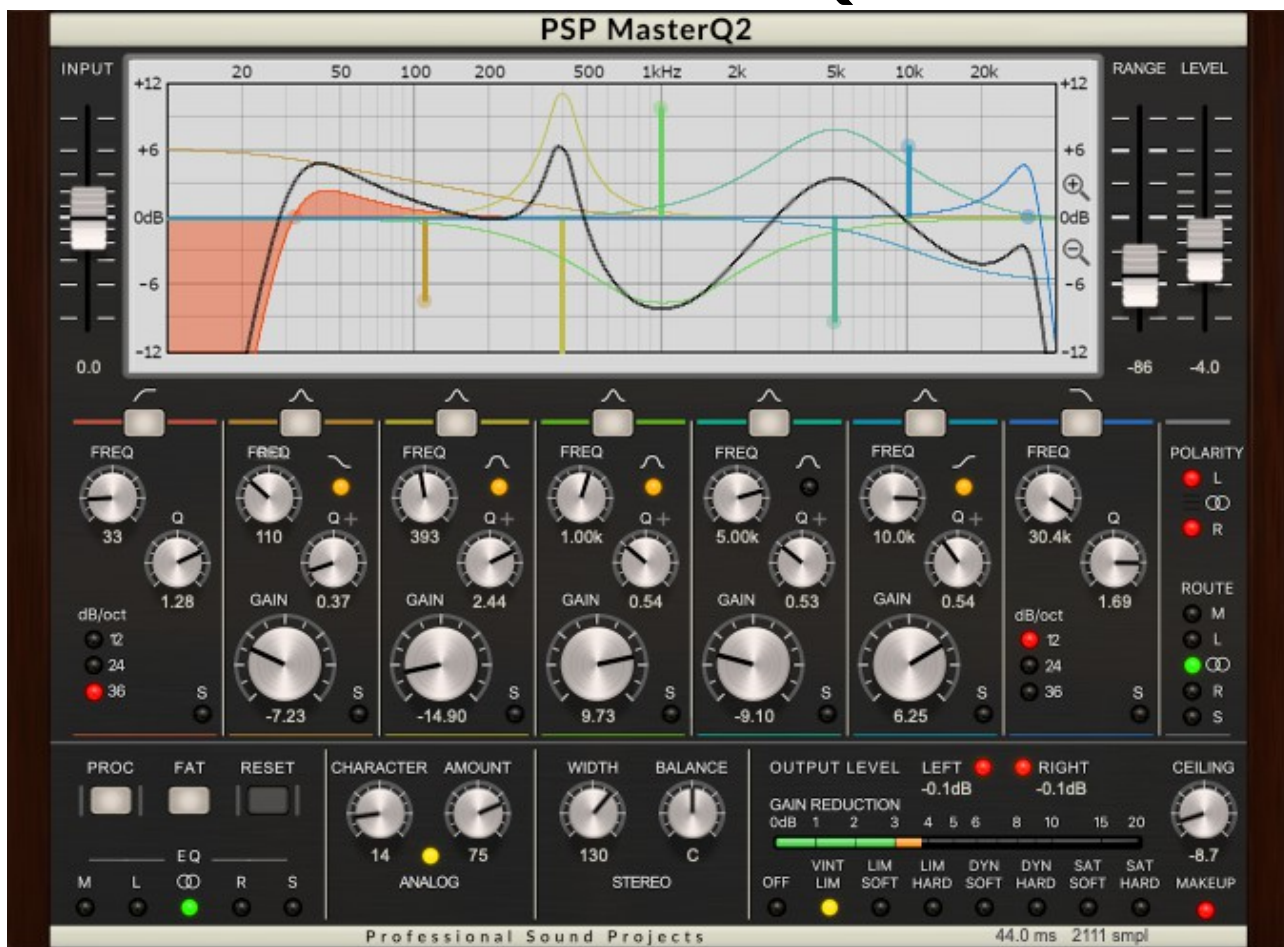


PSP MasterQ2



Operation manual

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Foreword

by Sakis Anastopoulos (Digisound Mastering)

<http://www.digisoundmastering.com>

Sound shaping is an art that the audio engineer must learn and understand well. There is a plethora of sound sculpturing tools in the market with various versions and flavors available, allowing us to get the sound that we want out of our project. Some tools will give you excellent results and some of them very poor or average results. What sets them apart is the ability of the unit to provide the engineer with the results that he/she has in mind.

The PSP MasterQ is a parametric EQ that provides you with excellent sonic results when you include it in your project. Is capable of high sampling rates (up to 768 kHz) and is designed for the audio professional who wants to take full control of the frequency spectrum.

Before using this fine tool I would highly recommend that you read and understand the manual first. There are many features and controls that you'll need to master before you use them. These vary from the different filter shapes to the limiting and saturation algorithms. The results that you can achieve from this EQ will depend on how well you understand and use its features. The manual that is provided with the plug-in is very well written and covers every single knob, button and function of the EQ.

I would like to share some tips with you before you start using this EQ.

Make small adjustments in gain when you boost or cut. If you are adjusting the right frequency with the correct bandwidth you don't need to apply a big amount of gain (unless you are trying to remove hum or other offending frequencies). Make sure that before you adjust the bass (especially frequencies below 40Hz) that your monitoring setup is capable of reproducing these frequencies effortlessly. Most projects that I receive for mastering suffer in the bottom end of the spectrum.

When you have found the right frequencies and bandwidth apply a small amount of gain in each frequency area and then use the range knob to fine tune the gain in these areas. The range knob will assist you to make decisions fast and will become your best friend during decision making.

When you are working in high sampling rates (96-768 kHz) and your final medium of delivery is CD, make sure that the adjustments that you make to your project are going to "survive" the severe degradation of the sampling rate conversion process, especially if you don't have the best tools that are available for this process.

PSP MasterQ2 supplies a very handy set of presets to use as starting points. Please use those to get an idea of what this plug in is capable of. These presets are not going to be the answer to all your problems, although some will help you achieve the desired results in some situations. Each project that you are working on has its own sound signature and there isn't a preset that will make all your projects shine. You'll need to adjust the gain, frequency and bandwidth for each project individually to get optimal results.

Good luck with your new plug-in and I hope that you'll end up using it, as I do, in every project that I work on.

The Role of Equalization in Mixing

by Craig Anderton (EQ magazine)

There's a reason why almost every mixer has built-in EQ: Next to level control, control over timbre is the most important element in creating a satisfying mix. However, too many people adjust equalization with their eyes, not their ears. For example: after doing a mix recently, I noticed the client writing down all the EQ settings I had made. When I asked why, he said it was because he liked the EQ and wanted to use the same settings on these instruments in future mixes. But EQ doesn't work that way, because it is part of the mixing process. Just as levels, panning, and reverb are different for each mix, EQ should be customized as well. To do that, you need to understand how to find the magic EQ frequencies for individual tracks and particular types of musical material, as well as how to use EQ in the context of a full mix involving multiple tracks.

When mixing, there are three main applications for EQ:

1. Solving problems
2. Emphasizing or de-emphasizing an instrument in a mix
3. Altering a sound's personality

Each application requires specialized techniques and approaches.

Solving problems

EQ can fix some obvious problems. As one example, slicing a sharp notch at 60 Hz (50 Hz in Europe) can knock hum out of a signal; trimming the high frequencies can remove hiss. Generally, problems occur in specific frequency ranges, which makes the parametric equalizer an ideal solution. Another common problem is an instrument with a resonance or peak that interferes with other instruments, or causes level-setting difficulties. The following procedure illustrates how to use a parametric EQ to deal with this issue.

I've recorded several albums with classical guitarist Linda Cohen, who has a beautiful instrument with a full, rich sound that projects very well on stage. This is due to a strong body resonance in the lower midrange that causes a major level peak. However, recording is a different matter compared to playing live. Setting levels so that the peaky, low frequency notes don't overload the recorder causes the higher frequency guitar notes to sound weak by comparison.

Although compression/limiting is always an option, it alters the guitar's attack. While this effect might be masked when playing en ensemble, it sticks out with a solo instrument. A more natural sounding answer is to use EQ to apply a frequency cut that's more or less equal and opposite to the natural boost, thus evening out the response. But there's a trick to finding problem frequencies so you can alter them; the following works extremely well.

1. Turn down the monitor volume — the sound might get nasty and distorted during the following steps.
2. Set the EQ for lots of boost (10-12 dB) and fairly narrow bandwidth (around a quarter-octave or so).
3. As the instrument plays, slowly sweep the frequency control. Any peaks will jump out due to the boosting and narrow bandwidth. Some peaks may even distort.
4. Find the loudest peak, then cut the amplitude until the peak falls into balance with the rest

of the instrument sound. You may need to widen the bandwidth a bit if the peak is broad.

This technique of boost/find the peak/cut can help remove midrange “honking,” reduce strident resonances in wind instruments, and much more. Of course, sometimes you want to preserve these resonances, or only reduce them a tiny bit. But often, applying EQ to reduce peaks allows instruments to sit more gracefully in the track.

Parametric EQ is particularly effective for this type of application due to its precision. In one of my more unusual projects, I needed to remove boat motor noise from some whale samples I had recorded in Alaska. Motor noise is not broadband, but exists at multiple frequencies. Applying several extremely sharp and narrow notches at different frequencies took out each component of the noise, one layer at a time, until the motor noise vanished.

This type of problem-solving also underscores a key principle of EQ: It’s often better to cut than boost. Boosting uses up headroom; cutting opens up headroom. In the example of solving the classical guitar resonance problem, cutting the peak allowed bringing up the overall gain, which allowed mastering at a much higher level - without using any kind of compression or limiting. (If you do this while recording, you can also record at a higher level.)

Emphasizing instruments

The same technique of finding and cutting specific frequencies can also eliminate “fighting” between competing instruments. For example, while mixing a Spencer Brewer track for Narada records, there were two woodwind parts with resonant peaks around the same frequency. When playing together they would load up that part of the frequency spectrum, which also made them difficult to differentiate. Here’s a way to work around this:

1. Find, then reduce, the peak on one of the instruments to create a more even sound.
2. Note the amount of cut and bandwidth that was applied to reduce the peak.
3. Using a second stage of EQ, apply a roughly equal and opposite boost at either a slightly higher or slightly lower frequency than the natural peak.

Both instruments will now sound very articulated, and because each peaks in a different part of the spectrum, they will tend not to step on each other.

New sonic personalities

EQ can also change a sound’s character - for example, turn a brash rock piano sound into a darker, more classical sound, or increase the upper midrange on bass so it has more of a “funk” characteristic. This type of application requires relatively gentle EQ, possibly at several different points in the audio spectrum. But really, there are no rules. As the whole idea is to create something new and different, use your ears to judge whether the EQ you’re adding creates the desired results.

EQ as part of the mixing process

So far, we’ve mostly discussed applying EQ to individual tracks. But we also need to consider how EQ works within the context of many tracks playing at once, as EQ can help dramatize differences between instruments and create a more balanced overall sound.

In general, problem-solving and character-altering EQ should be applied early on in the mixing process, as they will influence how the mix develops. But wait to apply most EQ until the level setting process begins. Remember, EQ is all about changing levels — albeit in specific frequency ranges. Any EQ changes you make will alter the overall instrumental balance.

Another reason for waiting a bit is that instruments EQ'ed in isolation to sound great may not sound all that wonderful when combined. If every track is equalized to leap out at you, there's no room left for a mix to "breathe." Also, you will probably want to alter EQ on some instruments so that they take on more supportive roles. For example, during vocals consider cutting the midrange a bit on supporting instruments (e.g., rhythm guitar) to open up more space in the audio spectrum for vocals.

When you start applying EQ, you'll likely work on the most important song elements first (typically vocals, drums, and bass) and once these all "lock" together, deal with the other parts. As you do this, remember that the audio spectrum has only so much bandwidth, and you need to make sure that each sound occupies its own turf without fighting other parts. Processing added to one track may affect other tracks; for example, if you boost a guitar's midrange, it may interfere with vocals, piano, or other midrange instruments. If you add more treble to a bass part so that it cuts better on little speakers, make sure it doesn't start fighting with the low end of a rhythm guitar part.

Sometimes boosting a frequency for one instrument implies cutting the same region in another instrument, as with the example given earlier involving vocals.

One common mistake I hear with recordings done by singer/songwriters is that they (naturally) feature themselves in the mix, and worry about "details" like the drums later. However, as drums cover so much of the audio spectrum (from the low frequency thud of the kick to the high frequency sheen of the cymbals), and as drums tend to be so upfront in today's mixes, it's usually best to mix the drums first, then find "holes" in which you can place the other instruments. For example, if the kick drum is very prominent, it may not leave enough room for the bass. So, boost the bass around 800 to 1,000 Hz to bring up some of the pick noise and brightness. This is mostly out of the range of the kick drum, so the two won't interfere as much.

Try to think of the song as a spectrum, and decide where you want the various parts to sit, and their prominence (see illustration). Consider using a spectrum analyzer if you're new to mixing, not because your ears don't work well enough for the task, but because it provides feedback (and valuable ear-training) on which instruments take up which parts of the audio spectrum. This can often alert you to a level buildup in a particular region.

Different instruments take up different parts of the frequency spectrum.

If you really need a sound to "break through" a mix, try a slight boost in the 1 to 4 kHz region. Just don't do this with all the instruments; the idea is to use boosts (or cuts) to differentiate one instrument from another.

To place a sound further back in the mix, rolling back the highs may be able to do the job by "dulling" the sound somewhat. Also, gently reducing low frequencies on instruments that veer toward the bass range, like guitar and piano, can help open up more space for the all-important bass and kick drum.

Finally, remember that EQ often works best when applied subtly. Even one or two dB of change can make a significant difference. However, inexperienced engineers often do something such as increase the bass too much, which makes the sound too muddy, so they increase the treble, and now the midrange sounds weak, so that gets turned up...you get the idea. One of your best "reality checks" is an equalizer's bypass switch. Use it often to make sure you haven't lost control of the original sound.

Now go mix some great music!

Overview

PSP MasterQ2 is a high quality parametric equalizer designed to operate at sampling rates ranging from 44.1 to 768kHz. The plug-in contains seven filters: 12, 24, and 36 dB/oct low cut and high cut filters; low shelf and high shelf filters which can be switched to peaking filters; and low-mid, middle and high-mid peaking filters—all with adjustable frequency and Q factor over a wide range. The plug-in also contains a high-precision EQ graph to display an overview of the EQ curve, as well as the individual characteristics of each filter.

High precision computations

The PSP MasterQ2 employs 80-bit extended precision floating point computations from input to the output to guarantee accurate low frequency filtration and great headroom across the entire audio spectrum. This results in a considerable increase in computation precision (compared to 32-bit floating point and 24- and 48-bit fixed point computations) and processing transparency. It also allows for optimal filter operation at even the lowest frequencies with high gain and Q factor settings.

Frequency Authentication Technique (FAT) Algorithm

The PSP MasterQ2, uses our proprietary Frequency Authentication Technique algorithm. FAT is a double sampling technique which allows for proper filter operation at the highest frequencies. FAT utilizes high order linear filters to prevent phase errors and linear errors from the audio bandwidth caused by double order sampling. This technique results in double sampling without sample rate conversion artifacts.

Most Infinite Impulse Response (IIR) filters typically used in EQs tend to focus on phase and linear digital signal errors in the higher octaves. Our FAT algorithm adds an octave above the Nyquist frequency (the Nyquist frequency is measured by sample rate / 2) and shifts phase and linear errors to that frequency region. This frequency region is then truncated just before the output section of the plug-in, meaning the phase and linear errors are removed from the signal. In other words, FAT gives sonic results approaching that of a sampled analog equalizer rather than a typical digital set of filters.

LIM-SAT

In addition to its powerful equalizer filters, the PSP MasterQ2 also offers a selection of limiting and “soft-clipping” algorithms to help shape the signal and keep it from digitally peaking.

One of the LIM-SAT algorithms, VintLim, is based on the limiter from PSP VintageWarmer’s single band mode. It provides the most transparent limiting algorithm, but at the price of slightly increased internal latency and CPU usage. LimSoft and LimHard are limiting algorithms with the same attack as in DynHard algorithm, but the Release time is much greater, which usually results in a softer feel and sound. DynSoft and DynHard offer dynamic saturation—these modes work like a limiter with material containing levels considerably below 0dBFS, and tightens the attack and release times when the level of the material is close to or exceeds 0dBFS for a more immediate response. SatSoft and SatHard are saturation algorithms from other PSP dynamics plug-ins.

The VintLim algorithm is designed to provide good results whenever artifact free processing is desired. Lim and Dyn algorithms are best for drums and percussive instruments while Sat algorithms are good whenever a pinch of distortion is required.

Controls

The graphic user interface of the PSP MasterQ2 is both elegant and functional, offering a convenient and aesthetically pleasing layout which gives you easy access to all of its functions.



Basic Operation

To activate the buttons, simply click them. They will glow yellow to indicate that they are active.

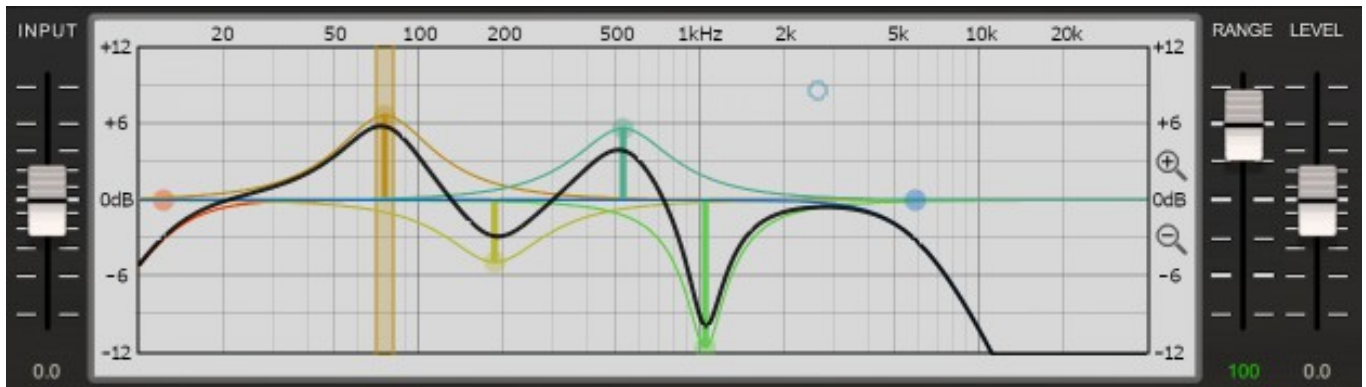
To avoid accidental resets, the Reset button requires double-clicking. To adjust the knobs, click on the knob and move your mouse up or down. The numeric display of the current value below the knob you are adjusting will change to reflect your mouse position. If you press the SHIFT key before you click on a knob, the knob will be in Fine Movement Mode for as long as you hold the key down, which allows you to make more minute value adjustments.

The knob value display below each knob can be clicked to reveal a text box. You can directly type the desired numerical value into this box. All knobs with deactivated functions are grayed.

Click on the LED-like buttons to select the desired settings for the HPF and LPF filter order, Low Shelf and High Shelf modes, Flat Peak mode for peaking filters, Solos, Processing Channel, Phase, Routing or Output Limiter section and Make-Up.

Finally, you can click on the PSP MasterQ2 label to switch the graph display to an information panel with details about the plug-in authors, the version number, and to whom the plugin is licensed. Simply click on the information panel to return to the standard graph display.

Graph Area



EQ GRAPH

The EQ Graph shows the exact EQ curve of each activated filter in the same color as the line bordering that filter's Filter Activation button, as well as the thick black line indicating the combined EQ curve of all activated

filters. Note that the EQ graph is not only just a display—you can manipulate the EQ curves by clicking active points on the graph itself. When you click on a filter's circle and drag it around the corresponding filter's bottom line becomes bright for identification purposes. You can use the Alt/Option key to manipulate the Q or the Shift key for more precise operation. Double-click on a filter circle to activate or deactivate the filter section.

Scale dB Range

The graph allows you to switch between three dB scale ranges: ± 4 , ± 12 and ± 24 dB. You can switch between them by clicking above or below 0dB.

INPUT Slider

The INPUT slider sets the input signal level of the plug-in.

RANGE Slider

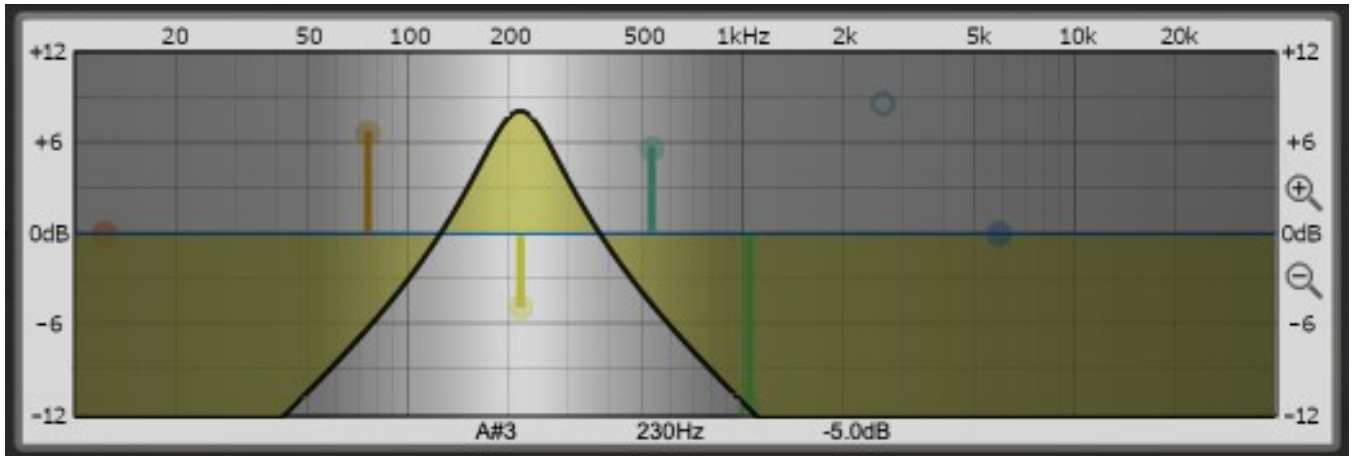
The RANGE slider acts as sort of a boost or cut for the entire EQ. It sets the multiplier assigned to all Gain controls of the individual filters. It allows you to increase, lessen or even invert the amplitude characteristics of all filters except low cut and high cut.

You can dial in settings from -150% to 150% adding up to an extra 50% boost or attenuation to the filters. Certain key numeric values are colored to make them stand out: 100% is indicated by green numbers, -100% is indicated by blue numbers, and 0% is indicated by red numbers. The default value is 100% , which leaves the gain settings of the filters unaltered.

LEVEL Slider

The LEVEL slider sets the output signal level of the EQ before it enters the limiting-saturation section. The setup of this slider affects the position of the black overall frequency curve on the graph. This slider refers only to a channel chosen for equalization.

Frequency Hunter Mode



Whenever you hold the CONTROL (PC) or CMD (Mac) key while hovering the mouse pointer on a filters' active point on the graph Frequency Hunter Mode is enabled. In this mode the graph view changes to show a single bell type filter curve and the graph view outside the filter's scope is dimmed. Now clicking and dragging the pointer while holding the CONTROL or CMD key will result in a retuning of the bell filter which allows you to easily search for resonances or any other critical frequencies.

Filters and Routing Area



PSP MasterQ2 provides seven filters. Two of them are permanently configured as low and high cut filters (2nd, 4th or 6th order), two are switchable between peak and shelf curves, while the remaining three filters are peak filters with an option to switch to a soft peak mode to slightly attenuate resonance. This optional soft peak mode prevents an unpleasant boost of distortion during deep equalization.

FILTER ACTIVATION BUTTONS

Each of PSP MasterQ2's filters can be turned on or off individually via its Filter Activation button. When you activate a filter, a line in the same color surrounding that filter's Filter Activation button appears in the EQ graph to represent that filter's EQ curve.

12/24/36 dB/oct Slope LED switch

Both the low cut and high cut filters have filter slope LED switches. Use this set of LEDs to select the order of the filter. The basic 12dB/oct (2nd order) slope provides a gentle curve. The 24dB/oct (4th order) reflects a slope with medium steepness while the 36dB/oct selection (6th order) offers extreme cutting possibilities at the expense of adding the phase distortion typical of heavy non-linear phase filtering.

Shelf LED Button

By clicking the LED under the Shelving curve you can switch between shelf filter and peak filter modes. Shelf mode is set by default and the LED is lit red.

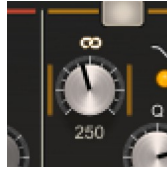
Soft Peak LED Button

Clicking and engaging the LED under the softened peak curves selects a special low resonance peaking filter mode. This mode eliminates harsh overprocessing when the sound is deeply equalized.

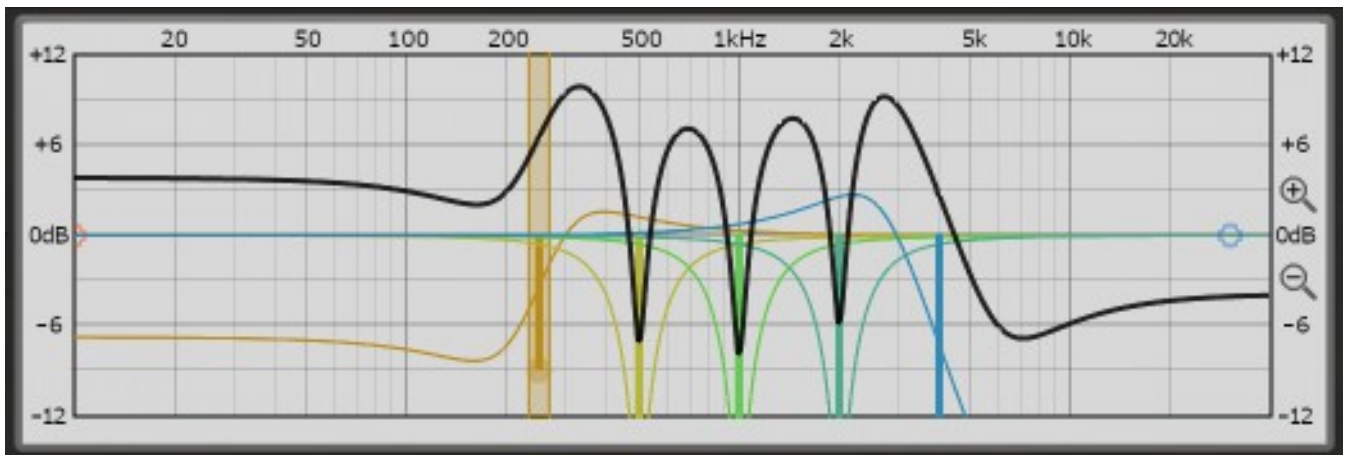
FREQUENCY Knob

This knob sets a cut-off or center frequency for an individual filter. The range of each filter's FREQ knob depends on the chosen filter. Each FREQ knob has a different default setting based on its range (see Operations for details).

Linked Filters Mode



The LowShelf/LowPeak frequency knob does not have the FREQ label, it contains a chain button instead. Click on the unlinked chain to engage the Linked Filters Mode.



When the Linked Filters Mode is set, all filters of the MasterQ can be adjusted together when the orange vertical bar associated with the Low Shelf/Peak filter is dragged or its frequency knob is adjusted. You can still adjust all other filters separately when their appropriate controls are used.

When in the Linked Filters Mode, small “+” and “-” rocker buttons appear to the left and to the right of FREQ labels above knobs. Those rocker buttons allow you to crawl between harmonics of the fundamental frequency associated with the Low Shelf/Peak filter. When going over a 15th harmonic the rocker changes its resolution to 1/12 of an octave.

Q Knob

The “Q factor” knob sets the width of a filter. The lower the Q, the wider the range of filter (which results in a softer slope to the EQ curve). Higher Q values result in the filter affecting a narrow frequency range (which looks like a notch or resonance peak on the EQ graph) The default value for each filter is set to match the purpose for that given filter (see Operations for details).

+ Button

All Qs of peak and shelf/peak filters have an alternative Q mode engaged with a “+” button next to the Q label. This Alternative Q provides a different shape and behavior of filters’ curves compared to a standard mode. This mode is the right choice to mimic behavior of various analog equalizers.

GAIN Knob

This knob sets the gain/attenuation for a filter. Its range is -24 dB to +24dB with increased resolution closer to 0dB. Note that the high and low cut filters do not have GAIN knobs.

Solo LED Buttons

Every filter has its own Solo LED button. You can select any number of filters to solo. The LED pulses red when the filter is soloed. When a non active filter is solo engaged the Solo LED is dimmed and the solo button of this filter doesn't influence the equalization curve.

Bottom Color Line

This line's color corresponds to the filter curve's color on the graph. When the filter is adjusted by the operation on the graph a corresponding Bottom Color Line glows for identification purposes.

Polarity LED Buttons

After the filters, the Level and the Stereo controls, there is a set of controls to set up the output channels' polarity. Click on either of the LEDs to set the corresponding channel into reversed polarity mode or use the darkened area in between those LEDs to switch both channels simultaneously.

ROUTE

Under the phase controls there is a vertical output Route LED switch. It allows you to select which channels get routed to the limiters/saturators section of the plug-in. You can choose between Middle, Left, Stereo, Right and Side settings. Please note that the signal level may rise up to 6dB when a Middle setting is engaged.

PROCeSS Setup Area



This area contains process/bypass, FAT, filter reset, and routing settings.

PROCeSS Button

The PROC button switches the plug-in into processing mode. The plug-in is bypassed when this button is disengaged. Please note that bypassing the PSP MasterQ2 using this button is not the same as turning the plug-in off (or bypassing the plug-in) using your audio application's plug-in window controls.

FAT

This button activates the Frequency Authentication Technique – a high quality double sampling technology which results in more accurate and transparent high frequency filtrations. This algorithm incurs a slight internal latency of 128 samples. In general, you will always want the FAT button engaged, however you might need to turn the FAT off if you need the lowest possible latency, or need to conserve CPU power.

RESET

Double-clicking this button will instantly reset the Range slider and all of the filters' settings to their default values. This is particularly useful if you find yourself wanting to zero out all your settings and start over.

Channel Routing LED Switch

Under the setup buttons is the Channel Routing LED Switch. It allows you to select a channel to be processed. By default the plug-in operates in stereo mode. In this mode both channels are processed equally by PSP MasterQ2. When another option is selected only a single channel (either Middle, Left, Right or Side) gets processed, yet both channels are still processed by STEREO knobs, the OUTPUT slider, and OUTPUT limiting and saturation section.

Please note that Middle and Side signals are not physical channels in the typical stereo track – they persist as components of a left/right stereo track and are extracted for processing losslessly. Middle and Side signals are feasible channels only in MS recordings before conversion to standard two channel stereophony. The PSP MasterQ2 is not intended to convert from MS to LR and back from the input to output. All channel routings and conversions in this plug-in are made internally for equalization purposes only.

Being able to select a single Middle or Side channel is a vital tool for mastering purposes or whenever a Mid or Side signal needs to be equalized separately from the stereo signal. Using separate plug-in instances for Left and Right channel processing will be necessary whenever problems with frequency response of those channels is not adequate.

ANALOG Setup Area



This section controls the Character and Amount of saturation inside all five of PSP MasterQ2's filters. Please note that the saturation doesn't occur in low cut and high cut filters.

CHARACTER Knob

This knob controls the ratio between even and odd harmonics which are added to the signal around the filter's resonant frequency. When set counter clockwise odd frequencies predominate. When set clockwise even harmonics become apparent.

AMOUNT Knob

This knob sets the amount of saturation of the filters. Please note that the level of harmonics is slight for most purposes, just like in high quality analog gear. Even when Amount is set to its maximum value the noticeable amount of harmonics stays pretty low.

ANALOG LED Button

Click on this LED Button to engage or disengage the analog saturation of filters. The saturation is engaged when the LED is lit.

STEREO Setup Area



This set of knobs allows you to adjust stereo width and balance.

WIDTH Knob

This knob controls the stereo width of the stereo audio track. The default position preserves the original input stereo width. Turning the knob counter clockwise reduces the stereophony, all the way down to mono. Turning the knob clockwise extends the stereo field. Please note that there is a balance between a middle and a side signal that should not be exceeded, otherwise the stereo image will collapse.

BALANCE Knob

This control manages the stereo balance between left and right channels.

OUTPUT Limiting and Saturation Area



This section provides a selection of controls for output limiting or saturation and a set of indicators for gain reduction and output level.

OUTPUT LEVEL LEDs and Numerical Indicators

Output level LEDs provide a visual indication of the output level presence (green), proximity (yellow) and exceeding the 0dBFS (red). Numerical values below the LEFT and RIGHT labels present digital peak held levels for the corresponding channel.

GAIN REDUCTION Meter

This bar meter indicates the amount of gain reduction which occurs due to the chosen limiting or saturation algorithm. The ideal amount of gain reduction differs depending on the particular track and the goal of your limiting – saturating. A good practice for mastering is not to exceed -1 or -2dB on a track with heavy transients; for limiting and saturation during mixing the reduction can go down to -3dB depending on the specific content. Obviously noticeable processing (if required) can be achieved with reduction below -6dB.

Lim-Sat LED Switch

Lim-Sat LED Switch lets you choose or disable an output limiting or saturation algorithm. Every provided algorithm limits and saturates the sound in a unique way. It is worth taking a while to get accustomed to each algorithm so you can choose wisely for a given track or a group. The Vint Lim algorithm is PSP VintageWarmer's limiting algorithm and usually provides a warm response.

Whenever more standard brick wall limiting is required please choose between Lim Soft and Lim Hard.

If the track requires a balance between saturation and limiting due to its transient content please consider Dyn (Saturation) Soft (Knee) or Dynamic (Saturation) Hard (Knee). You can also use more standard saturations: Sat Soft (Knee) or Sat Hard (Knee).

CEILING Knob

This control sets the ceiling level of all limiters and saturators. Although the default setting is 0dBFS you can adjust the ceiling level counter clockwise to lower the brick wall limiting/saturation level and increase a gain reduction or set the ceiling above 0dBFS to lower the impact of limiting / saturation algorithms. Please note that with settings above 0dBFS limiting / saturation algorithms will not prevent the signal from exceeding 0dBFS on output. Please also bear in mind that not all limiting and saturation algorithms are oversampled, hence, they won't prevent the signal from exceeding a set up ceiling level when measured with an oversampling meter.

MAKEUP LED Button

This control puts the limiter/saturator into makeup mode. When engaged, the output signal (post limiter/saturator) will be boosted or attenuated so the signal will not exceed -0.1dBFS

Operations

Filter Frequencies

The frequency range of each filter has been specifically tailored to be as useful and dynamic as possible for its given range. You'll notice that most of the filters in the PSP MasterQ2 actually have a far wider range than most other EQ plug-ins. This is to give you maximum flexibility in adjusting each filter. The frequency ranges and default values for each filter, from left to right across the PSP MasterQ2 interface, is as follows:

Low Cut Filter: 10 Hz – 20 kHz; 60Hz default

Low Shelving / Peaking Filter: 12 Hz – 12 kHz; 100Hz default

Low Mid Peaking Filter: 16 Hz – 16 kHz; 250Hz default

Mid Peaking Filter: 20 Hz – 20 kHz; 650Hz default

High Mid Peaking Filter: 24 Hz – 24 kHz; 4.5kHz default

High Shelving / Peaking Filter: 30 Hz – 30 kHz; 8kHz default

High Cut Filter: 20 Hz – 40 kHz; 40kHz default

Adjusting the Q

While there are no hard and fast rules as to how to set the Q factor for each filter, here are a few general rules that can help you get the most out of PSP MasterQ2.

Low shelf and high shelf Q factors within the range of 0.35 to 0.71 result in a smooth shelf curve. Values as low as 0.35 give the smoothest possible curve while values greater than 0.71 add a peak above and below the set frequency.

Low cut and high cut Q factors in the range of 0.35 to 0.71 result in a typical low cut and high cut curve without any resonance gain. A value of 0.71 is the maximum Q without an apparent peak in the characteristic. Lower values decrease the steepness of the curve, 0.5 provides a non-oscillating response. Values greater than 0.71 raise the resonance.

The Q ranges and default values for each filter, from left to right across the PSP MasterQ2 interface, are as follows:

Low Cut Filter: 0.35 – 2.83; 0.71 default

Low Shelving / Peaking Filter: 0.35 – 2.83; 0.71 default in Shelving mode

Low Shelving / Peaking Filter: 0.05 – 20; 1 default in Peaking mode

Low Mid Filter: 0.05 – 20; 0.41 default

Mid Filter: 0.05 – 20; 1 default

High Mid Filter: 0.05 – 20; 1 default

High Shelving / Peaking Filter: 0.35 – 2.83; 0.71 default in Shelving mode

High Shelving / Peaking Filter: 0.05 – 20; 1 default in Peaking mode

High Cut Filter: 0.35 – 2.83; 0.71 default

Preset Handling & View Options

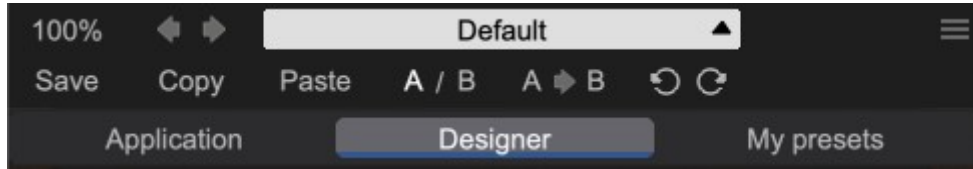
We provide all PSP plugins with a collection of factory presets. These presets can be used as a starting point for further adjustments, or for quick fixes.

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



Preset Browser

The PSP MasterQ2 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 management bar 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 grouped 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 MasterQ2.

To add categories to the preset list, you can create new subfolders in the preset directory.

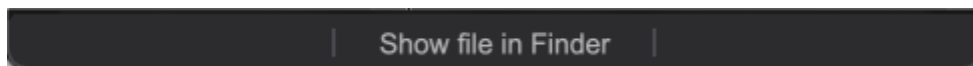
For Windows users, this is located at:

C:\Users\Username\Documents\PSPaudioware.com\User Presets\PSP MasterQ2

For Mac users, this is located at:

~/Documents/PSPaudioware.com/User Presets/PSP MasterQ2

< **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, 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 to open the designer's website.

Copy / Paste

A dark rectangular button with the text 'Copy' on the left and 'Paste' on the right, both in a light gray font.

The **Copy/Paste** feature allows you 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

A dark rectangular button with the text 'A / B' on the left and 'A → B' on the right, both in a light gray font.

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

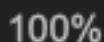


The **Undo/Redo** feature of the plugin lets you quickly navigate between setting changes.

To use this feature, 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

A dark rectangular button with the text '100%' in a light gray font.

Scroll (by the mouse wheel or use 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 just dragging the left bottom corner of the plug-in to any size you like.

CONFIG section



When clicking the three parallel lines glyph, 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 10.10 – macOS 12.01 Monterey
- 64-bit AudioUnit compatible host application

VST

- macOS 10.10 – macOS 12.01 Monterey
- 64-bit VST compatible host application

VST3

- macOS 10.10 – macOS 12.01 Monterey
- 64-bit VST3 compatible host application

AAX

- macOS 10.10 – 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)



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Processing

- All internal processing done with 64 bit double precision floats.
- 32 and 64 bit floating point audio streams supported
- Sample rates up to 768 kHz supported.

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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