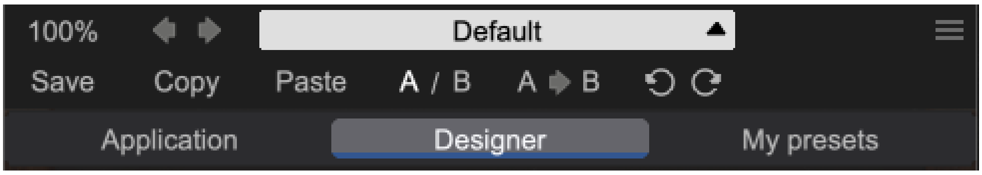
Every PSP plug-in comes with a large library of factory presets. You can use them as a starting point for experimenting with your own sounds, examine them to understand how the various features work, or keep them handy for when a track or mix needs a quick and high-quality way to create an effect or fix a problem.

To access the preset library, just click on the Preset Bar along the top of the plug-in window. If you're familiar with other PSPaudioware plug-ins, you'll find that this one works exactly the same way.



Preset Browser

PSP 285 features a comprehensive preset management and browser system. To access the preset browser, simply click on the preset name window at the top of the plug-in (which displays 'Default' when the plug-in loads).



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

Application – shows all factory presets, sorted by application or type of effect. These can be selected from a list on the left side of the preset browser. Applications include: Basic Delay, Ping-Pong, PSP 84 Legacy, PSP 85 Legacy, Bass, Guitars, Vocals, Keys, Winds, Drum & Percussion, Cyber Drums, and Dub.

Designer – shows all factory presets, sorted by designer. A photo of the designer is displayed for each of their presets. Click on the photo to open the designer's website.

My presets – shows only the presets you have created and saved, or downloaded and added to your custom presets for PSP 285.

NOTE: The Factory presets are built into PSP 285. While you can't edit them directly, you can make adjustments to them, and then save the result as a user preset.

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 285

For Mac users, this is located at:

~/Documents/PSPaudioware.com/User Presets/PSP 285

NOTE: You can find the exact file location by clicking on the **Show File in Finder** button at the bottom of the preset browser window.

To select a preset, simply click a preset name in the right window. On the first click, the preset will be temporarily loaded so that you can audition it while still in the preset browser. To confirm the preset choice and get back to the main user interface, double-click the preset name again.

Copy / Paste

The **Copy/Paste** feature is useful for when you're running two or more instances of PSP 285 and you want them to have identical settings.

Of course, you can always open a new instance and load the same preset as your first instance has, but this only works if your first instance hasn't been tweaked at all since the preset was loaded. To share your tweaks between instances, use **Copy** and **Paste**.

To use this feature, simply click the **Copy** button, open a new instance of PSP 285 where it's needed, and click the Paste button to load the first instance's settings.

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

A/B System

The **A/B system** lets you quickly audition changes to your settings. You can compare how different tweaks work in a track or mix, or even audition two different presets on the fly.

The **A/B Button** allows you to quickly switch between the current plug-in settings (**A**) and a previous group of settings that you've previously stored (**B**).

The **A>B Button** copies the **A** settings over to the **B** slot. This lets you temporarily 'bookmark' your current settings, make more tweaks, and then compare the new tweaks with your 'bookmarked' settings using the **A/B Button**.

Undo / Redo

The **Undo/Redo** feature can be extremely important when designing presets! We all know the frustration when we make one too many edits and ruin a previously great sound. With the **Undo** and **Redo** buttons (the counterclockwise and clockwise arrows as shown above), you can step backward and forward through your edit actions until you're back where you wanted to be.

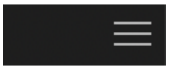
These buttons will let you undo a preset selection, returning you to your previous preset with all settings as they were when you stopped editing it.

GUI resizing

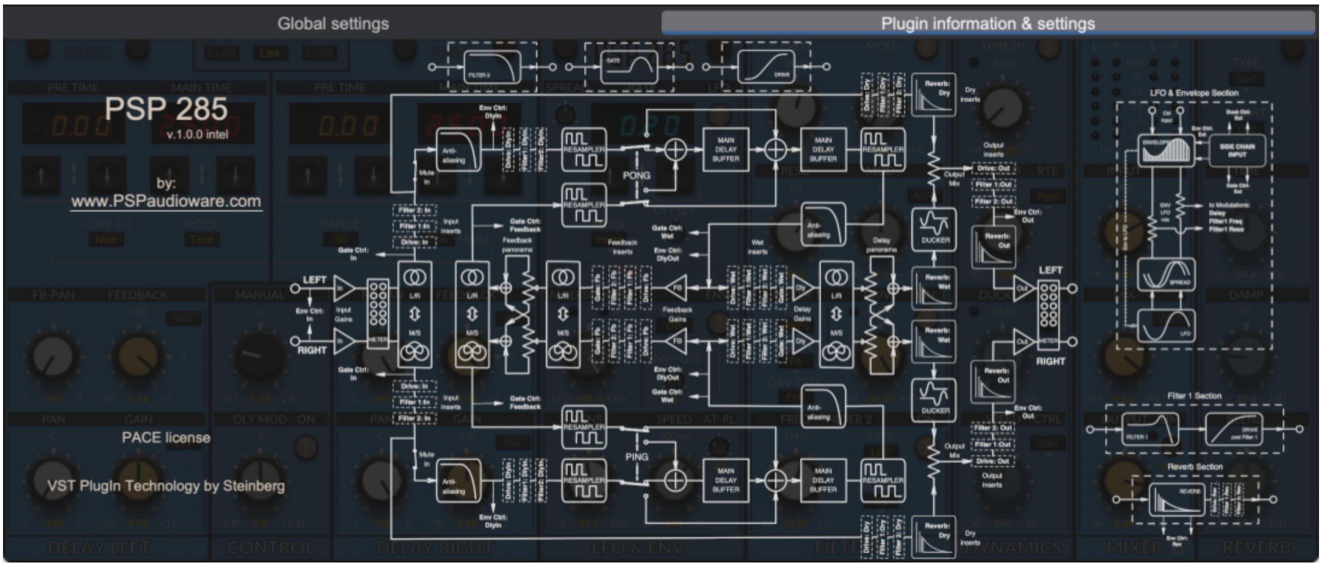
The percentage at the top left shows the current user interface size. Click on it to reveal a dropdown menu of size choices, or hover your mouse on it and scroll up and down to change the size quickly. Double-click to reset it to the default size (100%).

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

Config section



Click the icon with three parallel lines in the top right corner to open the **CONFIG** menu. You will find controls to open the manual, hide or show mouse-hover tool tips (Hints), check your current plugin version with build number and show back panel



The 285 back panel.

Minimum System Requirements

In order to run PSP BinAmp you need to install the free **iLok License Manager** application but you don't need any hardware dongle. By default we provide 3 licenses which can be activated in 3 separate locations, each of which can be either a computer or an iLok dongle (2nd generation or above). You can move these licenses at any time using PACE's iLok License Manager software.

Windows

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

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

AAX
- Windows 7 – Windows 11
- 64-bit Pro Tools

All DAWs
- Up to date iLok License Manager application installed

macOS Intel or macOS AppleSilicon

AudioUnit
- macOS 10.14 – macOS 14 Sonoma
- 64-bit AudioUnit compatible host application

VST
- macOS 10.14 – macOS 14 Sonoma
- 64-bit VST3 compatible application

VST3
- macOS 10.14 – macOS 14 Sonoma
- 64-bit VST3 compatible host application

AAX
- macOS 10.14 – macOS 14 Sonoma
- 64-bit Pro Tools

All DAWs
- Up to date iLok License Manager application installed

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Processing

- All internal processing done with 64-bit double precision floats.
- PSP 285 supports 32 and 64 bit floating point audio streams.
- PSP 285 supports sample rates up to 384 kHz.

Limitations of the demo version

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

Enjoy !

PSP team

Support

f 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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