5015 User Guide

Thank you for your purchase of the Portico™ 5015 Microphone Pre-amplifier and Compressor module. Everyone at Rupert Neve Designs hope you enjoy using this tool as much as we have enjoyed designing and building it. Please take note of the following list of safety concerns and power requirements before the use of this or any Portico 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 some reminders:

Heat generated by the 5015 is radiated through the case work and by convection through the ventilation holes, therefore the holes should not be covered or blocked. Portico modules may be stacked horizontally on a desktop or mounted vertically in a rack without heat problems. The anti-slip feet may be removed while used in a rack, but should be retained for desktop use. To avoid overheating, Portico™ modules should not be stacked immediately above or adjacent to other equipment that gets hot. Also bear in mind that other equipment may radiate strong hum fields which could spoil the performance of your Portico module.

Don't operate your Portico 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.

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

Power Requirements

Each Portico 5015 module has a high quality DC to DC converter that provides carefully stabilized and filtered +/- 17.5 VDC for the amplifiers. The meticulous audio quality of your PorticoTM is protected by the internal converter and does not depend primarily on the external mains power supply. The input is protected from reverse polarity. The connector center pin must be positive.

The converters will work from any DC supply from 9 to 18 volts that is reasonably "clean". The power supply normally provided with the 5015 is a high quality, robust, and very reliable switched mode power supply. There are no special requirements for the Portico low voltage units other than that they must be of good quality, reliable, and able to supply enough current for the number of modules in use.

The great advantage of this system is that there are no common D.C. supply rails that are directly shared by other modules.

INTRODUCTION

In a traditional console large, bi-polar regulated supplies were used, necessarily having a shared common 0 "ground" wire. Crosstalk between modules resulted, often accompanied by R.F. interference due to the unbalanced loop "antennas" that were inevitably present. This interference, in some cases, could actually be heard, but even at low levels below audibility there was a potential intermodulation with the desired signal. Of course this represented both a quantitative and subjective intrusion affecting sound quality.

One of the advantages of the Portico method of feeding equipment is that external power units will work from almost any of the very wide range of mains supply voltages and frequencies that are found world-wide. While many different types of mains power wall sockets are found in different countries, Portico 5015 module power units leave the factory with standard US plugs. If required, any suitable connecting cord may be substituted.

Avoid using a mains power outlet that is on the same circuit as air conditioning of other equipment that regularly switches on and off. Unplug your Portico power units during a thunder storm or if it will be unused for a long period.

Portico modules can alternatively be powered from a 12 volt battery, in which case the supplied AC power unit is not needed. When using a 12 volt battery, choose one that has enough capacity to power your Portico 5015 - or your complete assembly of Portico modules - for the expected duration of your session.

FEATURE LIST

The Rupert Neve Designs Portico™ 5015
MICROPHONE PRE-AMPLIFIER AND COMPRESSOR

The Rupert Neve Designs 5015 module is a half rack width, 1.75" (1U) module in the Portico style. As with the entire Portico™ 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. Alternative front panel layouts are available providing a choice of vertical or horizontal mounting. When the horizontal front panel is chosen, a single 5015 can sit firmly on a bench or desktop on its detachable rubber feet. Two 5015s can be joined with the optional Horizontal Joining Kit, model number 5221-RM, and mounted across a standard 19" rack.

When the vertical option is chosen, up to eight 5015s can be mounted in the optional vertical frame, model number 5285-RM, or they may be utilized in the 5088 penthouse frame. The vertical frame assembly is designed for rack mounting and includes basic rear cable management. Blank panels are available to fill any unused spaces when the full complement of eight modules is not fitted into the vertical frame.

Both Mic Pre and Compressor channels are transformer coupled on both the inputs and outputs, so their operation is completely independent.

The Microphone Preamplifier Section

MICROPHONE INPUT
The microphone input is balanced but not floating, being a variant of an instrumentation amplifier. My well-proven "Transformer-Like-Amplifier" (T.L.A.) configuration is used, which includes an accurate toroidal Common Mode Low Pass Filter that rejects Common Mode signals and excludes frequencies above 150 kHz. (There are high powered broadcast transmitters at and above this frequency in several Continents and, even if you can't hear them, any vestigial intermodulation products must be excluded!)

When the Mic Gain switch and Trim controls are set to Unity (0 dB) the Portico™ 5015 microphone pre-amplifier can handle a balanced input signal of more than +20 dBu without an input attenuator pad! This is a unique feature that enables this input to double as an additional line input.

THE MIC PRE OUTPUT
The output stage is identical with that of the Portico™ 5012, using single-sided circuitry, driving a carefully configured output transformer that can deliver a full +25dBu from the balanced and ground-free secondary winding.

This maximum output level provides a large margin over and above the likely maximum requirement of any destination equipment to which the Portico™ 5015 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 Portico™ modules. My original classic 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. 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™ transformer-coupled module 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 BUSS OUTPUT
The Buss output is balanced and high impedance, and the BUSS connection is derived pre-mute. It is intended for use with modules in the Portico™ range which are equipped with a matching MIX or BUSS input. The Portico™ 5015 BUSS output has dual, paralleled, TRS connectors that allow any number of Portico™ modules to be mixed to the BUSS input on any of these appropriate modules using a standard TRS patch cord.

MICROPHONE PREAMPLIFIER DESIGN NOTES

In former years, before the introduction of solid state amplifiers, transformers were necessary to step up to the very high input impedance of tubes, and to provide a balanced input for the microphone line. An input impedance of 1,000 or 1,200 ohms became established for microphones having a source impedance of 150 or 200 ohms, with connection being made on a twisted twin screened cable (This type of cable, while excellent for low impedance work, has high capacitance between its conductors and between each conductor and screen. Resultant high frequency losses are excessive with piezo pickups and may cause resonances with magnetic pickups.) Thus microphones were not heavily loaded. Condenser microphones worked off high voltage supplies (300V!) on the studio floor which polarized the diaphragms and powered a built-in pre-amplifier. More and more microphones were needed as "Pop" music gained ground and this led to the popular and efficient method of 48-volt "Phantom" powering that was built into the multi-channel recording Console - in place of numerous bulky supplies littering the studio, a miniature pre-amplifier now being fitted inside the microphone casing.

The 48-volt supply was fed to the microphone through balancing resistors so it was impossible for this voltage to actually reach the microphone, resulting in low polarizing volts and virtual starvation of the little pre-amp inside the microphone. Nevertheless amazingly good microphones were designed and made, becoming the familiar product we use today. If a low value resistive load is connected to the output of an amplifier, that amplifier has to produce power in order to maintain a voltage across that load. Obviously if we want more voltage (output from the microphone) we need to provide a larger supply for the amplifier or settle for a lighter load. A microphone is a voltage generator, not a power amplifier. Most microphones give their most accurate performance when they are not loaded by the input impedance of a traditional preamplifier. If the microphone uses an electronic circuit (transformerless) output, a low value of load impedance will likely stress the little microphone pre-amplifier, causing slew rate and compression at high levels.

On the other hand, a high value of load impedance allows the microphone to "breathe" and give of its best, this being particularly advantageous with very high level percussive sounds. If the microphone has an inductive source (such as would be the case if it has a transformer output) a low value of load impedance causes the high frequencies to roll off due to leakage inductance in the transformer in addition to the above amplifier distortion (This can be an advantage with some microphones!).

For this reason I have provided a high value of input impedance that will load microphones to the smallest possible extent and makes the best possible use of that limited "Phantom" 48-volts supply.

A NOTE ON DISTORTION

The human hearing system unconsciously perceives minute signals both within and well beyond the traditional audio frequency spectrum. When such frequencies that are not present in nature, such as high order harmonics produced by amplifier distortion, or by the presence of seemingly inaudible artifacts due to interference, or when those frequencies that are present in nature are missing, the human hearing system "reports" what we might describe as a deviation from faithfulness. It seems that we store a bank of information based on "natural" sound and are able, subconsciously, to compare reproduced sound with "natural" sound. This deviation from faithfulness gives rise to a feeling of discomfort and frustration that is very hard to describe or to explain. However, amazingly, it can be measured! When subject to nonfaithful sound, the brain actually emits electric activity that can be measured. (See "Audible Range Affects Brain Electric Activity and Sound Perception" (Ref: 1).

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!

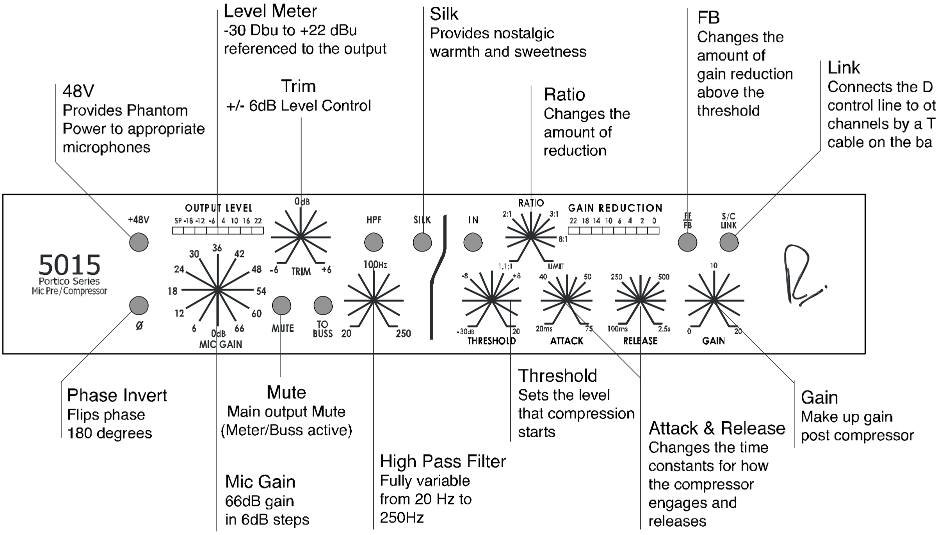
The way in which an analog amplifier handles very small signals is as important as the way it behaves at high levels. For low distortion, an analog amplifier must have a linear transfer characteristic, in other words, the output signal must be an exact replica of the input signal, differing only in magnitude. 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.

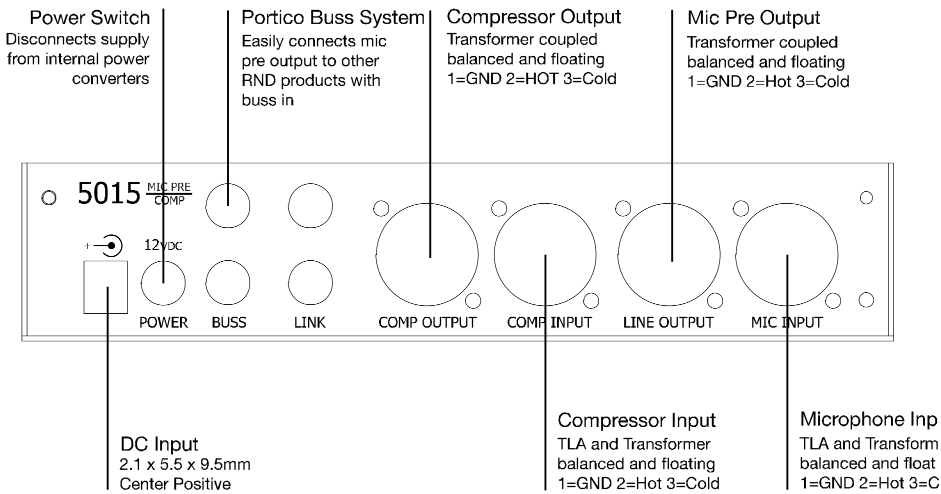
In order to control gain, a V.C.A. or Voltage Controlled Amplifier (or Attenuator) is used. There are many types of V.C.A.'s that include the use of tubes, discrete and integrated solid state circuits, or naturally non-linear devices. Each one has its own characteristic behavior that reflects sonically on the final performance, and, inevitably, gives it a character or signature that can be musically attractive or - not! In order to achieve a sonically musical signature, low level signals below the
"Threshold" level at which the VCA starts to operate, must be treated linearly, to avoid low level distortion.

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.

Portico 5015 Front Panel



Portico 5015 Back Panel



MIC PRE CONTROLS

MIC GAIN
A 12-way precision rotary switch covering from 0 to 66 dB in 6 dB steps.
Selecting the right gain optimizes Noise and Headroom.
TRIM
Provides further gain adjustment, continuously over a range of +/- 6 dB.
+48V
Push button makes phantom power available at the microphone input.
Ø
Push button inverts the phase of the signal path.
MUTE
Disconnects the main output.
SILK
Much could be written about this feature, but suffice to say that it gives a subtle option to enhance sound quality in the direction of vintage modules. The SILK button reduces negative feedback and adjusts the frequency spectrum to provide a very sweet and musical performance. We suggest you try it and make your own judgment.
HPF
A 12 dB/octave high pass filter providing continuously variable Low Frequency attenuation between 20 Hz and 250 Hz. The high pass filter is a valuable aid in any signal chain but particularly so in a microphone preamplifier. Signals within this band can be attenuated, leaving higher frequencies unaffected.
Helps get rid of building rumble, air handling motor hum etc.

METER
An eight segment LED bar-graph meter is fitted for the mic pre, calibrated in dBu as follows:
SP, - 18dBu, - 12dBu, -6dBu,
+4 dBu, + 10dBu, + 16dBu, +22dBu
With reference to the balanced output signal level.
The input level can be determined by reading the Meter indication, then subtracting the Gain settings of the Sensitivity switch and the Trim control. For example, if the meter is reading + 10 dBu, with the Trim control at, say, +2 and the main Gain switch is at 42, the level of the input signal is 10 - 42 -2 = -34 dBu.

Compressor-Limiter Section

THE NEED FOR DYNAMIC CONTROL OF SOUND LEVELS
The dynamic range of sounds we hear around us in normal life greatly exceeds the capability of our best recording and processing equipment - but even if this were not so, the scale of dynamic range must be accommodated to the venue in which it is to be reproduced. For example, actual volume levels of the dance hall would be deafening in a students bedroom. In the same way, late night listening in a quiet living room demands careful adjustment of dynamic range. In the constantly changing background noise of a car, drama dialog does not work without constant attention to the level control. In the field of communications, it is often necessary to ensure that the best possible signal-to-noise ratio is obtained, in the interest of intelligibility, within the limited performance of, say, a reporter's recording device.

Digital recorders are unforgiving when overloaded. Overload can be avoided with careful use of high ratio compression - on the verge of limiting - with careful choice of time constants. A recording that still sounds "loud" can be produced without non-musical harmonic distortion. A compressor-limiter is one of the most powerful, yet subjective items in the sound engineer's armory. Compression should never be obvious to the listener and this needs intuitive and effective controls on the part of the designer together with considerable skill on the part of the sound engineer.

THE WAY THE 5015 Compressor WORKS
A part of the audio signal is rectified and smoothed to produce a suitable control voltage for the V.C.A. which has to respond very quickly and have low distortion. If the response is too fast, low frequency signals will themselves, be "gain controlled"! If the response is too slow, the signal will overshoot and the first few cycles will not get compressed. The speed and accuracy of the response, known as the "attack", and the time frame that gain remains under the initial control, known as "release" or "recovery", play a large part in the way a compressor sounds.

The Portico 5015 makes use of a very accurate, low noise, low distortion V.C.A. having, essentially, no "signature" of its own. This leaves the designer free to use amplifier and transformer combinations that are well proven and produce the desired sonic quality.

All Portico modules use input and output transformers and almost entirely discrete component amplifiers to produce the musical "signature" for which they are known. These are factors that enable the Portico 5015 to work unobtrusively within the context of a very high quality audio chain.

Compressor Controls

THE CONTROL CHAIN: FEED FORWARD or FEED BACK
If the V.C.A. Control voltage is taken from the 5015 input, (i.e. before the V.C.A.) the V.C.A. "knows" right away that a gain change is required and there is almost immediate response. This is known, logically, as a "Feed-Forward" compressor.

If the V.C.A. control voltage is taken from the 5015 output, (i.e. after the V.C.A.) it cannot act immediately on the V.C.A. because it has already been modified by settings of the V.C.A. and circuits through which it has passed. This is known as a "Feed-Back" compressor. The two compression characteristics are quite different, there is more "overshoot" and both the attack and recovery ramps are changed, providing the user with powerful choices. A choice between "Feed-Forward" and "Feed-Back" circuitry is provided. Almost all Mr. Rupert Neve's earlier designs were "Feed-Back". They were more musical and sweeter than with "Feed- Forward" designs. The way in which these modes change the dynamic performance can be seen in the above graph - but the more interesting effects are noted by listening - "Feed-Back" produces a sweeter, warmer sound but is not as accurate if you need to protect a transmitter, for example.

THE 5015 LINE AMPLIFIER
The Portico™ 5015 consists of two identical Line Driving amplifiers having transformer balanced inputs and outputs. The sonic quality of these amplifiers is such that, by providing galvanic isolation, simple single-sided circuit topology, and freedom from grounding problems, they are capable of enhancing the sonic quality of many signal sources, especially those of digital origin. The sonic "signature" is one of extreme purity and the image is consistent with that of Mr. Rupert Neve's original designs of 35 - 40 years ago. More detailed discussion on the sonic image and the

way in which an analog designer's approach can sweeten and "warm" some of the cold, storage an editing processes may be found on www.rupertneve.com.

COMPRESSION
For signals below the "threshold" level that has been set, a compressor provides a linear pat allowing signals to be amplified without the gain being adjusted in any way. When signals excee the "threshold" level, the gain is reduced in a controlled manner that depends on the Ratio setting.

RATIO
Range 1:1 to LIMIT (i.e. 40:1)
Above a given THRESHOLD signals are reduced by an adjustable amount ranging from 1:1 (which is linear, or no reduction at all), to more than 40:1 which is a very high ratio, equivalent t that of a limiter.

RATIO is sometimes referred to as "Slope" because when depicted on a graph, the "slope" of th graph representing Output versus Input, is what changes. RATIO is measured in dB (Decibels). Th 40:1 figure mentioned above, means that a change in input signal level of 40 dB only results in dB change in output level!

RATIO and THRESHOLD are closely inter-dependent. If a RATIO as high as 40:1 has been se and if the THRESHOLD is set at 0 dBu, then even when a massive signal of +40 dBu (unlikely!) i presented to the input, the output signal will only be +1 dBu. RATIOS as high as this woul normally be set somewhere above 0 dBu - say at +14 dBu, in order to prevent the output signa level exceeding just over +14 dBu to protect, for example, a digital recorder. Similarly, if a RATI of 5:1 has been set, an input signal which is 10dB above THRESHOLD will only rise by 2d above that THRESHOLD at the output.

FEATURES
THRESHOLD
THRESHOLD control covers the range from below -30dB to +22dBu.
When THRESHOLD is set at a low level with a fairly high RATIO, the amount of gain reductio will be considerable and it may be necessary to use some GAIN after the compressor to restore th apparent signal level.

RELEASE (or RECOVERY) and ATTACK TIME
Range of RELEASE time is 100 mS to 2.5 Seconds.
Range of ATTACK time is 20 mS to 75 mS.
The notes above explain how the 5015 handles signals of constant amplitude such as pure tones Real program signals, however, are continually changing in level. The way in which a compresso deals with actual program material depends upon the magnitude and duration of peaks in th program level.

If the RELEASE TIME is set to be very short, a short duration signal will be compressed but th gain will return to normal very quickly, giving a fluctuating and un-natural sound known a "Pumping" when the background, or other signals, are forced up and down. The gain will also ten to follow the wave form of low frequency signals. RELEASE TIME should be set long enough fo the gain to remain reasonably constant between each bass note or between speech syllables. Th ATTACK time is the time taken for the compression circuits to start compressing. A long ATTAC time allows short duration peaks to "escape" and go through uncompressed. This may cause overload on subsequent digital circuits. A very short attack time sounds un-natural and robs the signal of "life" by removing transients. Some transients are extremely fast and have little effect on the sound quality. Setting a long attack time often means that almost no gain reduction occurs because the transient is history (!) before compression has had time to operate.

However, even the fastest circuits take time to operate which means that there is always some "Overshoot". Small amounts of "Overshoot" are musically desirable - there are exceptions, of course.
Setting the right values of RELEASE and ATTACK is what compression is all about! Once the principles are understood a Compressor-Limiter such as the 5015 provides a powerful tool that actually appears to enhance the dynamic range of a recording and so provide greater musical enjoyment.

GAIN
GAIN range provided is from -6dB to +20 dB.
As already noted, when compression has taken place, it may be necessary to increase the overall gain to restore the apparent program level

FB BUTTON
Switches from "Feed-Forward" compression mode to "Feed-Back" compression mode as described earlier in this user guide.

COMPOSITE OPERATION
Cascading two compression sections with two 5015's or a 5015 and a channel of a 5043 A and B, provides extremely powerful and comprehensive control of dynamic range. When an external audio connection is made from the output of one channel to the input of the other, the overall compression characteristics become a composite of the two. For example, channel A may be set up as a low ratio compressor and channel B may be set up as a high ratio compressor or limiter to produce a powerful composite characteristic. (as shown in the graph below)

STEREO
When the LINK push-button is engaged and the link buss is connected to another Rupert Neve Designs compressor with the two channels, A and B, set to approximately the same values, GAIN control on both channels will be the same to preserve stereo balance, levels normally being controlled by the channel with the higher signal level.

DUCKING
When the LINK push-button is Engaged and the link buss is connected to another Rupert Neve Designs compressor, the signal passing through channel A, may be used to control the amplitude of channel B. For example, the level of music through channel A can be controlled by speech on channel B, i.e. reducing the music level to make a "speech-over" announcement.

METERS
Two LED METERS are provided, OUTPUT LEVEL and GAIN REDUCTION.
These are switched between channel A and channel B on the central METER SELECT push-button.

OUTPUT LEVEL indicates the actual output level in dBu.
The GREEN segments cover the range from -10 to +10 dBu.
The YELLOW segments cover the range from +10 to +20 dBu and the RED segment, +22 dBu and above which is regarded as overload.

REDUCTION is calibrated in dB covering the range 1 to 22 dB, reading backwards.
This meter indicates the amount of actual reduction taking place and therefore is a valuable point of reference indicating the operation point of the 5015.

5015 Specifications

Mic Pre
Frequency Response:
Main Output, no load,
-3 dB @ 18 Hz
-3 dB @ 160 kHz
Noise:
Measured at Main Output, unweighted, 22Hz-22kHz,
Terminated 150 Ohms.
With gain at unity better than -100 dBu
With gain at 66 dB better than -62 dBu

Equivalent Input Noise better than -128 dBu

High Pass Filter:
Continuously variable swept frequency from 20 Hz to 250 Hz.
Slope: 12 dB/Octave
Gain:
Unity to +66dB in 6 dB steps,
Trim continuously adjustable from -6dB to +6dB
Buss Output:
Output is designed to feed a Buss-mix Amplifier (ie. Buss inputs on 5043) at the internal system level of -2.5 dBu.
Maximum Output Level:
Maximum output from 20 Hz to 40 kHz is +24 dBu.
Mute:
Mutes Main Output only.
Phantom Power:
+ 48 Volts DC +/- 1%
Total Harmonic Distortion and Noise:
@ 1kHz, +20 dBu output:
Main Output: Better than 0.002%
@ 20Hz, +20 dBu output:
Main Output: Better than 0.020%
Silk Engaged: Better than 0.2% Second Harmonic

Compressor

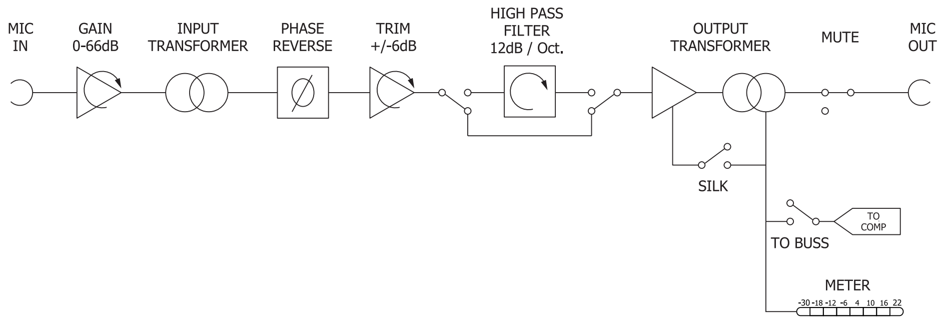
Gain Range:
Continuously variable from -6 dB to +20 dB.
Threshold Range:
Continuously variable from -36 dB to +22 dB.
Ratio Range:
Continuously variable from 1.1:1 to "Limit" (40:1).
Attack Range:
Continuously variable from 20mS to 75mS
Release Range:
Continuously variable from 100mS to 2.5 Seconds.
FF/FB:
Feed-Forward or Feed-Back VCA control.
LINK:
Multiple 5015's may be daisy-chained via the rear panel jacks. When an individual channel is engaged, its control voltage appears at the rear panel LINK jack.

Maximum Output Level:
Balanced and Floating Transformer Output +25 dBu.
Total Harmonic Distortion and Noise:
@ 2kHz, +20 dBu output level, no load.
Main Output, compressor bypassed: Better than 0.001%
Main Output, compressor engaged: Better than 0.06%

Noise: Mostly 2nd Harmonic
Measured at Main Output, un-weighted, 22Hz-22kHz, terminated 40 Ohms.
With Gain at Unity, Compressor disengaged Better than -103 dBu
With Gain at Unity, Compressor engaged: Better than -92 dBu
Frequency Response:
Main Output, Unity Gain @ 18 Hz -3 dB
@ 150 kHz -3 dB

5015 Power:
Voltage Range 9 to 18 Volts DC, 12 Watts
Connector: 5.5mm X 2.5mm DC jack, Center Positive
Current consumption: @ 9VDC = 1.0 A typical
@12VDC = 730 mA typical
@15VDC= 570 mA typical
@18VDC= 480 mA typical

Mic Preamplifier



Compressor