

Portico II

Master Buss Processor

By:

RUPERT NEVE DESIGNS

Serial #:



Operations Manual

Important Safety Instructions

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Clean only with a dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
10. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11. Only use attachments/accessories specified by the manufacturer.
12. Use only with a cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
13. Unplug this apparatus during lightning storms or when unused for long periods of time.
14. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.
15. This apparatus shall not be exposed to dripping or splashing, and no object filled with liquids, such as vases or beer glasses, shall be placed on the apparatus.
16. Do not overload wall outlets and extension cords as this can result in a risk of fire or electric shock.
17. This apparatus has been designed with Class-I construction and must be connected to a mains socket outlet with a protective earthing connection (the third grounding prong).
18. This apparatus has been equipped with a rocker-style AC mains power switch. This switch is located on the rear panel and should remain readily accessible to the user.
19. The MAINS plug or an appliance coupler is used as the disconnect device, so the disconnect device shall remain readily operable.



20. NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and the receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

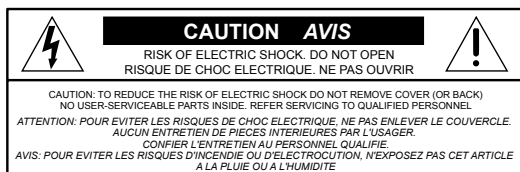
CAUTION: Changes or modifications to this device not expressly approved by Rupert Neve Designs LLC, could void the user's authority to operate the equipment under FCC rules.

21. This apparatus does not exceed the Class A/Class B (whichever is applicable) limits for radio noise emissions from digital apparatus as set out in the radio interference regulations of the Canadian Department of Communications.

ATTENTION — *Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques de class A/de class B (selon le cas) prescrites dans le règlement sur le brouillage radioélectrique édicté par les ministères des communications du Canada.*

22. Exposure to extremely high noise levels may cause permanent hearing loss. Individuals vary considerably in susceptibility to noise-induced hearing loss, but nearly everyone will lose some hearing if exposed to sufficiently intense noise for a period of time. The U.S. Government's Occupational Safety and Health Administration (OSHA) has specified the permissible noise level exposures shown in the following chart. According to OSHA, any exposure in excess of these permissible limits could result in some hearing loss. To ensure against potentially dangerous exposure to high sound pressure levels, it is recommended that all persons exposed to equipment capable of producing high sound pressure levels use hearing protectors while the equipment is in operation. Ear plugs or protectors in the ear canals or over the ears must be worn when operating the equipment in order to prevent permanent hearing loss if exposure is in excess of the limits set forth here:

Duration, per day in hours	Sound Level dBA, Slow Response	Typical Example
8	90	Duo in small club
6	92	
4	95	Subway Train
3	97	
2	100	Typical music via head phones
1.5	102	
1	105	Siren at 10 m distance
0.5	110	
0.25 or less	115	Loudest parts at a rock concert



The lightning flash with arrowhead symbol within an equilateral triangle is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure, that may be of sufficient magnitude to constitute a risk of electric shock to persons.

Le symbole éclair avec point de flèche à l'intérieur d'un triangle équilatéral est utilisé pour alerter l'utilisateur de la présence à l'intérieur du coffret de "voltage dangereux" non isolé d'ampleur suffisante pour constituer un risque d'électrocution.



The exclamation point within an equilateral triangle is intended to alert the user of the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

Le point d'exclamation à l'intérieur d'un triangle équilatéral est employé pour alerter les utilisateurs de la présence d'instructions importantes pour le fonctionnement et l'entretien (service) dans le livret d'instruction accompagnant l'appareil.

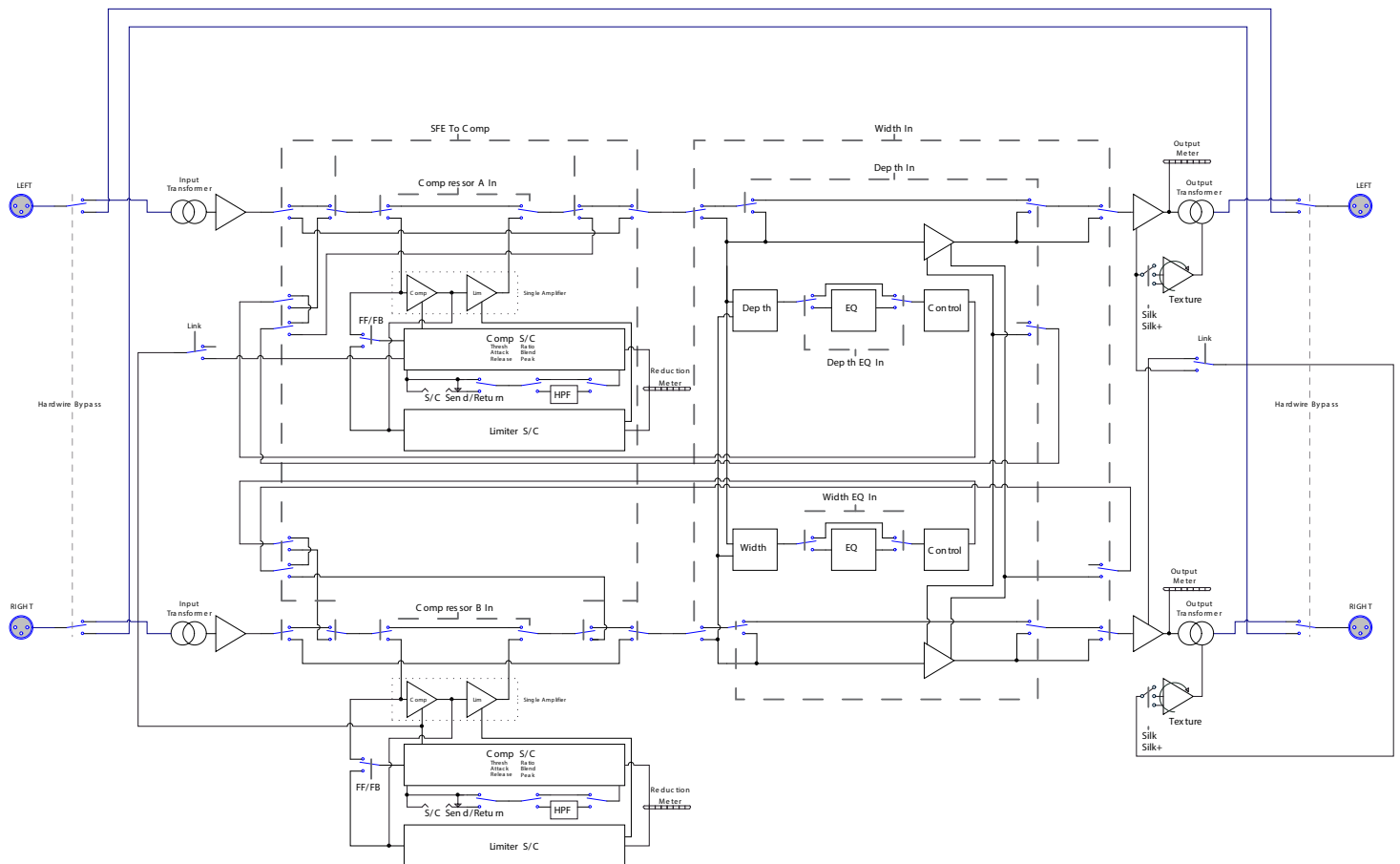
WARNING — To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture.

Portico II Master Buss Processor User Guide

Thank you for your purchase of a Portico II Master Buss Processor.

Everyone at Rupert Neve Designs hope you enjoy using this tool as much as we have enjoyed designing and building it. The name “Portico II” is being used for a series of new ultra-high end professional products that will share some traits. Please take note of the list of safety concerns and power requirements before the use of this product.

Portico II Master Buss Processor: Block Diagram



Power Requirements

Each Portico II unit has a high quality, low noise switching power supply that is further filtered and regulated for an exceptionally quiet and reliable power source for the audio circuits. The power supply is considered “universal” in the sense that it will accept 100V through 240V AC and with 50 or 60Hz. Be absolutely sure to disconnect mains power (remove the power cable from the IEC power connector at the back panel) before checking the fuse. The fuse is located in the IEC power input connector and is accessed by opening the small panel labeled “FUSE”. The fuse should always be replaced with the correct value and type. The Portico II power supply requires a 5x20 mm 2.5 amp fast acting ceramic body fuse Bussman type GDA 2.5A or equivalent.

The fuse is a protection device intended to prevent additional damage or hazard if the Portico II unit develops a problem. However sometimes a fuse may “pop” due to a mains surge and need to be replaced (it probably did protect the Portico II). The symptom of a blown fuse is simply that the unit does not power up. Disconnect the power cable, try replacing the fuse, re-insert the power cable, push the “Power” button. If this does not solve the problem or the fuse “pops” again there may be a problem with the Portico II you should contact your dealer or email <support@rupertneve.com> or **phone 512-847-3013**.

The Rupert Neve Designs Portico II™ MBP Overview

The Rupert Neve Designs Portico II Master Buss Processor (MBP) is a full 19” rack width, 3.5” (2U) with standard rack mounting “ears”. As with the entire Portico II range, the construction incorporates a heavy and robust steel shell that provides total magnetic screening and exceptional mechanical stability. The front panel is machined from a solid .20 inch aluminum plate with a steel sub panel behind it.

The name says a lot about the device but certainly not everything. The audio electronics use higher power supply voltage compared to most audio equipment and in this case use +36 and -36 volt power supply rails. The signal path has many similar circuits as the Rupert Neve Designs flagship console, the 5088. Amplification is handled with discrete (individual transistors) operational amplifiers (gain blocks). These discrete circuits are in the tradition of Rupert’s original circuits used in recording consoles from the 70’s and in many ways sound similar, however there are refinements in noise, slew rate, dynamic range and particularly avoidance of unpleasant high frequency distortion artifacts. Also following Rupert’s traditional approaches, these gain stages are married to custom designed input and output transformers. Not only is the output transformer custom designed for the Portico II, it is the largest toroid output transformer that he has used for a line level device.

The Master Buss Processor is a unique mixing and mastering device in that it brings together three distinct elements; the compressor, limiter and stereo field editor. While these three separate components are used in their own individual products, the MBP gives the ability to combine the three, enhancing the creative potential of each feature. Although the feature set is quite expansive, signal paths are minimal, and only incorporate the utilized components. All in all, the Portico II MBP is an outstanding tool for mixing and mastering.

THE LINE INPUTS & OUTPUTS

The input and output stages are similar to that of the 5088 console, using high voltage class-A discrete circuitry, driving a carefully configured output transformer that can deliver a full +25dBu from the balanced and ground-free secondary winding.

This maximum level provides a large margin over and above the likely maximum requirement of any destination equipment to which the module may be connected. This is especially true when feeding digital equipment!

Freedom from the interference fields that are inevitably present in any control room is virtually guaranteed by the balanced, ground-free design used in the MBP. The classic Rupert Neve designed modules always used transformers, as do a number of other high quality vintage modules still in current use.

High quality transformer connectivity has been used for many years, enabling modular amplifier units to deliver the sonic performance for which they are famous. The outputs are very appropriate for driving unusually long lines that may be needed when used remotely.

Bear in mind that human ears are very sensitive and can perceive incredibly minute interference signals that are not part of the “desired” signal. If unbalanced connections are used, great care must be exercised to avoid ground loops and common signal paths. Reduced immunity from various forms of interference can be tolerated (sometimes) but usually results in a loss of that transparent musical resolution that we all love.

However, the output of any Portico II transformer-coupled XLR may be used with one side grounded if necessary, for example to use with “Hi-Fi”, “consumer” or other unbalanced audio gear, without degrading the performance of such devices. Care must be exercised when using ancillary equipment to avoid overloading it.

THE SIDE CHAIN INSERT JACKS

These pairs of jacks are only used to perform some fine tuning of the compressor operation. The audio that normally controls the compressor is available on the “SIDECHAIN OUTPUT” jack. One can take this audio and pass it through an external equalizer then return it back to the “SIDECHAIN INPUT” jack. Now changes on the external EQ affect the sensitivity of the compressor. For example, if you cut some low frequencies on the external EQ, then the compressor will tend to not want to reduce the gain as much on bass notes. This is not the same thing as simply boosting lows on an equalizer, because the latter changes the frequency response while the former changes levels. Similarly, one could boost highs (6kHz for example) on the external EQ and cause the compressor to be extra-sensitive to that part of the spectrum which happens to correspond with sibilance (esses). This can act as one type of a de-esser that pulls down the volume of the whole signal if it senses an “ess”.

A few notes about these jacks. They are unbalanced and ideally best suited to be used with unbalanced equipment. However most balanced EQ's will work fine interfaced to these jacks. The simple symptom of a balanced / unbalanced mismatch is that the “compression meter” will indicate a significant change of compression depth (like zero) when the EQ is set flat. Keep in mind that nobody hears this signal path so that any old nasty sounding EQ you tossed in a closet years ago might be perfectly suited for a side chain insert task.

A NOTE ON DISTORTION

The human hearing system is a remarkably complex mechanism and we seem to be learning more details about its workings all the time. For example, Oohashi demonstrated that arbitrarily filtering out ultrasonic information that is generally considered above our hearing range had a measurable effect on listener's electroencephalo-grams. Kunchur describes several demonstrations that have shown that our

hearing is capable of approximately twice the timing resolution than a limit of 20 kHz might imply ($F=1/T$ or $T=1/F$). His peer reviewed papers demonstrated that we can hear timing resolution at approximately with 5 microsecond resolution (20 kHz implies a 9 microsecond temporal resolution, while a CD at 44.1k sample rate has a best-case temporal resolution of 23 microseconds).

It is also well understood that we can perceive steady tones even when buried under 20 to 30 dB of noise. And we know that most gain stages exhibit rising distortion at higher frequencies, including more IM distortion. One common IM test is to mix 19 kHz and 20 kHz sine waves, send them through a device and then measure how much 1 kHz is generated ($20-19=1$). All this hints at the importance of maintaining a sufficient bandwidth with minimal phase shift, while at the same time minimizing high frequency artifacts and distortions. All of the above and our experience listening and designing suggest that there are many subtle aspects to hearing that are beyond the realm of simple traditional measurement characterizations.

The way in which an analog amplifier handles very small signals is as important as the way it behaves at high levels. For low distortion, an analog amplifier must have a linear transfer characteristic, in other words, the output signal must be an exact replica of the input signal, differing only in magnitude. The magnitude can be controlled by a gain control or fader (consisting of a high quality variable resistor that, by definition, has a linear transfer characteristic.) A dynamics controller - i.e. a compressor, limiter or expander - is a gain control that can adjust gain of the amplifier very rapidly in response to the fluctuating audio signal, ideally without introducing significant distortion, i.e. it must have a linear transfer characteristic. But, by definition, rapidly changing gain means that a signal "starting out" to be linear and, therefore without distortion, gets changed on the way to produce a different amplitude.

Inevitably our data bank of "natural" sound is built up on the basis of our personal experience and this must surely emphasize the importance of listening to "natural" sound, and high quality musical instruments within acoustic environments that is subjectively pleasing so as to develop keen awareness that will contribute to a reliable data bank. Humans who have not experienced enough "natural" sound may well have a flawed data bank! Quality recording equipment should be capable of retaining "natural" sound and this is indeed the traditional measuring stick. And "creative" musical equipment should provide the tools to manipulate the sound to enhance the emotional appeal of the music without destroying it. Memory and knowledge of real acoustic and musical events may be the biggest tool and advantage any recording engineer may possess.

One needs to be very careful when one hears traces of distortion prior to recording because some flavors of distortion that might seem acceptable (or even stylish) initially, may later prove to cause irreparable damage to parts of the sound (for example, "warm lows" but "harsh sibilance") or in louder or quieter sections of the recording. Experience shows that mic preamps and basic console routing paths should offer supreme fidelity otherwise the engineer has little control or choice of recorded "color" and little recourse to undo after the fact. Devices or circuits that can easily be bypassed are usually better choices when "color" is a consideration and this particularly is an area where one might consider comparing several such devices. Beware that usually deviations from linearity carry at least as much long-term penalty as initial appeal, and that one should always be listening critically when recording and generally "playing it safe" when introducing effects that cannot be removed.

1. Tsutomu Oohashi, Emi Nishina, Norie Kawai, Yoshitaka Fuwamoto, and Hishi Imai. National Institute of Multimedia Education, Tokyo. "High Frequency Sound Above the Audible Range, Affects Brain Electric Activity and Sound Perception" Paper read at 91st. Convention of the A.E.S. October 1991. Section 7. (1), Conclusion.

2. Miland Kunchur, Department of Physics and Astronomy, University of South Carolina. "Temporal resolution of hearing probed by bandwidth restriction", M. N. Kunchur, *Acta Acustica united with Acustica* 94, 594–603 (2008) (<http://www.physics.sc.edu/kunchur/Acoustics-papers.htm>)
3. Miland Kunchur, Department of Physics and Astronomy, University of South Carolina. Probing the temporal resolution and bandwidth of human hearing, M. N. Kunchur, *Proc. of Meetings on Acoustics (POMA)* 2, 050006 (2008)

COMPRESSOR-LIMITER CONTROLS

THRESHOLD

Sets level where the compressor may begin to react from -30 dB to +20 dB. Minimal or no compression is with this control fully clockwise and it gets more sensitive and tends to cause more gain reduction as the knob is rotated counter-clockwise (which may be counter-intuitive to some).

RATIO

Sets the "slope" of the compression from 1.1:1 (minimal) to LIMIT (approximately 20:1) (or drastic). For example if this knob is set for 3:1 then if the signal goes 15 dB over the threshold then it attenuates 10 dB and allows the output to rise 5 dB. In general, low ratios can not damage the music as much as high ratios but high ratios may be more useful to minimize clipping and OL lights in the recorder.

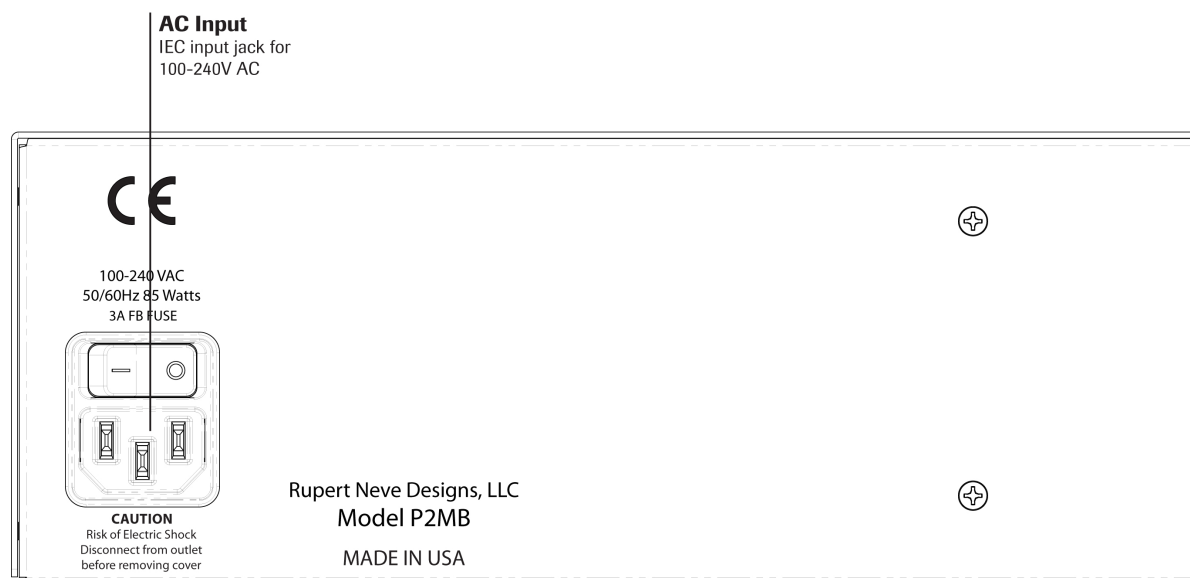
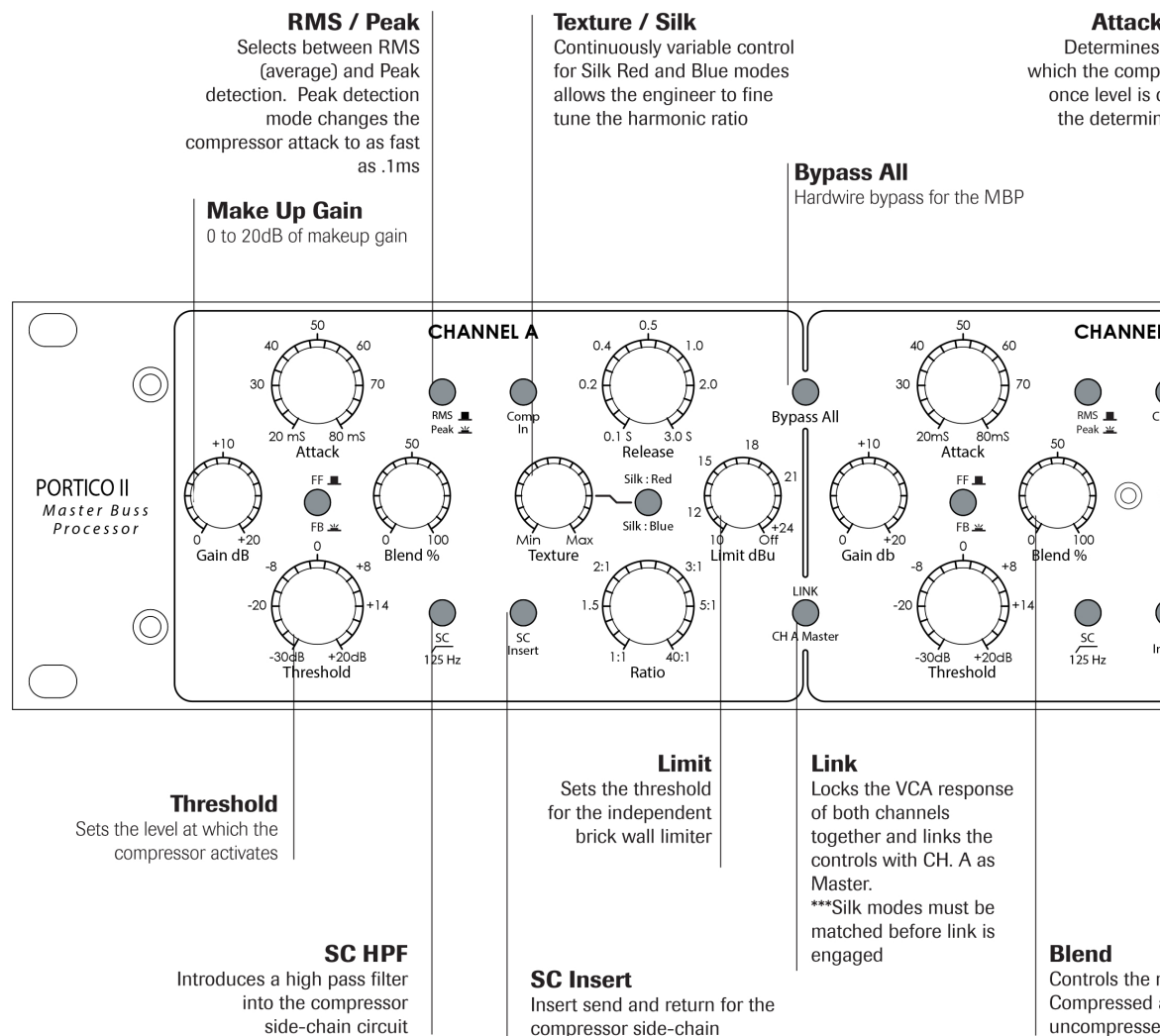
Excuse the front panel jump and digression, but it is appropriate to mention the GAIN REDUCTION and OUTPUT meters now because so many of us learned to set up compressors by watching the all-important GR meter. When setting the Threshold and Ratio it does help to use the meters however, as always, we should rely on our ears and ye olde "COMP IN" (Bypass Button) most of all. One might try to interpret the OUTPUT meter as one exercises ye olde "COMP IN" button too, as this might be more appropriate to your goals than a target of X dB or number of LEDs of compression. Is the goal to make great sounds or to blink LEDs half way down the meter because you read it somewhere?

BLEND

This just mixes the dry or raw uncompressed signal and the compressed signal. Fully counter-clockwise is uncompressed and is very similar to not having the "COMP IN" button pushed in. Fully clockwise is 100% compressor-limiter path, and one can "blend" or mix how much compressed-limited signal vs. dry signal. This control is not found on vintage compressors and is a recent invention that has gained popularity. Why? Compressors generally attenuate the louder signals and leave the quiet bits untouched, which might be re-worded as compressors tend to reduce transient peaks. If one then mixes in some dry signal, peaks and all, with the compressor (usually with the Threshold lowish, Ratio steepish and "GAIN" turned up to compensate) then the "blend" will be mostly dry for the loud bits and compressor for the quiet bits, which can likewise be re-worded to the quiet stuff became louder or the room reverb increased. Obviously one has a fair amount more control over the entire dynamic range given this control and it should be mentioned less likelihood of damage to parts of the music, because "blending" implies not 100% compression electronics and there is important musical information in those transient peaks (like the hit of drums and percussion where a lot of the groove lives).

COMP IN

The compressor-limiter section is engaged with this button in. The Comp In button does not effect the Silk or SFE controls.



Portico II Master Buss Processor: Front / Back Panel

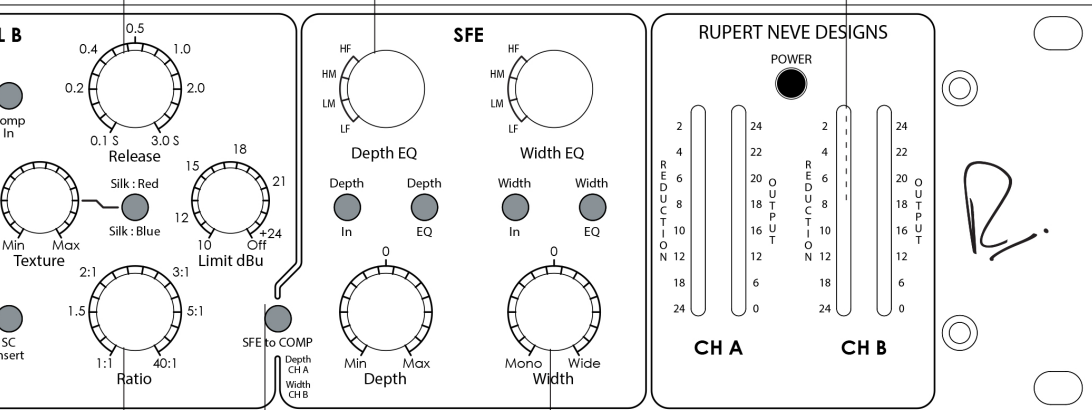
/ Release
the speed at
the compressor reacts
over or under
the set threshold

Depth & Depth EQ

Increases or decreases the
amount of mid information in
the stereo signal. Engaging
the EQ tunes the depth effect
to the selected frequency
band

LED Metering

Dual 16 point peak meters
display compressor gain
reduction and output levels



SFE to Comp

Routes the SFE M=S Signals
into the two compressors,
with mids in CH. A. and sides
in CH. B.

Width & Width EQ

Increases or decreases the
amount of side information in
the stereo signal. Engaging
the EQ tunes the width effect
to the selected frequency band

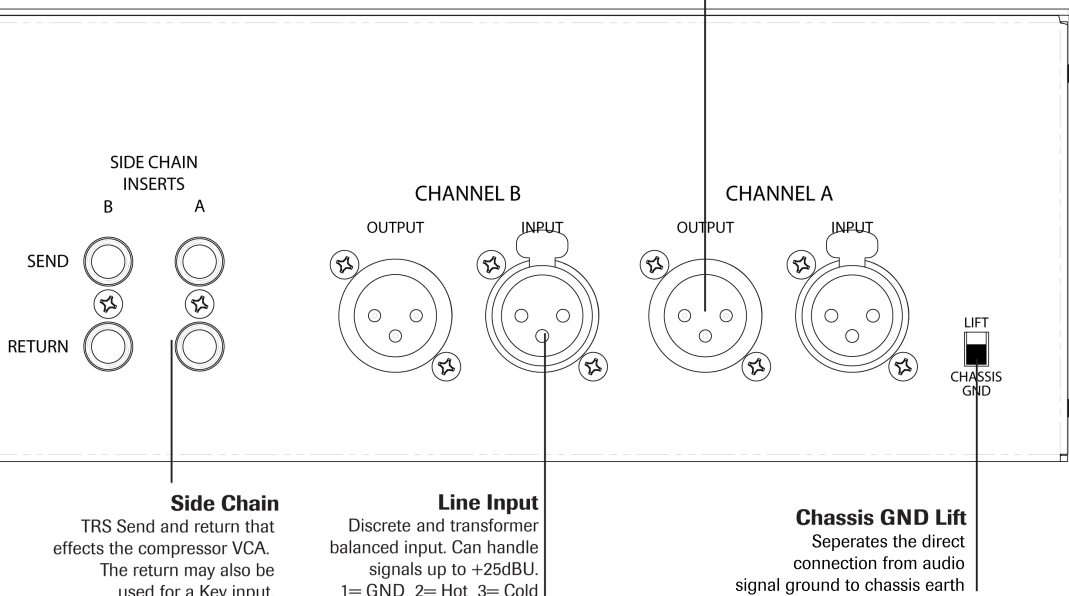
Ratio

Sets the slope of
compression about the set
threshold

mixture of
and
d signals

Line Output

Transformer coupled fully
balanced and floating. Can
Output signals up to +25dBu
1= GND 2= Hot 3= Cold



BYPASS ALL

This may be the most useful control on the MBP, because it is there for “confidence checks”. In particular, exercise it in both the quietest and the loudest sections of the song. Watch out for the compressor and limiter exaggerating noise and room sound in the quiet sections and chopping off transients and consonants in the loud sections, either of which suggest maybe too much compressor “action”. Conversely, losing quiet phrases or inadvertent clipping might suggest that a little more compression is warranted, keeping in mind that you can always do a bit more in the mix but undoing over-compression is not fun and often not even possible.

FF/FB

These are two very different compressor modes, FEED BACK and FEED FORWARD. Vintage compressors almost always were FEED BACK designs which means they looked at the output after the VCA (Voltage Controlled Amplifier) or gain changing element and used that signal to control the attenuation. FEED FORWARD compressors became popular in the 80’s and read the signal before the VCA and through more elaborate electronics to control the attenuation because this has some advantages in regards to the ratio and control timing. Perhaps a more useful generalization is that the FB mode (button in) tends to sound smoother and often more natural and tends to be quicker to set up. The FF mode can be more useful for shaping the envelope of the sound and introducing more bounce and pumping in time with the song, when that is the goal. FF compressors were often used on 80’s dance tracks. Some engineers prefer FB with lower ratios and FF with higher ratios.

GAIN

Often referred to as Make Up Gain. Considering the compressor’s VCA is generally being forced to attenuate louder signals some method of returning the average level to a volume comparable with the compressor bypassed is desirable. The GAIN control is mostly used for this purpose especially for those of us that depend on comparing compression to bypass. GAIN is often pushed for even more level than “bypass” because it is understood that the compressor should be providing some effective headroom (besides “louder is better” being the oldest trick in the book).

ATTACK

This sets how quickly the compressor reacts and starts attenuating. If set fast (20mS) the compressor should react to very quick transients like the initial stick hit of a snare drum and can attenuate the “hit” so the “note” of the drum seems relatively emphasized. If set slower, the compressor will tend to ignore the fastest transients and react more to the drum resonance and attenuate the “note” of the drum so that the hit seems more emphasized. Similarly, on a mixed track, if a compressor is set too fast, it will tend to remove drums, which may help to explain why mastering engineers tend to use medium to slow attack times and lower ratios.

RELEASE

This control sets how fast the compressor will return to unity after attenuation. Typically, engineers use relatively slow releases when trying to minimize obvious compressor action or changes in gain. In modern pop music, the usual goal is to time the compressor gain changes with the rhythm of the music, making the RELEASE control setting vital to proper compression.

Another common use of compression is to maximize apparent volume, accomplished by fast release settings that allow the compressor to return to unity as quickly as possible. Compressor-limiters have a tendency to introduce “modulation distortion” when they are set to a combination of fast attacks, fast releases, and high ratios. This modulation distortion never sounds good, and should be avoided.

S/C 150 Hz

This routes the 150Hz High Pass Filter into the circuit that the compressor uses to determine level, commonly referred to as “the side-chain”. Note that the rest of the circuit and output will not have those lows filtered out. This function tends to be very useful because typical sounds often have more energy in the low octaves and can cause excessive compression. Our ears may tend to associate loudness with mids or high mids for some sounds and one may be wanting the compressor to regulate and smooth perceived loudness. Removing some amount of low frequencies that the compressor “sees” can help especially if one is compressing deeply (-8 dB or more).

RMS / PEAK

This changes the compressor from essentially responding to the RMS level of the audio to also responding to the PEAK level. RMS (root mean squared) circuits are considered to better mimic the way the ears perceive apparent loudness, while Peak circuits tend to directly respond to the waveform voltage which may be more of a concern for prevention of clipping and maximizing levels. In this case, pushing RMS / PEAK uses a combination of both methods to get the best of both worlds and avoidance of the drawbacks of each method on its own.

Do we recommend any particular compressor settings for particular instruments? No, but we will recommend that you not depend on hear-say settings and that you always listen carefully to the levels and mix values as you tweak. You can generally regard a compressor as a semi-automatic volume control and be aware of the kinds of artifacts that you would get moving a fader quickly. You can also listen for changes in tone that are probably due to altering the relative strength of transient hits and plosives that often contain more high mid and highs.

LIMIT dBU

At first glance, one might question the limiter’s single knob operation; however, this independent limiter is extremely intelligent, and appropriately responsive to a wide range of dynamic signal. Our new Serial Adaptive Release Technology is behind this revolutionary performance.

Using a blend of release time constants, this limiter can simultaneously respond to transient material (such as the “snap” of a snare drum) as well as slower signals (such as a bass guitar). This configuration allows the limiter to grab a transient and let go in an instant, while dealing with the more constant signals in a slower, more musical way. In this manner, the MBP can provide a more aggressive, distortion free limiter, while maintaining the essential musical character of the music.

Typically, there is a trade-off between how fast a limiter can react compared to the amount of modulation distortion in the lower frequency range. This is due to the lower frequencies rapidly triggering the compressor on and off through the side-chain’s influence, modulating the overall signal. The MBP does not fall victim to this compromise, allowing for a quick and snappy response while maintaining integrity and smoothness in the low end. In addition to the adaptive time constant circuitry, the release time is also varied with the limiter control knob. As the limiter knob is turned counter-clockwise, the release time is increased proportionally, achieving a longer release time as gain reduction increases.

The MBP’s limiter is designed to respond as fast as 0.03 ms in order to reduce the first half of a 20 kHz waveform over the limiter’s threshold. It has a “medium knee” initial ratio, increasing to more than 40:1 within 3 dB of the threshold. A soft clip circuit catches any transients that may be in the limiter’s “knee” when its threshold knob is set higher. With the limiter knob fully clockwise, both the limiter

and soft clipper are switched out of circuit. The limiter's release times are fully automatic and adjust themselves to the average depth of limiting, responding to the duration of transient material above the threshold. Since the limiters share the same discrete Class A gain module and VCA as the rest of the MBP compressor, there are no additional stages introduced into the signal path when the limiter is used. This combination of features provides exceptionally transparent limiting, often allowing for twice as much gain reduction before any noticeable artifacts appear, as compared to other limiters.

SILK / TEXTURE

Pushing the Silk button engages the "Silk-Red" circuit, and pushing it a second time introduces "Silk-Blue" circuitry. "Silk" reduces the negative feedback on the output transformer, adding harmonic content as the texture is increased. "Silk Blue" mode features more saturation in the lows and low mids, where as "Silk Red" accentuates the saturation in the high-mids and highs.

In the Portico II series, Both "Silk" modes are modified and fine tuned by the "Texture" control. By manipulating the "Texture" control, the amount of "Silk" can be changed from essentially absent, to roughly twice the amount of coloration found in "Silk" from the original Portico Series.

With silk/texture engaged, the distortion characteristic and harmonic content of the unit are very reminiscent of many of Rupert's class-A vintage designs. These controls add an unparalleled range of tonal options to the Master Buss Processor and should be explored creatively with a variety of different sources for best effect.

LINK

The link control connects the two compressor channel's VCA signals and settings, with ch. A as the master. With the inclusion of linked and detented controls, calibrated stereo setup for mastering is made incredibly easy in the MBP. When using the silk circuitry in Link mode, the Silk settings must be matched before Link is engaged. The Silk settings will cycle together when the Ch. A Silk button is pressed in link mode, but if they are not matched before linking, they will always be mismatched.

Stereo Field Editor Controls

The stereo field editor on the MBP takes traditional M-S techniques to new heights using width, depth and corresponding band-pass filters in concert to manipulate spacial elements.

WIDTH

The width control enables the user to increase or decrease the perceived width of the stereo image (wide/mono), while adjusting the amount of ambience heard in the recording. As the width control is rotated toward the "wide" setting, the amount of difference material is boosted, often resulting in more ambient material with accentuated stereo reverberation. Conversely, the perceived stereo field will contract when the width control is rotated towards the "mono" setting. Listening with experimentation is essential for best results, since the effective amount of perceived width is entirely dependent on the inherent amount of stereo field information and the interplay between the stereo field editor's additional controls.

DEPTH

The depth control of the MBP adjusts the spatial positioning of elements in the sound stage. Center-panned elements like solo instruments or vocals can be brought forward in a mix in relation to the supporting instruments. In many cases, these same elements may be virtually eliminated without adversely affecting any sonic information. If the left and right channels are highly phase coherent, their

mono content will be enhanced. If the phase of one of the input channels is then reversed 180 degrees, the mono content may be virtually eliminated. The depth and width controls can effectively alter the perceived room ambience and dimension.

WIDTH & DEPTH EQ

In order to fine tune the SFE, there are individual filters that allow the user to choose what information is reintroduced from the width and depth circuits; tailoring each effect to a specific bandwidth. For example, if the user wanted to augment the amount of low frequencies in the center image, one could engage the SFE Depth and LF Depth EQ to filter out everything in the Mid signal except for those frequencies below the LF filter point (in this case 250 Hz). Once reintroduced to the original signal, the result would be a perceived increase in the center image low frequency content. It is also possible to accomplish the same effect using the Width EQ, removing low frequencies from the Side information, thereby tightening up low frequency perception in the center field. To achieve spatial widening of instrument reverberation and background vocals, set the Width EQ to HM (or LM as the case may be). This will allow the user to increase the amount of band-passed Side information, creating the illusion of spatial widening that can envelope the listener.

SFE to COMP

The SFE to Comp button routes the Mid and Side signals to the Channel A and Channel B compressors, respectively. The Depth and Width changes are routed through the compressors, allowing the user to utilize the compressor features while manipulating the Mid and Side information. With these controls, it is possible to not only increase the side information, but to utilize the compressor to bring up some of the low level side information, or allow the user to tame an overly expressive lead singer in the mids. With the addition of using the EQ section on the Depth and Width, a wide range of tools are available to the engineer.

GAIN REDUCTION METER

A 16 segment LED bar-graph meter is fitted for the compressor, calibrated in dB to show how much gain reduction is taking place. After 4 dB of dynamic attenuation the LEDs become amber and after 8 the LEDs become red. Is that a hint? Not really, because sometimes we need significant amounts of compression, particularly with some powerful singers. However, the LED colors typically can serve as a general guide.

OUTPUT METER

The factory calibration sets the top RED LED at the specified output clip point of the Portico II which is +25 dBu, which is +21 dB over 0 VU. The LEDs are set in one dB increments. That means the bottom red LED indicates +18 over 0 VU which is the most typical digital full scale level for pro A to D converters and the top amber is one dB below clipping for most pro converters. An alternate typical DFS calibration for semi-pro converters is +14 VU which corresponds to the bottom amber LED (18 dBu). There happens to be a very good and new trend where we avoid trying to get the signal as close as possible to digital full scale because it can help our digital tracks sound more analog. In other words, feel free to not light up any of those amber and red LEDs and don't feel compelled to make every green one blink.

POWER

We obviously saved the best for last. If this button is not pressed then the Portico II is maximally "green" and exhibits its absolute lowest noise floor. However, for any of the previously described features and fun controls to have any significance, the POWER button should be pressed. If nothing happens when the button is pressed and not one LED even winks at you, then you may also want to plug in the Power

cord too.

Specifications

Frequency Response:

Main Output, no load,
-3 dB @ 4 Hz
-3 dB @ 120 kHz

Maximum Output Level (Compressor and SFE Bypassed):

Maximum output from 20 Hz to 40 kHz is +24.5 dBu.

Total Harmonic Distortion and Noise:

@ 1kHz, +20 dBu output: Better than 0.006%
@ 20Hz, +20 dBu output: Better than 0.20%
- As above, Silk Blue Engaged, Texture fully CW: Better than 2.5%
- As above, Silk Red Engaged, Texture fully CW: Better than 3.5%
@ 20kHz, +20 dBu output: Better than 0.020%

Slew rate Better than 4 V/uS

Compressor

Threshold: Variable from -30dBu to +20dBu
Ratio: Variable from 1.1:1 to 20:1 (LIMIT)
Blend: Variable from 0% to 100% Compressor
Gain: Variable from -6 dB to +20 dB
Attack: Variable from 20 mS to 75 mS (0.1mS with "FAST")
Release: Variable from 100mS to 2.2 Seconds

Power Consumption .65A @ 117 VAC, .35A @ 220 VAC

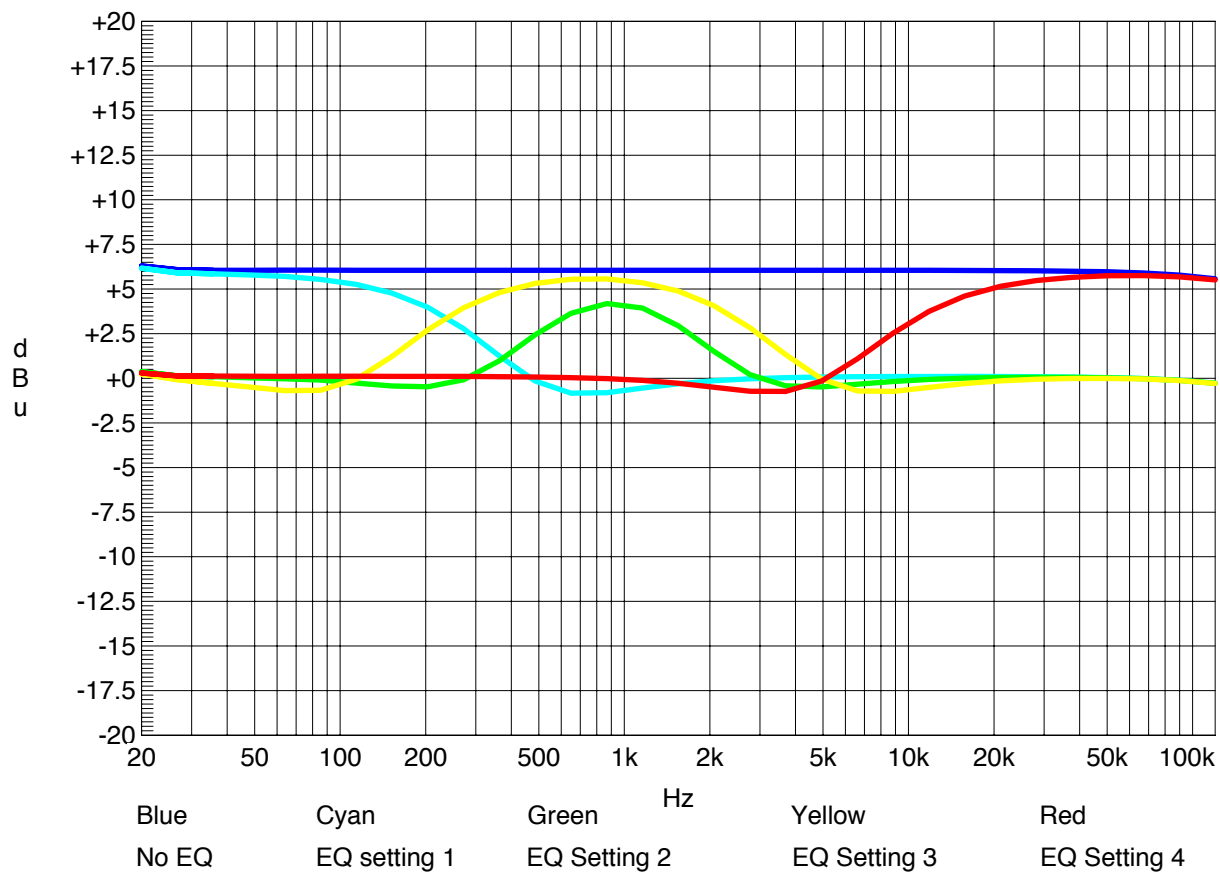
Fuse 5mm x 20mm, 2.5 Amp, fast acting, ceramic body - Bussman type GDA 2.5A or equivalent

Size 3.5"H (2U), 19"W, 12"D

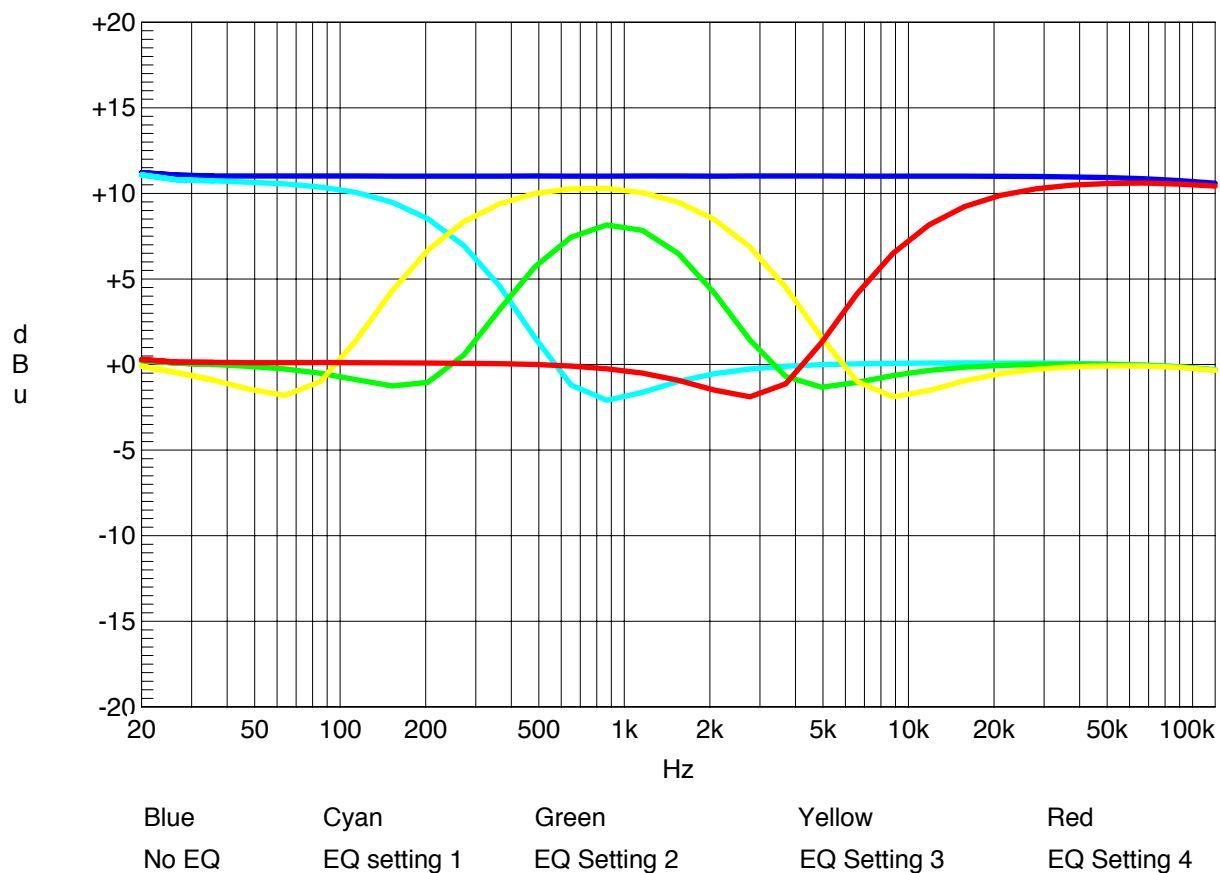
Shipping Size 5"H (2U), 23"W, 12"D

Shipping Weight 20 lbs

SFE Depth EQ Curves



SFE Width EQ Curves



The image displays the Portico II Master Buss Processor software interface, which is a digital emulation of a vintage analog audio processor. The interface is organized into several functional sections:

- Channel A and Channel B:** Each channel features a comprehensive set of controls including a 3-band EQ (Low, Mid, High), a compressor with attack, release, and threshold parameters, a limiter, and various routing options like 'Bypass All', 'Link', and 'ChA Master'.
- SFE (Stereo Field Effect):** This section includes controls for 'HF' (High Frequency) and 'LF' (Low Frequency) processing, as well as 'Depth' and 'Width' knobs to manipulate the stereo field.
- RUPERT NEVE DESIGNS:** This section contains a 'POWER' knob and two sets of frequency response curves (24, 22, 20, 18, 16, 12, 0 dB) for both 'CH A' and 'CH B'.
- PORTICO II Master Buss Processor:** The bottom section includes a 'Master Buss' section with a 'Width' knob, a 'Master Buss' EQ, and a 'Master Buss' compressor. It also features a 'Ratio' knob and a 'Threshold' knob for the master bus processing.

The interface is characterized by its high level of detail, with numerous knobs, sliders, and buttons that provide a wide range of audio processing options. The overall aesthetic is that of a professional-grade audio software application designed to replicate the sound and functionality of classic analog hardware.

Session:
Song:
Date:
Notes:

The image shows the Portico II Master Buss Processor software interface. It is a complex audio processing tool with multiple sections:

- Channel A and Channel B:** Each channel has a set of four circular frequency response graphs (HF, HM, LM, LF) and a series of sliders for parameters like Attack, Release, and Mix. There are also buttons for 'Bypass All', 'Link', and 'Ch A/Master'.
- SFE (Stereo Field Effect) Section:** This section includes a 'Rupert Neve Designs' logo and a 'Power' knob. It also features a 'Limit dBu' knob and a 'Ratio' knob.
- Bottom Section:** The bottom of the interface displays the 'Portico II Master Buss Processor' title and a 'Bypass All' button.

Session:
Song:
Date:
Notes:

Master Buss Processor: Recall Sheet

PORTICO II
Master Buss
Processor

Gain db

+10

0

+20

-20

-30dB

+20dB

Threshold

FF

20ms

80ms

Attack

FB

0

+8

-8

-20

-30dB

+20dB

Threshold

Blend %

0

100

SC

125 Hz

Texture

Min

Max

SC

Insert

Release

0.1

0.2

0.4

0.5

1.0

3.0s

Silk: Red

12

18

21

Limit dbu

Silk: Blue

10

12

15

Limit dbu

Ratio

1:1

40:1

5:1

Bypass All

LINK

CH A Master

Channel A

Channel B

Depth EQ

HF

HM

LM

LF

Depth

In

0

Min

Max

Depth

Width EQ

HF

HM

LM

LF

Width

In

0

Min

Max

Width

SFE

HF

HM

LM

LF

Depth

In

0

Min

Max

Depth

Width EQ

HF

HM

LM

LF

Width

In

0

Min

Max

Width

POWER

2

4

6

8

10

12

14

16

18

20

22

24

CH A

POWER

2

4

6

8

10

12

14

16

18

20

22

24

CH B

PORTICO II
Master Buss
Processor

Gain db

+10

0

+20

-20

-30dB

+20dB

Threshold

FF

20ms

80ms

Attack

FB

0

+8

-8

-20

-30dB

+20dB

Threshold

Blend %

0

100

SC

125 Hz

Texture

Min

Max

SC

Insert

Release

0.1

0.2

0.4

0.5

1.0

3.0s

Silk: Red

12

18

21

Limit dbu

Silk: Blue

10

12

15

Limit dbu

Ratio

1:1

40:1

5:1

Bypass All

LINK

CH A Master

Channel A

Channel B

Depth EQ

HF

HM

LM

LF

Depth

In

0

Min

Max

Depth

Width EQ

HF

HM

LM

LF

Width

In

0

Min

Max

Width

SFE

HF

HM

LM

LF

Depth

In

0

Min

Max

Depth

Width EQ

HF

HM

LM

LF

Width

In

0

Min

Max

Width

POWER

2

4

6

8

10

12

14

16

18

20

22

24

CH A

POWER

2

4

6

8

10

12

14

16

18

20

22

24

CH B

Session:
Song:
Date:
Notes:

PORTICO II
Master Buss
Processor

Gain db

+10

0

+20

-20

-30dB

+20dB

Threshold

FF

20ms

80ms

Attack

FB

0

+8

-8

-20

-30dB

+20dB

Threshold

Blend %

0

100

SC

125 Hz

Texture

Min

Max

SC

Insert

Release

0.1

0.2

0.4

0.5

1.0

3.0s

Silk: Red

12

18

21

Limit dbu

Silk: Blue

10

12

15

Limit dbu

Ratio

1:1

40:1

5:1

Bypass All

LINK

CH A Master

Channel A

Channel B

Depth EQ

HF

HM

LM

LF

Depth

In

0

Min

Max

Depth

Width EQ

HF

HM

LM

LF

Width

In

0

Min

Max

Width

SFE

HF

HM

LM

LF

Depth

In

0

Min

Max

Depth

Width EQ

HF

HM

LM

LF

Width

In

0

Min

Max

Width

POWER

2

4

6

8

10

12

14

16

18

20

22

24

CH A

POWER

2

4

6

8

10

12

14

16

18

20

22

24

CH B

PORTICO II
Master Buss
Processor

Gain db

+10

0

+20

-20

-30dB

+20dB

Threshold

FF

20ms

80ms

Attack

FB

0

+8

-8

-20

-30dB

+20dB

Threshold

Blend %

0

100

SC

125 Hz

Texture

Min

Max

SC

Insert

Release

0.1

0.2

0.4

0.5

1.0

3.0s

Silk: Red

12

18

21

Limit dbu

Silk: Blue

10

12

15

Limit dbu

Ratio

1:1

40:1

5:1

Bypass All

LINK

CH A Master

Channel A

Channel B

Depth EQ

HF

HM

LM

LF

Depth

In

0

Min

Max

Depth

Width EQ

HF

HM

LM

LF

Width

In

0

Min

Max

Width

SFE

HF

HM

LM

LF

Depth

In

0

Min

Max

Depth

Width EQ

HF

HM

LM

LF

Width

In

0

Min

Max

Width

POWER

2

4

6

8

10

12

14

16

18

20

22

24

CH A

POWER

2

4

6

8

10

12

14

16

18

20

22

24

CH B

Session:
Song:
Date:
Notes:

Notes

PRODUCT WARRANTY

Rupert Neve Designs warrants this product to be free from defects in materials and workmanship for a period of one (1) year from date of purchase, and agrees to remedy any defect identified within such one year period by, at our option, repairing or replacing the product.

LIMITATIONS AND EXCLUSIONS

This warranty, and any other express or implied warranty, does not apply to any product which has been improperly installed, subjected to usage for which the product was not designed, misused or abused, damaged during shipping, damaged by any dry cell battery, or which has been altered or modified in any way. This warranty is extended to the original end user purchaser only. A purchase receipt or other satisfactory proof of date of original purchase is required before any warranty service will be performed. THIS EXPRESS, LIMITED WARRANTY IS IN LIEU OF ALL OTHER WARRANTIES, EXPRESS OR IMPLIED, TO THE EXTENT ALLOWED UNDER APPLICABLE STATE LAW. IN NO EVENT SHALL RUPERT NEVE DESIGNS BE LIABLE FOR ANY SPECIAL, INCIDENTAL, OR CONSEQUENTIAL DAMAGES RESULTING FROM THE USE OF THIS PRODUCT. Some states do not allow the exclusion or limitation of consequential damages or limitations on how long an implied warranty lasts, so this exclusion may not apply to you.

WARRANTY SERVICE

If you suspect a defect in this product, please call us at 512-847-3013 or email us at support@rupertneve.com to discuss the suggested defect (it is possible that a suspected defect could be due to improper usage) and to obtain a return authorization number. It shall be your responsibility to pay for shipping the product to us, and, if the product is determined to be defective, our responsibility to pay for shipping the product back to you.



Rupert Neve Designs

PO Box 1969

Wimberley TX 78676

www.rupertneve.com

tel: +1 512-847-3013

fax: +1 512-847-8869