# Portico II

By: RUPERT NEVE DESIGNS

## *Master Buss Processor*

**Serial #:**


# Operations Manual

# 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 following list of safety concerns and power requirements before the use of this or any Portico II Series product.

## Safety

It’s usual to provide a list of “do’s and don’ts” under this heading but mostly these amount to common sense issues. However, here are important safety requirements that must be adhered to:

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 dry cloth.
7) Do not block any ventilation openings. Install in accordance with the manufacturer’s instructions.
8) Do not install near any heat source 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) Unplug this apparatus during lightning storms or when unused for long periods of time.
13) Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as when 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.
14) Do not expose this apparatus to rain or moisture.
15) The apparatus shall be connected to a mains socket outlet with a protective earthing connection.

Heat generated by the module is radiated through the case work and by convection through the ventilation holes, therefore the holes should not be covered or blocked. To avoid overheating, Portico II units should not be stacked immediately above or adjacent to other equipment that gets hot, and one rack space above the unit should be left open for heat ventilation. Also bear in mind that other equipment may radiate strong hum fields which could spoil the performance of your Portico II module.

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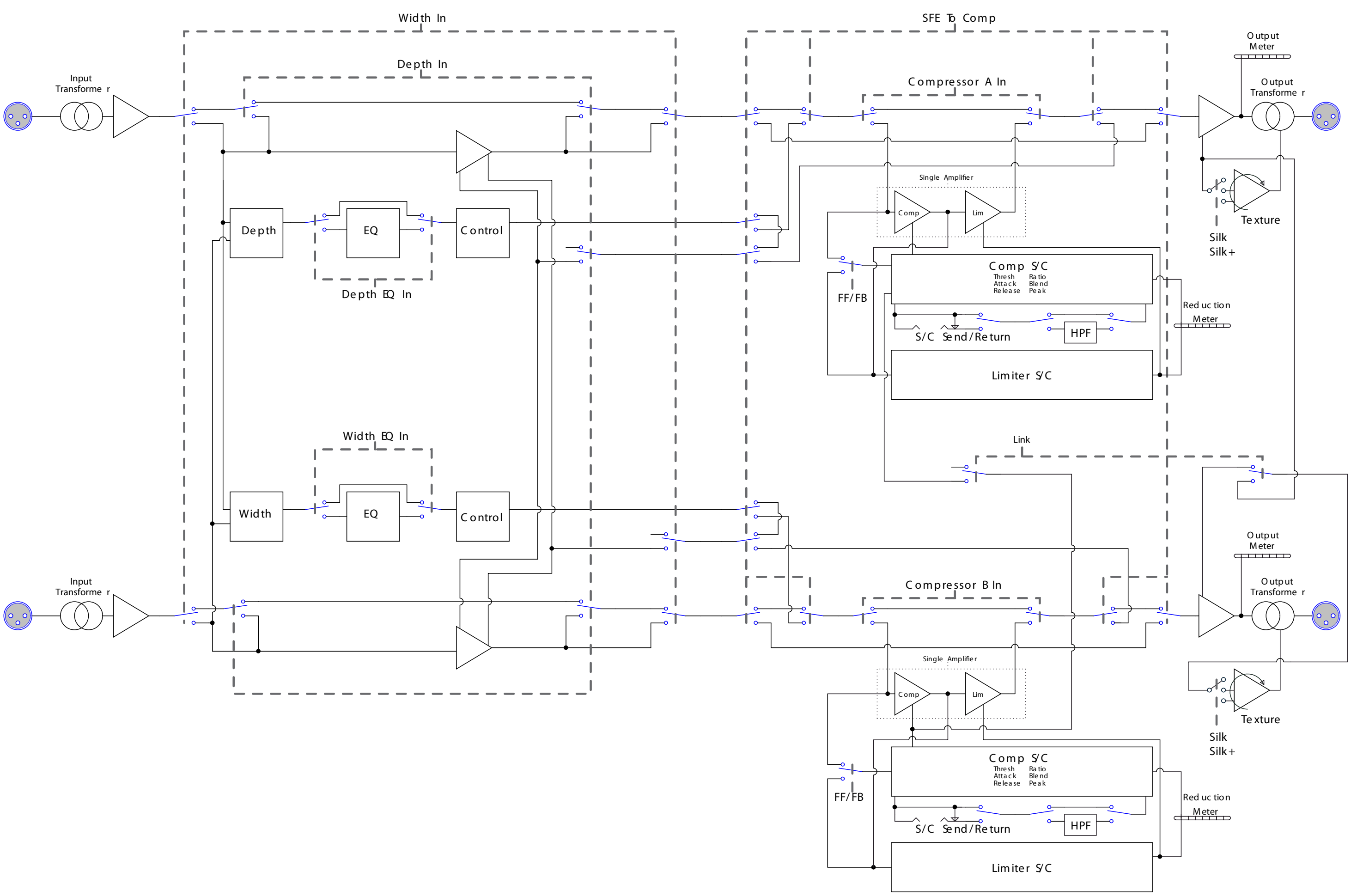
Don't operate your Portico II module in or around water! Electronic equipment and liquids are not good friends. If any liquid is spilled, such as soda, coffee, alcoholic or other drink, the sugars and acids will have a very detrimental effect. Sugar crystals act like little rectifiers and can produce noise (crackles, etc.). SWITCH OFF IMMEDIATELY because once current starts to flow the mixture hardens, can get very hot (burnt toffee!) and cause permanent and costly damage. If it gets wet and you suspect that good clean water may have gotten in, immediately unplug the unit, and remove it from the source of water. Please contact support as soon as possible at support@rupertneve.com for resolution. Clean only with dry cloth.

Don't be tempted to operate a Portico II unit with the cover removed. The cover provides magnetic screening from hum and R.F. stray fields.

## 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,Depart 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,Depart 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

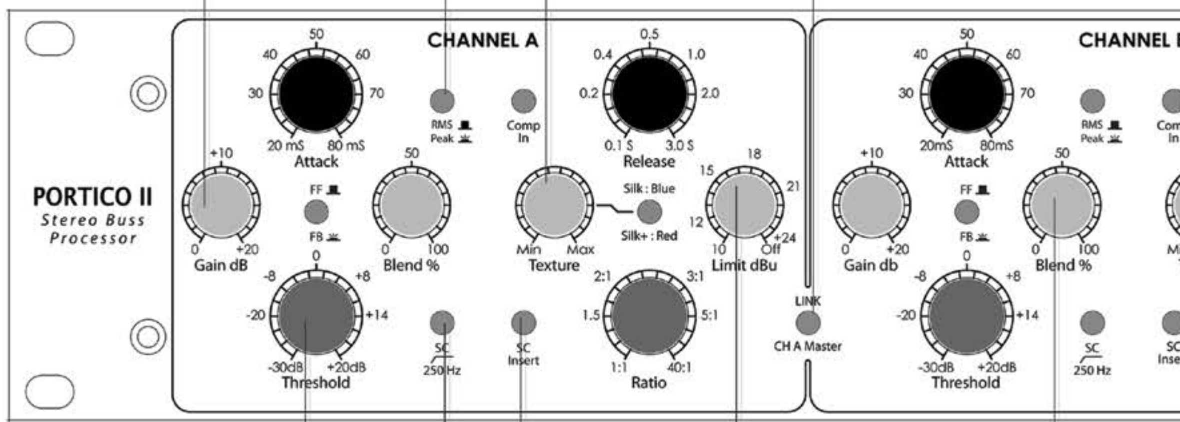
The compressor-limiter section is not bypassed with this button in. This may be the most useful control on the compressor because it is there for "confidence checks". In particular, exercise it in both the quietest and the loudest sections of the song. Watch out for the compressor and limiter exaggerating noise and room sound in the quiet sections and chopping off transients and consonants in the loud

**RMS / Peak**
Selects between RMS (average) and Peak detection. Peak detection mode changes the compressor attack to as fast as .1ms

**Texture / Silk**
Continuously variable control for Silk Red & Silk Blue modes allows the engineer to fine tune the harmonic ratio

**Link**
Locks together the VCA response of both channels and links the controls with CH. A as Master.
***Silk modes must be matched before link is engaged

**Make Up Gain**
0 to 20dB of makeup gain




**Threshold**
Sets the level at which the compressor activates

**Limit**
Sets the threshold for the independent brick wall limiter

**SC HPF**
Introduces a high pass filter into the compressor side-chain circuit

**SC Insert**
Insert send and return for the compressor side-chain

**Blend**
Controls the mixture of Compressed and uncompressed signals

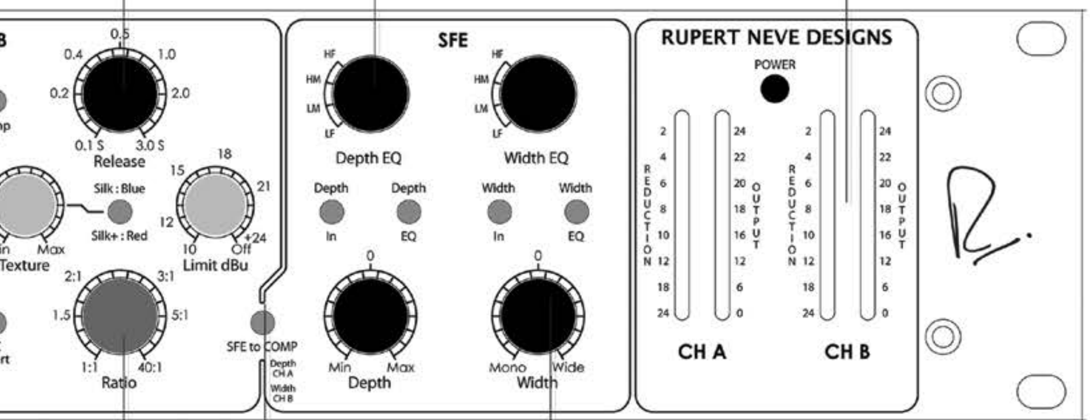
**AC Input**
IEC input jack for 100-240V AC


# Portico II Master Buss Processor: Front / Back Panel

### Attack/ Release
Determines the speed at which the compressor reacts once level is over or under the determined threshold

### Depth & Depth EQ
Increases or decreases the amount of mid information in the stereo signal. The engaging the EQ makes the depth affect only the selected frequency band

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

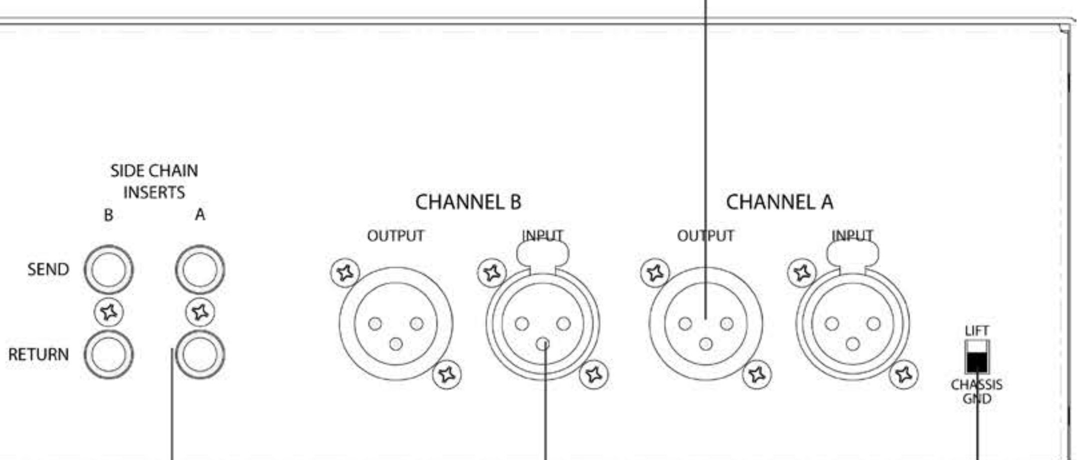



### SFE to Comp
Routes the SFE M–S Signals into the two compressors, with mids in CH. A. and sides in CH. B.

### Width & Width EQ
Increases or decreases the amount of side information in the stereo signal.
Engaging the EQ makes the width affect only the selected frequency band

### Ratio
Sets the slope of compression about the set threshold

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




### Side Chain
TRS Send and return that effects the compressor VCA. The return may also be used for a Key input.

### Line Input
Discrete and transformer balanced input. Can handle signals up to +25dBU.
1= GND 2= Hot 3= Cold

### Chassis GND Lift
Seperates the direct connection from audio signal ground to chassis earth

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 sets how fast the compressor returns back to zero after attenuating. Typically engineers have used quite slow releases when the need is to minimize any obvious compressor action or gains changing. For modern pop music quite often the goal is to have the compressor change gains approximately in time with the music and the RELEASE control setting becomes important for this. On the other hand, many of us were trained to avoid having compressors "pump" and sound like they are breathing. Another (maybe too) common use of compressors is to maximize the apparent volume and for this one generally wants very fast release settings so that after reducing peaks the compressor returns to maximum levels as quickly as possible. And by "as possible" there is a practical matter regarding the tendency of compressor/limiters to introduce a nasty form of distortion called "modulation distortion" when they are set for a combination of fast attacks, fast releases and high ratios.

**S/C HPF**

This routes the 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 scoff at the single knob operation, however the independent limiter is extremely intelligent, knowing how to appropriately respond to the various signals presented to it. Our new Serial Adaptive Release Technology is behind this revolutionary performance.

Using a blend of release time constants, this limiter will simultaneously respond quickly to transient material (such as the “snap” of a snare drum) and slowly to more sluggish signals (such as a bass guitar) when the set threshold is surpassed. This configuration allows the limiter to grab a transient and let go just an instant later, while also dealing with more constant signals in a slower, more musical way. In this manner, the MBP Limiter can provide a much more aggressive amount of limiting than typically possible, while maintaining the essential character of the music and remaining free of the modulation distortion usually found in a fast acting limiter.

Typically there is a trade-off between how fast limiter can react and the amount of modulation distortion in the lower frequencies. This is due to the lower frequencies finding their way into the side chain signal, triggering the compressor on and off very quickly, which ends up modulating the overall signal. This is interesting to look at with sine waves, but sounds quite undesirable with music. The MBP does not have this trade-off, and one is able to have the best of both worlds: a quick, snappy response while maintaining the integrity and smoothness of the low end. In addition to the adaptive time constant circuitry, the release time is also varied with the position of the knob. As the knob is turned counter clock-wise, the release time is increased accordingly, as typically one would want a longer release time with a larger amount of reduction.

The limiter found in the MBP is designed to respond as fast as .03 mS in order to reduce the first half of a 20 kHz waveform over the threshold. It has a “medium knee” initial ratio and within 3 dB of the threshold attains a better than 10:1 ratio. A soft clipper circuit catches transients that may have been in the “knee” when the threshold knob is set quite high. Both the limiter and soft clipper are switched out of circuit with the knob is fully clockwise. The release times are fully automatic and adjust depending both on the average depth of limiting and the duration of the transients above the threshold. The limiters share the same discrete, class-A gain module and VCA with the compressors, so using the Limiter does not introduce more stages that the music would have to pass through. This combination of features provides exceptionally transparent limiting, and often allows twice as much gain reduction compared to other limiters before objectionable artifacts become apparent.

### 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 width of a stereo image (wide/mono) and adjust the amount of ambience inherent in the recording. As the width control is rotated toward wide, the amount of difference material is boosted, often resulting in more ambient material, and accentuated stereo reverbs. Conversely, the stereo field is contracted when rotated to mono, and, if the left and right channels are highly coherent (i.e. both channels include closely similar material that is in phase), this mono content is enhanced. If the phase of one of the input channels is then reversed the mono content may be virtually eliminated. Because the amount of effect the width control has is entirely dependent on the amount of stereo information in the original source material and the interplay between the stereo field editors other controls, listening and experimentation are essential for the best results.

### DEPTH

The depth control of the MBP adjusts the spatial positioning of elements in the sound stage. Center-panned elements like solo instrument or vocal can be brought forward in a mix, in relation to supporting instruments. In many cases, these same elements may be virtually eliminated without adversely affecting the music bed. Used in conjunction, the depth and width controls effectively alter the perceived room ambience and dimension.

**WIDTH & DEPTH EQ**

To fine tune the SFE, there are individual filters that allow a fine tuning of what information is reintroduced from the width and depth circuits, thus tailoring each effect to a specific bandwidth. For example, if one wanted to increase the amount of low frequencies in the center image, engaging the SFE Depth and Depth EQ, set to LF, would filter out everything in the Mid signal except what is below the filter point (in this case, 250 Hz), and once reintroduced to the original would result in a perceived increase in the low frequencies in the center image. It is also possible to do the same thing with the Width EQ, except instead of boosting the width, cutting it, which removes low frequencies from the Sides, tightening up the low frequency perception in the center. Using the Width EQ again, this time set to HM (or LM as the case may be), increasing the amount of band-passed Side information can provide a wonderful spreading of instruments, reverberation and background vocals, giving the illusion that the sounds are spread further out, enveloping 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 O 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

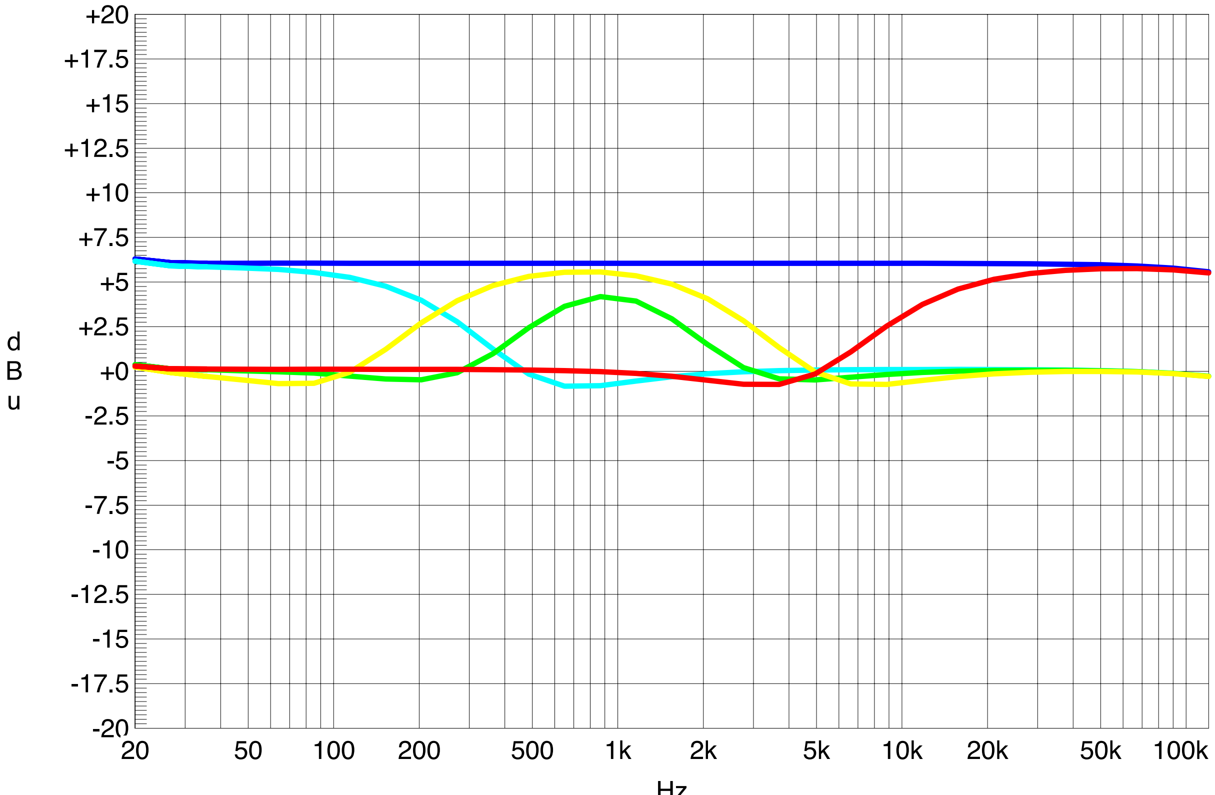
**Compressor**

**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

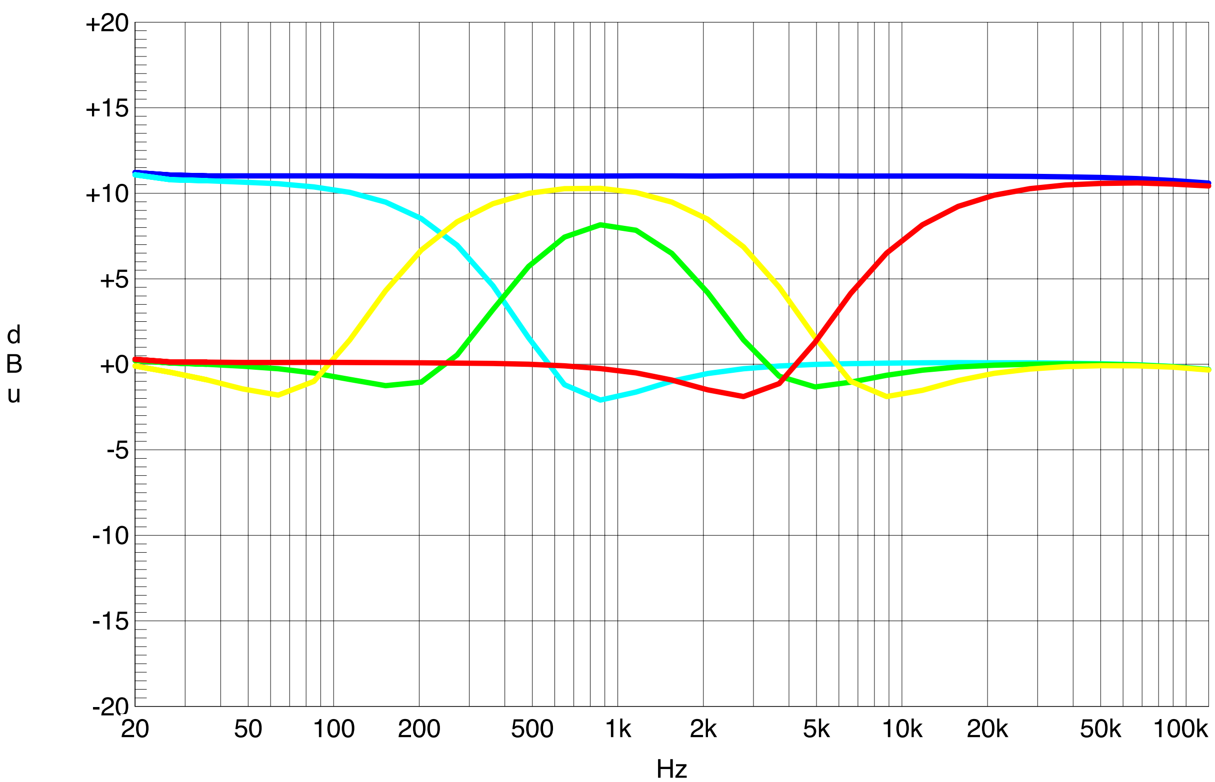
**Size** 3.5″H (2U), 19″W, 12″D
**Shipping Size** 5″H (2U), 23″W, 12″D
**Shipping Weight** 20 lbs

## SFE Depth EQ Curves



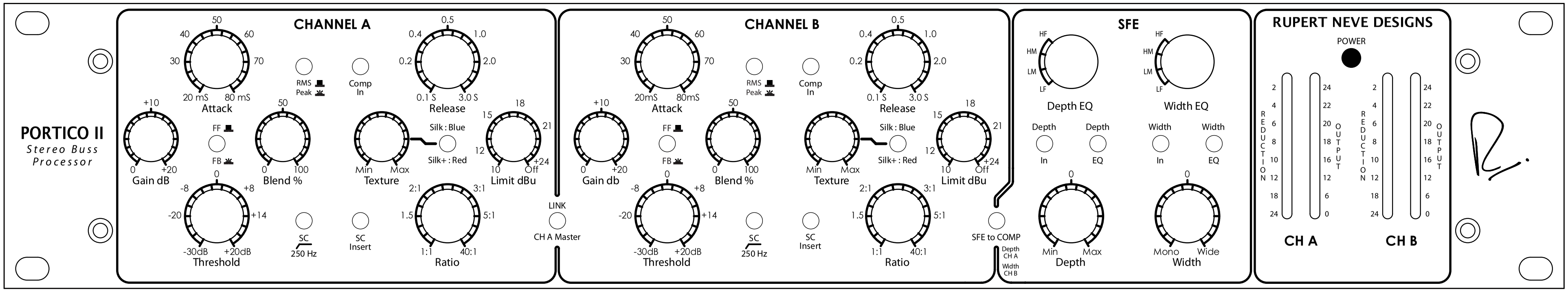

| Blue | Cyan | Green | Yellow | Red |
|---|---|---|---|---|
| No EQ | EQ setting 1 | EQ Setting 2 | EQ Setting 3 | EQ Setting 4 |

## SFE Width EQ Curves

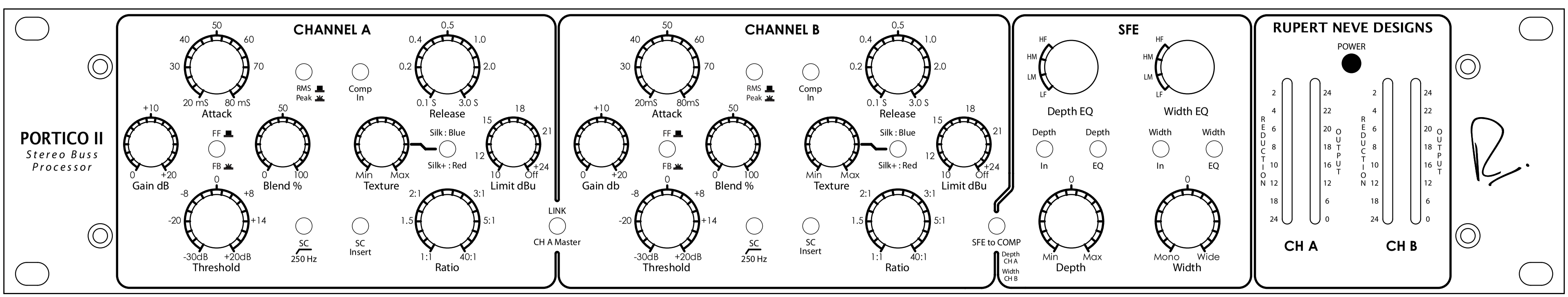



| Blue | Cyan | Green | Yellow | Red |
|---|---|---|---|---|
| No EQ | EQ setting 1 | EQ Setting 2 | EQ Setting 3 | EQ Setting 4 |

# Master Buss Processor: Recall Sheet

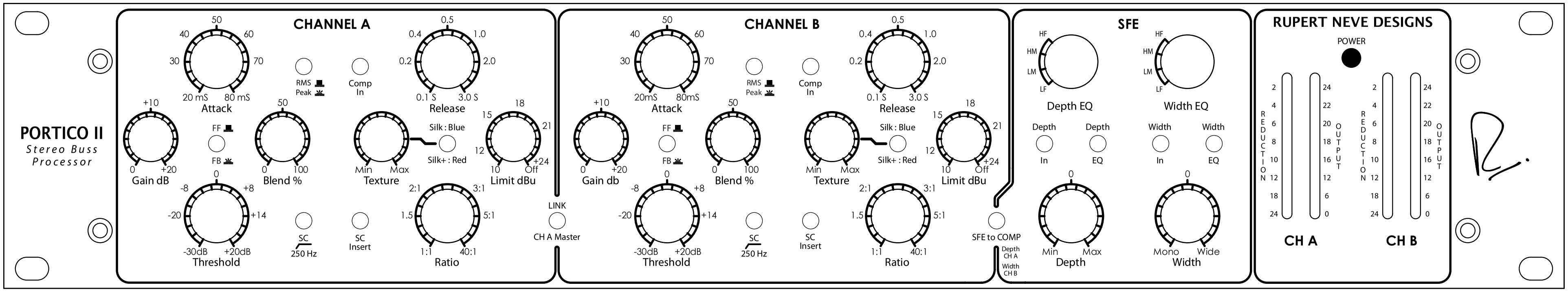



Session:
Song:
Date:
Notes:

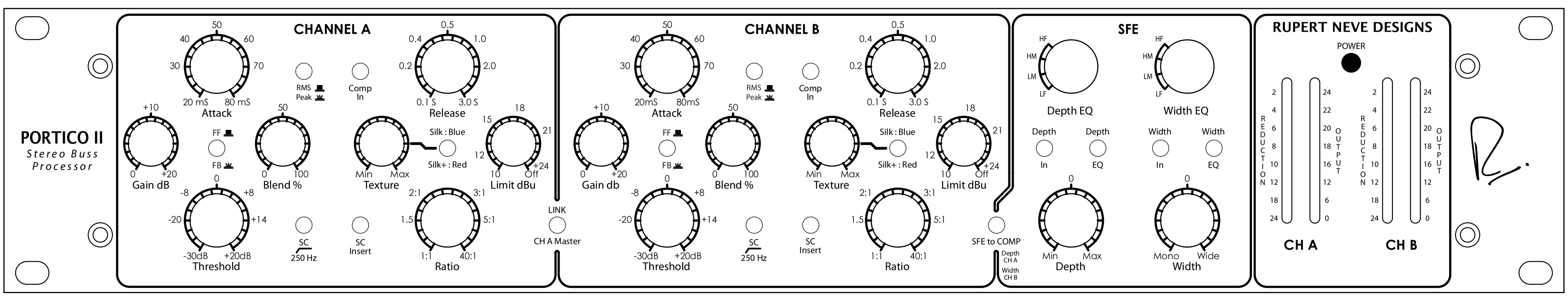



Session:
Song:
Date:
Notes:

# Master Buss Processor: Recall Sheet




Session:
Song:
Date:
Notes:




Session:
Song:
Date:
Notes:

## 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 EXTEND 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.

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