# EQ REVERB UNIT OWNERS MANUAL 2017

# ARGIRIADIS ANALOGUE ELECTRONICS

# 1st FLOOR, COLLAGE ARTSPACE 2, LONDON, N22 6UJ

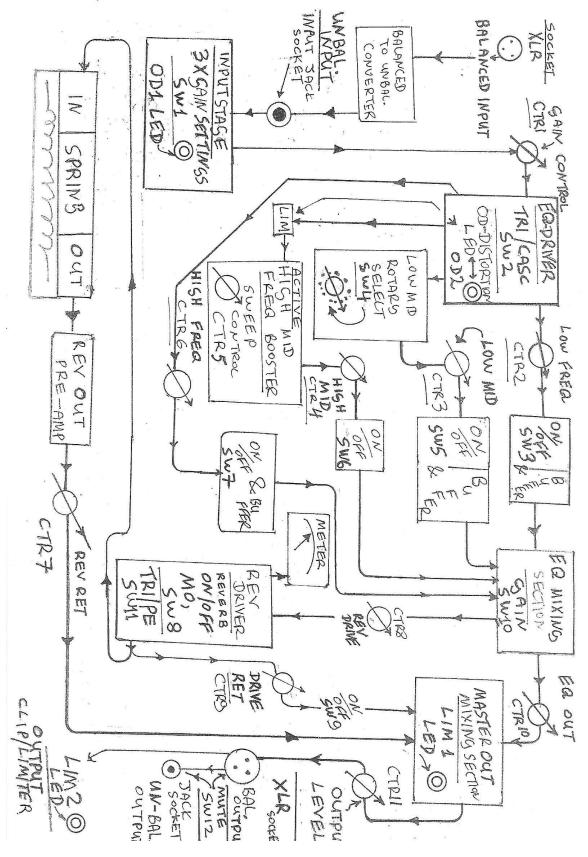
UK. TEL: +44 203 146 8779

email: theo@tube-electronics.co.uk

# Contents

INTRODUCTION	5
OBJECTIVES	5
OPTIONS	5
EXTRA FEATURES	6
DETAILS & HINTS	6
INPUT STAGE & EQ DRIVER	6
EQUALIZATION	7
REVERB DRIVER, REVERB RETURN & OUTPUT SECTION.	9
REVERB DRIVER	9
DIFFERENCES BETWEEN TRIODE AND PENTODE	10
THE OUTPUT TRANSFORMER	10
FREQUENCY RESPONSE	10
THE REVERB TANK AND THE SPRINGS	11
SPRING OVERDRIVE	12
SPRING OVERDRIVE THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY	
	12
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY	12
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY REVERB DRIVE RETURN	12 13 13
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY REVERB DRIVE RETURN REVERB RETURN	12 
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY REVERB DRIVE RETURN REVERB RETURN THE OUTPUT SECTION	12 13 13 13 13 17
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY REVERB DRIVE RETURN REVERB RETURN THE OUTPUT SECTION FRONT PANEL CONTROLS & SWITCHES	12 
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY REVERB DRIVE RETURN REVERB RETURN THE OUTPUT SECTION FRONT PANEL CONTROLS & SWITCHES PREPARATION BEFORE SWITCHING ON	12 
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY REVERB DRIVE RETURN REVERB RETURN THE OUTPUT SECTION FRONT PANEL CONTROLS & SWITCHES PREPARATION BEFORE SWITCHING ON PRECAUTIONS	
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY REVERB DRIVE RETURN REVERB RETURN THE OUTPUT SECTION FRONT PANEL CONTROLS & SWITCHES PREPARATION BEFORE SWITCHING ON PRECAUTIONS SWITCHING ON	
THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY REVERB DRIVE RETURN REVERB RETURN THE OUTPUT SECTION FRONT PANEL CONTROLS & SWITCHES PREPARATION BEFORE SWITCHING ON PRECAUTIONS SWITCHING ON OPERATION	

SAFETY	
TUBE ISSUES	
FUSES	
ABREVIATIONS	



Page **4** of **38** 

# INTRODUCTION

# OBJECTIVES

1) This dual mono unit processes the audio signal in a such a manner as to investigate, exploit and achieve the best possible way the input transducer of a reverb spring can be driven. In the audio world, the term "best possible" is highly subjective and diverse so to be able to satisfy as many choices of sound as possible I offer a lot of options, and that necessitates a high number of controls and switches. This is an analogue project, and the user must investigate these options through practice, experimentation and continuous use to utilize the full potential of the unit.

2) This project is not just about driving a spring to produce reverb. It can also be used as an analogue tube sound processor so:

a) Part of its output is the sound coming out of the **equaliser section** (which is driven by the two input stages), **before it enters the output mixing stage.** 

b) Similarly, the processed medium power drive signal that is developed across the spring input transducer also enters the output mixing stage before it is propagated through the springs.

c) The **reverberated signal** that comes out of the output transducer of the spring is fed to a high gain tube pre- amp and then it **enters the output mixing stage**.

Therefore: The output of the unit contains a mix of these three signals (a, b &c mentioned above), each one of them can then be adjusted by their own level controls and the final output signal amplitude is set by a master output level control.

This **output is limited** after it has reached a maximum amplitude through a **hard clipping circuit** for safety.

# OPTIONS

a) To be able to distort the signal anywhere along its path from the input to the medium power output stage that drives the spring. This distortion is adjustable from mild to harsh and vacuum tubes are used for voltage amplification all throughout the signal chain.

To achieve this, the amount of drive in each individual stage (along the audio path all the way to the spring input) can be adjusted through signal level controls to ensure that each stage (if so desired) can contribute to the overall sound coloration/distortion.

b) This harmonic distortion can be combined with adjustment and boosting of certain frequencies through the various **EQ sections**, and this processing can also be effectively used to shape and animate the sound qualities captured in digital recordings, and make them sound more "natural"

The **HIGH and LOW MID EQ** subsections produce their own distortion and operate in a range of frequencies where the **spring becomes very responsive**.

The EQ section is not a standard type and the four EQ controls (LOW FREQ, LOW MID, HIGH MID and HIGH FREQ) are more like boost level controls of each frequency range and each EQ subsection can be removed from the entire system through a switch situated under its level control.

c) The drive output stage can operate in either triode (TRI) or pentode (PE) mode.

d) The **spring input transducer** and the **spring itself**, can be **driven hard and** this can produce overdrive **spring distortion**.

# EXTRA FEATURES

1) Indicator LED's monitor the overload condition of the various stages, plus the output limiting action and a VU meter gives an idea on how hard the spring is being driven.

2) The inputs and outputs are either balanced (XLR type) or unbalanced (JACK type). If the JACK input is used, the XLR output is automatically disconnected. The XLR output is not disabled by inserting a JACK plug into the JACK output but **this must be avoided**. *This is because if the JACK output is connected into a low impedance input the signal coming out of the XLR output will lose its balance symmetry*.

Finally,

\* The signal level coming out of the XLR (balanced) output is twice in amplitude than the signal coming out of JACK (unbalanced) output.

Please ensure that none of the signal carrying XLR output terminals is shorted to ground by an incorrectly wired XLR to JACK lead.

# DETAILS & HINTS

# INPUT STAGE & EQ DRIVER

This section operates on a high enough voltage (400V) to maintain a good dynamic range. Both of the triode amplifiers, INPUT STAGE and EQ DRIVER (when set on triode mode through SW2) are designed to be as linear as possible if a clean sound is chosen through correct gain control/switch adjustments. These two stages are also designed to clip very softly and as the signal increases the distortion does not start abruptly, but it builds up gradually.

The input stage can accept a wide range of signals, and its sensitivity and gain is set through switch **SW1**. **This switch has no effect when the** next stage is set on **Cascode mode** through switch **SW2**.

SW1 offers three gain settings and on the highest **(H)** the 12AY7 input tube works without any **N**egative Feed Back (**NFB**). This triode tube then produces even harmonic distortion (predominantly second) by an amount proportional to the amplitude of the input signal. This generates sound coloration that manifests itself as "warmth".

For a more transparent sound SW1 can be switched onto lower gain/input sensitivity settings where local NFB (a distortion reduction mechanism) is applied and the input stage produces very little (if any) sound coloration and can accept higher signal levels without noticeable distortion.

The output of the input stage is fed to the GAIN control **CTR1** which adjusts the amount of signal entering the EQ DRIVER stage. This is the **gain control of the input section of the unit** before **the EQ** stages and it offers a finer control range when used in conjunction with the **SW1** gain switch. With careful use of both CTR1 and SW1 any kind of sound is possible ranging from very clean to subtly colored or highly distorted.

To avoid overloading, especially if SW1 is set on high (H), CTR1 should be kept low, on 5 or even less (it all depends on the input signal level).

This control is a logarithmic type, that offers finer adjustment for settings lower than 7.

Switch **SW2** selects the mode of operation to either **Triode or Cascode**, in triode mode the points made about second and even harmonic distortion apply here too. However, some of the even harmonic distortion generated in the INPUT stage will be cancelled by the EQ DRIVER stage. The 12AU7 EQ DRIVER tube can accept higher signal levels than the INPUT stage 12AY7 tube and can drive the EQ circuits better due to its lower output impedance.

However if CTR1 is turned up higher, and SW1 is on High (H), the INPUT STAGE will begin overloading the EQ DRIVER stage and more distortion will result including **odd harmonic type of distortion** due to the **grid current/clipping effects plus the output soft clipping** of the 12AU7 EQ driver circuit. Indicator LED **OD1** will light up when this slow process of soft clipping starts taking place.

#### CASCODE mode:

By pressing the CASC switch **SW2** down the EQ DRIVER stage changes from a triode to a cascode stage This is a low noise Pentode made out of two triodes (=CasCaded Triode). The input **gain switch SW1 has no effect in CASC mode,** because the gain of the INPUT stage is now fixed, but its output (which is the input of the EQ driver) is still controlled by CTR1. In this mode the EQ DRIVER produces **more distortion (and gain)** and that happens at even lower gain control (CTR1) settings. The signal contains both **even** and **odd harmonic distortion products,** producing **a harsher and "dirtier " sound.** 

# EQUALIZATION

The signal is tone shaped before it is sent to the input of the spring. The four frequency bands are of the boost type and the amount of boost is adjusted by their level controls : LOW FREQ, LOW MID, HIGH MID and HIGH FREQ and the outputs of these controls are fed to the internal EQ MIXER. If these controls are set too high their combined signal level could overload this mixer and make it distort too. This mixer is a tube based design using half of the PCL86 (14GW8) tube, it is designed to clip softly, and it produces its own sound.

### You can avoid overloading it:

a) By setting EQ MIXER gain **switch SW10** in the **up position** to keep the **gain** of this stage **low**. When this switch is on **H** (the down position), the **gain** of this stage increases by approximately **+9dB**.

b) If one or two frequency band(s) dominate(s) the spectrum its/their corresponding level control(s) may have to be turned down, sometimes all four may be turned down by an equal amount.

If EQ MIXER **distortion** is **desired**, it is better to set the CTR2/3/4/6 controls as high as possible first, for lower noise levels, and then set SW10 on H if it is still necessary. If both controls and SW10 are set high, the CTR11 **master out control must be kept as low as possible** 

to avoid high level signals on the main output.

In this unit there is also a -9dB attenuation switch at the back near the main output.

The LOW FREQ and HIGH FREQ bands are shelving types, the LOW MID and HIGH MID are peaking types, the LOW FREQ, LOW MID and HIGH FREQ are passive and the HIGH MID is active.

The LOW MID peaking frequencies can be selected discretely through a twelve position rotary selector switch the LOW MID SELECT (SW4) control.

The HIGH MID peaking frequencies can be continuously swept and selected through the FREQ SWEEP (CTR5) control.

The passive EQ filter networks load the EQ driver and play a serious role in the sound:

a) due to this loading effect they slightly increase the distortion,

b) by selectively **emphasizing** certain **harmonics** (which have been generated by the tube amps before and after the EQ) and **reducing** others (harmonics).

The passive LOW MID section contains a small inductor that produces a small amount of core saturation distortion when the signal coming out of the EQ DRIVER is too high.

The active HIGH MID circuit also distorts in its own way especially in the frequencies it rejects that are near its bandwidth. For instance if it is set to peak at say 2kHz there will be a lot of distortion at the frequencies: 1.5kHz..2.5kHz if the EQ driver (INPUT GAIN CONTROL) is turned up high. The HIGH MID has a softclip limiter at its input that contributes to this distortion when the signal coming out of the EQ DRIVER has reached a certain level, the LED OD2 will then light up, indicating that the HIGH MID peak level will not increase any further even if the EQ DRIVER input level is increased through the input GAIN control CTRL1.

When the above happens, for more HIGH MID: a) Turn up the HIGH MID level control.

b) The other three EQ controls may need to be slightly reduced so that the HIGH MID is higher in relation to the other frequencies.

Normally a combination of a) and b) may be necessary.

The HIGH MID is an **active variable booster (VFB)** filter based on the "Wien bridge oscillator" circuit a hybrid design that uses high voltage transistors and the 12AX7 tube. Both negative (NFB) and positive (PFB) are applied here and the circuit "swings" between these two conditions, as if it is trying to oscillate at the chosen frequency but never actually gets there. As a result the peak is very steep (high Q).

The higher **LOW MID** (rotary switch SW4) selected frequencies **overlap with the lower HIGH MID** (CTR5 SWEEP control) frequencies to offer ample choice and boost if required. This is because at frequencies around **1kHz** the springs become very responsive.

Each frequency band control has an **ON/OFF switch** underneath it that disconnect this entire band (it "kills" it when it is not on ON but in the **up** position) a feature that helps when dealing with the frequency response of the spring.

Also, each band can be removed for comparison, something that helps adjustment... for instance you can momentarily remove the two 'mids' (low and high) whilst the low and high band are on to hear the effect and vice versa.

# REVERB DRIVER, REVERB RETURN & OUTPUT SECTION.

After the EQ mixing section the signal is split in two parts :

a) It enters the OUTPUT SECTION through the EQ OUT control CTR10) which adjusts its level.

b) It enters the medium power amp that drives the spring through the CTR8 reverb drive control which adjusts the input signal level this amp receives.

The output of this amp is connected to the INPUT of the **reverb tank** and then to the **input transducer** which is an electromagnet. This is **mechanically linked to the springs** and converts this electrical signal into mechanical vibrations which create sound waves that propagate along the springs. The input transducer can be compared to a loudspeaker, and the **output transducer** which is connected to the OUTPUT of the **reverb tank** can be compared to a microphone that picks up the signal waves of the vibrating springs.

### **REVERB DRIVER**

This is a single ended stage (SE) using the pentode section of the PCL86/14GW8 tube, which can **operate as a triode or pentode** and switch **SW11 selects between these two modes.** 

# IT IS IMPORTANT FOR THE LONGEVITY AND SAFETY OF THE OUTPUT TUBE AND TRANSFORMER TO GO ONTO STAND-BY MODE BEFORE OPERATING THE TRI/PE SWITCH SW11.

That means:

a) Disconnect the high voltage (B+) from the unit by setting the S-BY switch in the UP position (labelled as S-BY on the front plate).

b) Operate SW11 to choose between triode or pentode

c) Reconnect the high voltage by setting the S-BY switch ON (down position).

During this procedure the **MUTE switch SW12** can be used to **silence the noises that these switches (SW11&S-BY) make.** 

# DIFFERENCES BETWEEN TRIODE AND PENTODE

As I already mentioned **triodes** produce predominantly second harmonic distortion which subjectively speaking is more musical and the amount of this distortion is proportional to the amplitude of the signal. The SE output transformer drives the input transducer. The transducer itself plus the springs interact less with the power tube in **triode** mode, because triodes have a lower output impedance.

If a SE triode amp is connected to a proper loudspeaker the low frequency (LF) response is always better than in pentode mode (*unless a pentode output amp uses NFB to improve its LF performance*) but the very low inductance of the input transducer substantially shunts signals below 100Hz both in triode and pentode mode.

However, due to the triodes lower output impedance both reverb drive and reverb return signals sound "fuller" in the low mids (say 200...500Hz) and less "aggressive" and trebly in the high mids, (2kHz...3kHz) in triode mode.

In **pentode** mode, the distortion is higher, starts at lower CTR8 reverb drive control settings and the signal contains both even and odd harmonic products. The **gain is higher in pentode** mode but so is its **output impedance**. As a result the inductances of the **output transformer and spring input transducer affect both distortion and frequency response a lot more**.

Also there is more interaction between the tube, the output tube/transformer and the vibrating springs.

The distortion is higher in the low frequencies, and the gain is higher in the high mid and high frequencies, making the reverb drive return signal a treble booster.

All these effects result in a harsh and "edgy" sound.

# THE OUTPUT TRANSFORMER

When the output transformer is driven hard, it will produce **core saturation distortion** regardless on whether the output tube is working in triode or pentode mode.

However there is more transformer coloration (low level signals) and distortion in (high level signals) in pentode mode because **pentodes higher gain** forces the output transformer to enter its **saturation** region **quicker** 

### FREQUENCY RESPONSE

The tubes **output impedance** forms a high pass filter with the low inductance of the spring transducer which is much lower than the primary inductance of the output transformer and thus it shunts it as already mentioned.

The EQ system has been designed so that low and low mid frequencies can be boosted whilst high mid and high frequencies can be attenuated before the reverb driver. This way the low frequency performance of the reverberated signal can be enhanced if so desired.

# THE REVERB TANK AND THE SPRINGS

The reverb return signal that comes out of the tank is proportional to the reverb drive signal applied to the input transducer (inside the tank) but only up to a certain level. This is because this transducer being an electromagnet reaches magnetic saturation after this level has been reached. In fact the return level will be higher at slightly lower levels than the transducers saturation point, something that can be experienced by trying different settings of the reverb drive control CTR8.

As you turn it up from zero, the reverb return signal increases proportionally until the point of saturation; just before that point the springs have gained enough momentum to produce the highest reverb return signal. At drive levels beyond this saturation point the oscillation decays, but transducer distortion and spring vibration noise is added to the reverb return sound as an **extra interesting effect. This sound is noticeable after practice.** 

However, if a maximum clean return signal is critical then turn up the drive control CTR8 until the saturation and spring noise is heard and then reduce the drive (just a bit) by turning CTR8 slightly anticlockwise to a lower setting.

The audio signal is propagated through the springs in a **random manner**, and for any steady signal level applied at the input of the spring tank the received signal level at the spring **output can fluctuate considerably**.

Often it can reach a certain **predictable value**, especially if the excitation of the input transducer is done in **bursts** by

a) **quickly varying the REV DRIVE** control CTR8 whilst the **SW8** switch is in the **ON** position.

b) by setting CTR8 at a certain level and flicking SW8 from its center to its ON(MO) position ie upwards.

The reverb return signal will not remain at that level for long though, and as soon as the drive burst is removed it will decay in an unpredictable way.

When however the input signal entering the spring tank is **steady and continuous**, the return output level will not necessarily remain constant, but it can **change with time by a significant amount** sometimes as much as 100% or more.

For this reason it is **nearly impossible** to obtain a **return output level of equal strength on both channels in a true STEREO or DUAL MONO** unit when it is being fed by a **constant amplitude signal** like for instance the output of an **electronic organ, even if both (left and right) channels have identical control settings.** 

This **random** amplitude difference between the two channels is a **feature** of a true **stereo unit**, and should be exploited as an **effect**. I have come across so called "stereo" reverb units which are available in the market which are not really stereo because they use only one spring for both channels since this is the only way to achieve equal reverb return level on both outputs. In these units a mix of both channels is applied to the spring tank input and the signal that is received at the spring output transducer is then pre-amplified and sent to both left and right channels through separate level controls.

In this unit an equal output reverb return level on left and right channels is also possible by operating the return section in mono. There is a switch at the rear panel that connects the two reverb return pre-amp outputs in order to obtain a combined left and right mono output that is equal to both channels.

Each channels reverb return control CTR7 can then be used for adjustment and balance purposes, before the two left and right reverb signals are being fed to the two (left and right) master output mixing sections.

Other factors that increase these **random level variations** of the reverb return signal are the **frequency** of the input signal, continuous **changes** of **frequency**, and pulses like a drum **beat** for instance.

Now most audio is a combination of such signals and often these variations are random (and fast) but they tend to average out as equal (say within a duration of a few seconds) on both channels, so the left/right level difference in reverb return only becomes significant with constant amplitude/frequency signals.

**External movement and/ or mechanical vibration** in the vicinity of the unit can modify the **oscillation** of the **springs** and will influence the **reverb return output sound at any frequency**. Therefore **anything that will cause the springs to move will affect the reverb return signal too**. Even if the spring movement is not affected by **external noise and vibration** the output transducer, which acts as a sensitive microphone will **pick it up**.

**Electrical noise** from mains transformers, fans, fluorescent lights etc will also be picked up by the output transducer and then appear at the **output** as noise. For this reason the unit must be placed as far away as possible from such items. This is the reason why the **power supply** unit that contains a mains transformer is in a separate case which **must not be positioned next to the unit**.

Finally the output transducers generate a small amount of hiss.

# SPRING OVERDRIVE

You can overdrive the springs up to a certain extent, but the input transducer can be damaged if the signal that is applied to it is too high, in a similar way a loudspeaker coil can become an open circuit if the music is too loud. The output amplifier is not powerful enough to make this happen and it has not happened yet in other units I have built but please **overdrive the spring with caution, and avoid doing it for long periods of time.** 

Spring overdrive produces a very interesting crunchy, buzzing sound, and it sounds good if it is done in bursts through the MOMENTARY action (MO) of reverb drive switch SW8, or even better by continuously varying the REV DRIVE control from 0... to 10 to create a kind of a reverb build up.

# THE METER & SPRING EFFICIENCY IN RELATION TO FREQUENCY.

On each channel there is a meter that monitors the signal applied to the **input of the reverb tank**. At low frequencies the input transducer loads the output amplifier and this produces distortion when the CTR8 drive control is set high and this loading effect also prevents the signal from reaching high amplitudes. For instance at **100Hz the meter will reach a maximum of -8** and under these conditions the overdrive distortion **is mainly due to the SE output amp**.

The **springs are not very efficient** in propagating the signal to the output transducer at **frequencies below 150Hz**, so even if the drive signal increases through CTR8, the **reverb return** level will **not rise** substantially.

If a higher reverb return signal is required at these low frequencies then :

a) Reduce high mid CTR4 and high freq CTR5 controls by a small amount.

b) Increase the **reverb return** (CTR7) control a bit.

In the mid frequencies around **1kHz** the springs **transmit a lot of sound energy** as they are driven hard into vibration and the meter will reach its maximum at **+5**.

### REVERB DRIVE RETURN

The sound signal across the spring input transducer is picked up and then through the **drive return control** (CTR9) it enters the **output mixing section**.

This signal contains **no reverberation** whatsoever, it is the **medium power out signal** coming out of the PCL86/14GW8 amp whose input is the EQ OUT signal and its level is controlled by the CTR8 DRIVE control.

The reverb drive return signal has been affected by all the issues discussed in the previous paragraphs :

Power amp distortion (triode or pentode), output transformer saturation distortion, transformer and input transducer effects on frequency and amplitude and spring vibration interaction with the input transducer and power amp.

The output stage also generates **harsh grid current distortion** especially in triode mode, and as already mentioned in pentode mode it **boosts the high mid and high frequency signals**.

The reverb drive return sound is thus **sharp and "edgy"**, very different than the EQ OUT signal with its soft and gradual distortion.

# REVERB RETURN

A high sensitivity 12AX7/ECC83 tube pre-amp picks up the reverberated signal developed across **output transducer** and feeds it to the **output mixing section**. The level of the reverberated signal is controlled by REV RET CTR7.

# THE OUTPUT SECTION

This is a discrete solid state circuit that works on a medium power supply voltage (+70V) to retain the high dynamic range of the unit but also ensure that the output will never be high enough to damage any equipment that the unit is connected to.

The input of this section is a **mixer** that combines the three signals:

a) The signal coming out of the equaliser(EQ OUT) before it is fed to the reverb driver and its level is adjusted by EQ OUT CTR10.

b) The reverb drive signal adjusted by DRIVE RET CTR9

c) The reverberated signal adjusted by REV RET CTR7.

The output of this section is a line level amplifier that can drive the equipment that the unit is connected to. **The minimum impedance of this equipment must be 10K.** 

This amp has a soft clip limiter similar to a tube amp, and when the signal reaches the level to activate this **limiter LED LIM 1 will light up**.

The output of this amplifier is fed to the **OUTPUT LEVEL (CTR11) control that adjusts** the output of the whole unit.

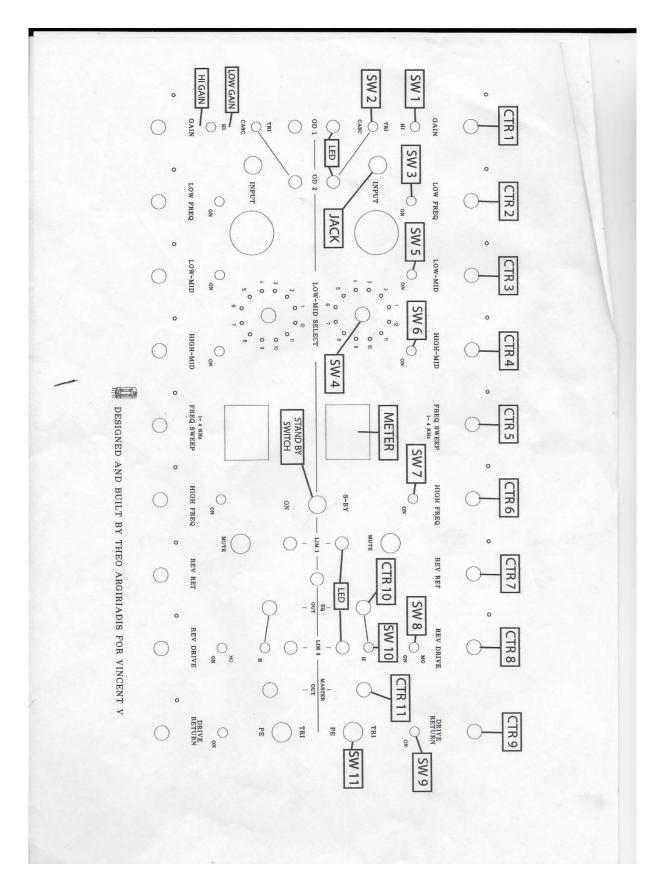
At the output there is a **hard clipping circuit** that **abruptly limits the signals amplitude to a safe level.** When this hard limiting takes place it does not sound good, **LIM 2 LED lights up, and when that occurs the OUTPUT** level control must be turned down.

Finally, there is a momentary **MUTE** switch SW12 that mutes the output of the unit, it is useful to avoid clicking noises when changing from **triode to cascode** (or vice versa, EQ driver ) or **STAND-BY/triode to pentode** (or vice versa, REV driver )

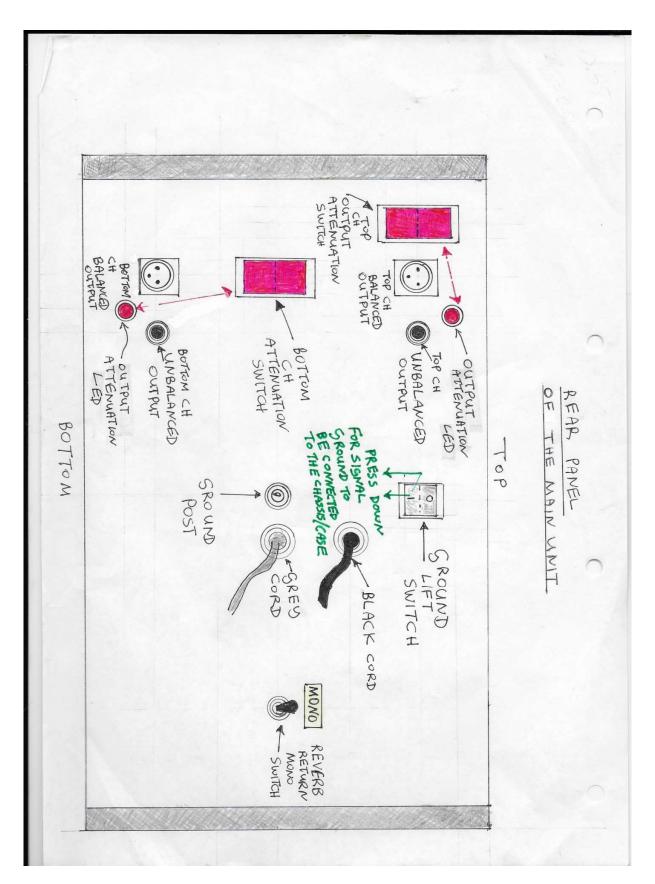
### **Extra Addition**

At the rear of the unit there is an **attenuation switch** per channel that reduces the output by **-9dB**. A red **pilot indicator lights up** (also at the rear of the unit near this switch) when **this attenuation take place**.





Page **15** of **38** 



Page **16** of **38** 

# FRONT PANEL CONTROLS & SWITCHES

\* In the middle of the front-plate (from top to bottom) and next to the meters (on the right of them) there is a **large** toggle switch labelled as **S-BY/ON** (S-BY= Stand- By). When this switch pressed down (**ON** position) it connects the **high** (**B+**) voltage lines from the external power supply unit to the audio circuits of the unit.

\* The pilot LED (between LIM 1 and LIM 2) on the right of this switch and on the same centre line that separates the top and bottom channels will light up **red** when the mains switch in the **power supply unit** is pressed on. This signals that the **low voltage** lines have entered the **main** unit and the tube filaments are lighting up.

As soon as the S-BY switch is in the ON position the pilot LED color will change from red to **amber**. This signals that the unit is ready to operate.

### THE BOTTOM CHANNEL IS THE MIRROR IMAGE OF THE TOP CHANNEL.

#### Abbreviations: Control=CTR, Switch=SW

(1) CTR1=GAIN, it controls the gain of the whole of the input section and specifically the signal level entering the EQ DRIVER.

(2) SW1=(HI), a three-position gain switch of the input stage, can be used as an input sensitivity switch of the whole unit. Up position is the lowest gain, middle position is average gain and bottom position is high, (maximum) gain.

This also applies to the bottom channel even though the 'HI' is labelled on top of the SW1 switch.

(3) SW2=TRI/CASC: this switch selects the second stages (EQ DRIVER) operating mode. In the up position is triode (TRI), in the down position cascode (CASC) mode, in both channels.

The input gain switch **SW1 has no effect in cascode mode** because the first (INPUT stage) gain in CASC mode is fixed. The whole of the **input section gain however is at its highest level in cascode** mode.

(4) CTR2=LOW FREQ, a shelving type control for the low frequencies.

(5) SW3=ON, an on/off switch for the low frequencies. When this switch is pressed down the signal set by the CTR2 control is on, when it is in the up position is off.

(6) CTR3=LOW MID, a peaking type level control for the low mid frequencies.

(7) SW4= LOW MID SELECT, a twelve position rotary switch that selects the low mid frequencies.

These frequencies are: 1=150Hz 2=200Hz, 3=250Hz,4=330Hz, 5=400Hz, 6=500Hz, 7=660Hz, 8=750Hz, 9=840Hz, 10=1kHz, 11=1.4kHz and 12=1.9kHz.

### There is no stop between 12 and 1

(8) SW5=ON, an on/off switch for the low mid frequencies. When this switch is pressed down the signal set by the CTR3 control is **on**, when it is in the **up** position is **off**.

(9) CTR4=HIGH MID, a peaking type level control for the high mid frequencies.

(10) SW6=ON, an on/off switch for the high mid frequencies.

When this switch is pressed **down** the signal set by the CTR3 control is **on**, when it is in the **up** position, is **off.** 

(11) CTR5=FREQ SWEEP, frequency sweep control that selects the frequency to be boosted, at 0 this frequency is 1kHz, at 10 is 5.6kHz.

(12) CTR6=HIGH FREQ, a shelving type control for the high frequencies.

(13) SW7= ON, an on/off switch for the high frequencies.

When this switch is pressed **down** the signal set by the CTR6 control is **on**, when it is in the **up** position is **off**.

(14) CTR7= REV RET, it controls the level of the **reverberated** signal that enters the MASTER OUTPUT SECTION.

(15) CTR8= REV DRIVE, controls the level of the signal that drives the spring reverb amplifier (REV DRIVER).

(16) SW8=MO/ON, a three position toggle switch that can switch the reverb drive on/off and also trigger bursts of reverberation when set in the MO (momentary) position.

\* In the **middle** position **no** signal enters the reverb driver.

\* In the **down** position the signal controlled by CTR8 **enters** the reverb driver.

\* the **up** position is momentary and it enables the signal controlled by CTR8 to **enter** the reverb driver. As soon as you press it up, the lever will **return to the middle position on its own, "killing" the signal** entering the reverb driver.

(17) CTR9=DRIVE RETURN, it controls the level of the signal that is sampled across the spring input send transducer **before** it is propagated through the springs.

This is also the reverb drive amplifier output (but **not the reverberated**) signal. CTR9 adjusts the level of this signal that enters the MASTER OUTPUT SECTION.

(18) SW9=ON, on/off switch for the drive return signal.

When this switch is pressed **down** the signal set by the CTR9 control is **on**, when it is in the **up** position is **off.** 

(19) **MUTE:** a red momentary push button switch (non locking type) that silences the output of the channel when pressed.

Whenever you use the TRI/PE (SW11) and the stand-by (S-BY) switches you can **momentarily press and** hold the MUTE switch to avoid loud pops and clicking noises at the output of the unit.

(20) LIM 1: when this LED (one on each channel) is lit it indicates that the output stage has reached overload mode, this stage is designed to clip softly. This clipping happens before the signal is applied to the CTR11 MASTER OUT control.

It is normal for this LED to light up occasionally, especially if the EQ gain SW10 switch is (pressed down) set in H.

(21) CTR10=EQ OUT, this control sets the amount of signal coming out of the whole EQ before it enters the MASTER OUT section.

(22) SW10=EQ OUT, an extra gain switch for the EQ MIXING SECTION it introduces approx. +9dB of gain when pressed down in H.

(23) LIM 2: when this LED (one on each channel) is lit it indicates that the output signal of the whole channel (after the CTR11 MASTER OUT control) has reached the maximum permissible level. The output then is limited and the clipping is very hard.

(24) CTR11= MASTER OUT, it controls the output level of the whole channel.

(25) SW11= TRI/PE, a large toggle switch to select the reverb output amplifier operating mode.

# PREPARATION BEFORE SWITCHING ON

- Remove the protective plate which is screwed in front of the front plate. The 10mm wrench for the nuts is supplied but you will also need a flat screwdriver.
- Carefully unscrew the top cover, make sure that all of the tubes are firmly placed in their sockets. The best way to do that is by holding them **softly** from the **top tip** and giving them a **very gentle** push downwards to check that the pins are inserted all the way into the sockets. Also, if you **softly** hold the tubes from the **top tip** and **very gently** move them from side to side, a slight play (approx. 1/8"maximum in the tip area) is normal.

### (1) PLACEMENT

The location of the EQ REVERB unit is important.

• Its location/mounting **must allow plenty of air circulation** from below, the sides and above it to disperse the heat that the tubes/semiconductors generate.

Avoid hot locations such as near radiators or other heating units

### IF THE UNIT IS MOUNTED ON A RACK:

(a) Ensure that the unit below it does not get hot.

(b) Even if the unit below the EQ REVERB runs cool, there must still be a distance of at least a 2U (3(1/2) inches, 90mm) of empty air space between the EQ REVERB and the unit below. If the unit below get warm this gap must be increased to at least 4U (7 inches, 180mm).

(c) Leave a 4U distance of empty air space between the EQ REVERB and the unit that is on top of it.

### (d) both sides of the whole rack must be open to allow air passage.

(e) Also, if the unit is mounted on a **rack** it will need **extra support** from below due to its **weight**. A sturdy **rack shelf is** suitable the 1" feet will allow air to enter from below.

(f) Keep the top clear of items such as papers or anything that could block air passage and cause overheating.

### PLACE THE UNIT AWAY FROM SOURCES OF INTERFERENCE

The output transducers inside the tanks are susceptible to electrostatic and magnetic interference.

The springs are susceptible to **mechanical and magnetic interference**. The tanks are mounted along the side of the unit, **so it must be as far away as possible from:** 

(a) AC power lines, fluorescent lights, fans etc.

(b) The power supply cords that are plugged to the power supply must always run from behind the unit (never along its sides) to the power supply.

(c) If the room where the unit is operated is electrically and/or magnetically noisy, you may have to slightly move it around and find the most suitable location in order to completely eliminate any noise that the springs may pick up.

#### (2) GROUND LIFT SWITCH

If this switch is used **incorrectly** the unit will become susceptible to **interference**, **noise and hum**. Ensure that this switch which is situated in the top middle of the rear panel is in the **on** position which the one with the **(I) pressed down**. This is to ensure that the metal case and chassis are connected to the negative signal ground of the unit's electronic circuits so that they act as a **shield against external interference** and **prevent internal instability and oscillation**.

In most studios (especially if the balanced inputs and outputs are used) this ground switch may never be used. For this reason, there is a piece of **tape** on it **to keep it in the ON position at all times.** If you need to operate this switch (see **notes on ground lifting & loops**) just pull this tape off.

It is important to remember that unless the chassis and case are connected to the signal ground by other means (in order to act as shields) if the GROUND LIFT SWITCH is in the off position the unit will be noisy.

A high **pitched noise** if the **REV RET** control is set high is a common symptom when the ground switch is used incorrectly.

#### (3) STAND-BY SWITCH

Ensure that the front panels stand-by switch **(S-BY)** is off, ie the up position. This is because the high voltage must not be applied to the tubes and the circuits until the tubes warm up.

#### (4) FIRST TIME SWITCHING ON

If you switch on for the first time and until you get used to the unit:

\* Turn all controls and **especially the OUTPUT** CTR11 control fully anticlockwise, set input gain switch SW1 for **average gain (middle position)**, the TRI/CASC switch SW2 in triode (TRI,**up**) and the EQ mixing gain switch SW10 in low(**up**).

- \* Set all EQ ON-OFF switches SW3, SW5, SW6 and SW7 off (up).
- \* Set the triode/pentode switch SW11 on triode (TRI, **up**).

\* Set the REV DRIVE switch SW8 in the **middle** position.

\* Connect an audio signal to the input of the first channel to be tested. The input can accept a wide range of **line level** signal amplitudes, and offers ample gain adjustments.

However, for the following first tests I would recommend to use a source with an average output of approximately 0.775Vrms/ 0dBu.

(a) The input of the unit can also accept old type domestic devices with an average line output level of **0.3162VRMS/-7.8dBu**. In this case the input gain settings (SW1,CTR1) must be higher than the ones I recommend in the following paragraphs. However, If the signal at the input is below the **-7.8dBu** level, the unit may not to be able to distort as much.

(b) For input signal levels higher than the **0dBu** level the input gain (SW1,CTR1) may need to be set lower for a clean sound.

\* Connect the output of the unit to the LINE input of a mixer, a monitor amplifier or any device that accepts studio line level signals. The input impedance of this device must not be less than 10K. A simple monitor amplifier is the best choice for the first introductory tests.

### (5) THE OUTPUT ATTENUATION SWITCHES

These are **red** rocker switches located on the rear panel, one per channel that **reduce the output signal** (of this channel) by **9dBs.** They are right **next** to the **XLR** output of each channel. If you are switching on for the first time, it is **well advised to introduce** this **extra output** attenuation by pressing down these switches. There are **two LEDs** also **next to the XLR outputs** which will **light up red when this attenuation is introduced**.

The price to pay when this attenuation is applied is that for some **clean low distortion settings the output volume may be low** but it is a price worth paying until you get to know how to control the output level of the unit and how to reduce this output when it is necessary. This will be the case for instance when you are overdriving the input stages or the spring to create serious distortion.

The setting of these switches also depends on the sensitivity of the equipment that you drive and you may end up setting them in the low or high output position permanently once you get to know the unit.

### (6) THE POWER SUPPLY UNIT

Before you connect the power supply to the mains, connect the two power supply cords to the power supply sockets. These are the two cables, one grey and one black, that come out from the back of the main unit. Please treat the black cord with extra care, it carries the three high voltage lines: 220V, 270V and 400V. Both of the plug connectors at the end of these cables and their respective sockets on the power supply front panel are mated in only one possible orientation.

The smaller round **high voltage plug** at the end of the **black cord** features **keyway polarisation**: it has a notch to make possible to **fit in one way only**. The bigger **plug** on the **grey cord** and its **socket** (on the power supply) are also **non-reversible**, they have a flat side in the middle to make them **mate one way only**.

Please examine both sockets and plugs before connecting to recognise their mating features and do not use any force when inserting the plugs into their sockets.

The **power supply is also a source of electrostatic and magnetic interference** and it must be placed as far away from the main unit as possible, **preferably on the floor.** 

**Please allow plenty of ventilation for the power supply unit too.** Avoid hot locations such as near radiators, heating units, hot central heating pipes etc.

### PRECAUTIONS

### PLEASE DO NOT LEAVE THE UNIT ON WHEN NOT IN USE

\* Each time you increase the input gain settings, equalizer output, reverb drive and return levels ensure that you decrease the MASTER OUT (CTR11) control so that the output of the unit remains as constant as possible.

The unit is capable of producing very high output signals. Even though its output is limited, a high enough signal such as a loud buzzing noise caused by a faulty input lead may damage an amp or speakers if the input gain levels as well as the MASTER OUT control CTR11 are set too high. It is therefore best to start quietly and gradually turn the levels up so that you can keep control of the units overall output level.

\* If you are switching on for the first time, it is well advised to introduce **extra output attenuation** by pressing **the red attenuator switches on** the **rear panel of the main unit on both channels**.

\*Always ensure that output overload LED LIM 2 is off, as soon as it lights up that means that the maximum output limit has been reached. If the LIM 1 LED is lit it does not matter it is an indication that the output stage has reached soft clipping point.

\*Remember, the output level on the XLR balanced output is twice as high as the unbalanced jack output.

\*Do not use the XLR and JACK outputs simultaneously

\* Always set the S-BY/ON switch on S-BY (up) before you use the TRI/PE SW11 switch and after you have used SW11 press S-BY/ON down again (ON).

### SWITCHING ON

- \* Connect the power supply to the mains.
- \* Switch on the power supply
  - (a) the red LED on the front panel of the power supply will now light up
  - (b) the PILOT LED on the front panel of the main unit will light up (red) too.

\* After approximately 40 seconds press the S-BY down (ON) to operate. The PILOT LED on the main unit will change from red to amber.

### OPERATION

### EQ SECTION

Test one channel at a time, for instance go through (1)...(9) tests below on the top channel first and then do the same tests on the bottom one.

These are just guidelines for the first tests and familiarity through experimentation is necessary in order to develop your personal gain settings for the kind of sound that you want to produce.

# Due to the complexity of the unit, when doing the initial sound tests, it is better to become familiar with the input stages and the equaliser first, and then move on to the reverb section.

### TESTING

(1) Set the CTR10 EQ OUT control to around 12 o'clock (middle position) and the CTR1 input GAIN control to 10 but keep SW1 gain switch in the middle position for an average input stage gain.

LED OD2 will light up if the signal level is high enough to trigger the limiter in the HIGH MID EQ booster.

(2) To understand the workings of the EQ test the **low frequency and the high frequency bands first**. So keep SW5 and SW6 in the off position (up) for the time being.

Press SW3 (LOW FREQ ON/OFF) and SW7 (HIGH FREQ ON/OFF) switches **down i.e. ON**, set LOW FREQ (CTR2) and HIGH FREQ (CTR6) controls to **5**.

(3) (a) By gradually turning up the MASTER OUT control CTR11 you should be able to hear the audio signal from the speakers. If the input signal happens to be higher than 0dBu (0.775Vrms), the EQ DRIVER will start producing mild distortion and LED OD1 will then light up. If this distortion is not desirable turn down CTR1 GAIN control a bit and slightly increase CTR11 to compensate for the loss of volume.

(b) Kill (remove) the LOW FREQ band to isolate the HIGH FREQ band through SW3, and vice versa through SW7, and try different settings of CTR2 (LOW FREQ) and CTR6 (HIGH FREQ) controls.

(c) Turn the CTR10 (EQ OUT) down to zero first and then increase the EQ gain by setting the SW10 gain switch in (H). Now turn up CTR10 again, to hear the EQ MIXING SECTION gain boost (approx. 9dBs).

If you set the LOW FREQ and/or HIGH FREQ **EQ controls high** enough (8...10) the **EQ MIXER** stage will generate **distortion** and eventually will start **clipping very softly**.

(d) To set the input stage gain higher:

- \* first turn CTR1 (GAIN) down to zero
- and
- \* set SW10 (EQ OUT GAIN switch) back to low (up)
- \* while keeping both CTR10 (EQ OUT) & CTR11 (MASTER OUT) to 12 o'clock (middle position).
- \* Now set SW1 input gain switch to high (**down** position).

Gradually start turning CTR1 up again while (if necessary) **simultaneously turning** the master out control **CTR11 down** to ensure that the **output** remains approximately at the **same level**. At first, **LED OD2** will light up indicating when the HIGH MID **limiter** is being activated. As you further increase the input gain control CTR1, LED **OD1** will light up indicating the beginning of a **smooth overloading** process in the **EQ DRIVER** stage resulting in **distortion**.

### (e) Cascode operation.

- \* First turn the input GAIN CTR1 down to zero
- \* Switch on to cascode (CASC) through the TRI/CASC switch SW2.

Gradually turn CTR1 up again, while keeping the output constant through CTR11 as in (d), remember the EQ DRIVER produces **extra gain** in **cascode** mode as well as higher amount of harmonic distortion. At **input signal levels of 0dBu (0.775VRMS)**, the distortion is mainly second for CTR1 settings of up to approximately 6 and it is noticeable. Consequently, for mild coloration in this mode, do not set CTR1 too high. At CTR1 settings higher than 6 the EQ driver produces more third and higher odd and even orders, resulting in a harsher sound. Eventually clipping occurs, but it remains soft even at high levels creating a compression effect.

The rest of the equaliser bands can now be tested one at a time LOW MID first and then HIGH MID. Set the LOW and HIGH FREQ (CTR2 and CTR6) controls to 5 again and the input section gain for low distortion at **0dBu (0.775mV RMS) signal at the input:** 

\* in triode (SW2 in TRI) set SW1 for average gain (**middle position**) again and GAIN control (CTR1) to 10.

\* in cascode (SW2 in CASC, where **SW1 settings have no effect)** set the GAIN control (CTR1) to5; at this setting the cascode produces enough second harmonic distortion to introduce "harmonic colour" to the sound.

### (4) The LOW MID

The **SW4 is a twelve position rotary** switch that selects the following frequencies : 1=150Hz, 2=200Hz, 3=250Hz, 4=330Hz, 5=400Hz, 6=500Hz, 7=660Hz, 8=750Hz, 9= 840Hz, 10=1kHz, 11=1.4kHz and 12=1.9kHz.

### There is no stop between 12 and 1.

On each selected frequency the LOW MID EQ section contains an inductor type bandpass tuned circuit. These networks have a **lower Q factor** (than the active circuitry used in the HIGH MID band) in order to create a **"smooth resonant tone"** in both triode and cascode, but in the harmonically rich cascode mode the effect sounds more pronounced. This is because the **resonance of these LOW MID** frequency band networks enhance the sound colouration generated by the EQ DRIVER.

(a) Switch on the LOW MID by pressing SW5 switch down (ON), and select the low mid frequencies one by one and bring them in and out of the the mix by turning the LOW MID (CTR3) control from 0 to 10 and then back to zero again to understand the effect.

(b) Investigate the CTR3 range (from 0 to 10) on each frequency band.

(c) Now isolate the LOW MID by removing the LOW and HIGH FREQ bands through the SW3 and SW7 ON/OFF switches.

(d) You can now test the three bands in isolation, pairs and as a mix:

(i) First isolate each one of the three LOW FREQ, LOW MID & HIGH FREQ bands to get a good "feel" of each EQ subsection by using SW3,SW5&SW7 switches and CTR2,CTR3,SW4& CTR6 controls (note that SW4 is a rotary switch so I call it a control here).

(ii) Then mix two of the bands and finally all three.

(e) Try Increasing the EQ OUT gain (switch SW10 in H), input gain (SW1 in H) and cascode mode (SW2 in CASC) as in (3) (c),(d) and (e).

(6) With the HIGH MID still out of the mix (SW6 off, up position) and all the other bands on the setting for the most possible flat response is: LOW FREQ/CTR2=5, LOW MID/CTR3=5, LOW MID / SW4 rotary switch set on 11, HIGH FREQ / CTR6 = 4

Even at these settings there is a drop at approx. 330Hz which can be changed by switching SW4 on to a different position.

### The HIGH MID

(7) The HIGHMID level control (CRR4) is logarithmic as a result fine signal level adjustment is easier for settings below 7.

This EQ booster has a much steeper response (high Q) and by overdriving it a more extreme and "edgy" kind of sound is generated, with a characteristic HIGH MID distortion.

The **soft limiter** at the input of this frequency booster affects the output level and generates distortion **as** well as the active filter electronics of the booster (see EQUALISATION SECTION page 5).

For a clean **HIGH MID tone** you must set the input gain levels low enough so **LIM 2 LED is just about to light up. EQ gain switch SW10** can be set to high (down) to **increase the EQ MIXER output** if necessary.The output in some frequencies may vary slightly but on average the **boost on the selected frequency can reach +30dBs.** 

The best way to get a "feel" of the HIGH MID SWEEP control CTR5 is to **test it in isolation** first. So switch off all of the other bands through SW3,SW5&SW7 and press **SW6 down** to switch on the **HIGH MID**.

(a) Turn up the HIGH MID CTR4 level control and sweep the high mid frequencies through the CTR5 **FREQ SWEEP** control which selects frequencies from **1kHz** to **5.4kHz**.

Due to the high output that this circuit produces when it resonates, if the HIGH MID (CTR4) control is set high it overdrives the EQ mixer even if the SW10 switch is set in the low (up) position.

\* Repeat procedure outlined in (3)(d) to set the input gain higher while you sweep the HIGH MID frequencies to overdrive the high mid booster.

\*Set the SW2 switch in CASC as in (3)(d) to blend the distortions that the EQ driver and the HIGH MID booster generate.

(8) You can now start mixing all of the frequency band sections:

(a) Mix the LOW FREQ & HIGH FREQ bands and keep the LOW & HIGH MID off at first; then bring both mid bands in, so that all the bands are combined.

(b) Now remove the HIGH&LOW FREQ bands in order to isolate on the two midrange bands.

(c) An interesting but **sharp tone** occurs when **only** the HIGH MID and HIGH FREQ bands are switched on and the FREQ SWEEP control CTR5 is continuously varied from 0 to 10.

\* Remember that the two **mids overlap in the 1...2kHz range**, the frequencies where the springs become very efficient and responsive.

(9) Try different input gain settings, change between triode and cascode (TRI/CASC switch SW2) and different frequency level control settings but always ensure the output level is never high enough to light up LIM2.

Now you can repeat (1)...(9) on the second channel.

# THE REVERB SECTION

### PRECAUTIONS

(A) Please ensure that you apply all of the precautions outlined in in the EQ SECTION tests regarding maximum output level, and ensure that LIM 2 LED is always off.

This is because in order to generate overdrive distortion from the unit you must increase:

- \* the input gain
- \*and/or the four frequency band levels (especially the HIGH MID)
- \* and/or the EQ mixer gain
- \*and/or the reverb drive

Also, overdive distortion is produced when:

- \* the EQ driver stage operates in cascode mode
- \* the reverb driver stage operates in pentode mode
- \* the reverb drive return is brought onto the output mix through CTR9 control

All of these gain increases/ modes of operation result in high signal levels being generated inside the unit.

(B) PLEASE SET THE S-BY SWITCH ON STAND FIRST SO THAT NO HIGH VOLTAGE IS APPLIED TO THE REVERB DRIVER OUTPUT TUBE DURING THE TIME YOU OPERATE THE TRI/PE SWITCH SW11

(a) Disconnect the high voltage (B+) from the unit by setting the S-BY switch in the UP position (labelled as S-BY on the front plate).

(b) Operate SW11 to choose between triode or pentode

(c) Reconnect the high voltage by setting the S-BY switch ON (down position).

(d) You can use the momentary MUTE switch if you wish to silence the clicking sounds generated.

(e) Note that the output level will always be higher in pentode mode so always turn the MASTER OUT control down a bit when you switch from triode (TRI) to pentode (PE).

\*It is advisable to do the first reverb tests in TRIODE (TRI) mode and then in PENTODE (PE),

\*Always check the output level and ensure that output overload LED LIM 2 is off, as soon as it lights up that means that the maximum output limit has been reached.

\* The output must be well below the signal level at which LIM 2 is lit.

\*Remember, the output level on the XLR balanced output is twice as high as the unbalanced jack output.

-----

### (1) The EQ OUT signal

#### CTR10 EQ OUT control can either:

- (a) be turned down to zero so that you can test the reverb section in isolation first and then experiment with different CTR10 and CTR7 (REV RET) settings to get a dry/reverb mix. or
- (b) set to 12 o'clock (middle) for the tests to start with a **dry/reverb** output signal **mix** and then you can **isolate** the **reverb return** signal at any time you wish by turning it (CTR10) down to zero.

### (2) The rest of the controls

There are three ways to test the REVERB section for the first time:

(a) Try the different gain settings and the four frequency bands **separately first** (ie one band/gain setting at a time) and **then mix** everything as outlined step by step in (2)...(9)**TESTING/EQ SECTION**.

OR

(b) Set the EQ controls for a flat response as outlined in (6)/TESTING/ EQ SECTION.

- \* start without the HIGH MID, first
- \* try different gain settings as explained in (3)(a),(c),(d)&(e)/TESTING/EQ SECTION.
- \* repeat everything with the HIGH MID included.
- \* repeat the same tests on each frequency band **separately** and then mix the bands again.

(c) Repeat (8) and (9) from TESTING/ EQ SECTION with the reverb included so that you can try a mix of bands/gain settings that you may have already chosen.

Eventually, it is advisable to do all of the above

### TESTING

#### (3) TRIODE mode, SW11 TRI/PE switch in the up (TRI) position

**Temporarily** set the MASTER OUT CTR11 control to **zero** and then :

\* Set the REV RET (reverb return, CTR7) to 8

\* Press REV DRIVE SW8 switch down (ON)

and

\* set the CTR11 **MASTER OUT** control to around 12 o'clock (middle)

(a) Turn up the CTR8 REV DRIVE control from 0 to 10 to trigger and hear the reverb. At the same time the meter will indicate the level of the drive signal across the input spring transducer.

(b) Now you can repeat (2) ...(9) as outlined in the EQ SECTION/TESTING paragraphs, and mix the EQ and the reverb sections.

### (4) PENTODE mode, SW11 TRI/PE switch in the down (PE) position.

Temporarily set the MASTER OUT CTR11 and the REV DRIVE CTR8 controls to zero.

# Switch off the high voltage from the unit before you operate the TRI/PE switch; see PRECAUTIONS.

- \* Switch in to **pentode** mode
- \* re-set the **MASTER OUT** control to around 12 o'clock (middle)

and

\* repeat (3) (a)&(b) above.

### (5) The REV DRIVE control CTR8

The higher the CTR8 control is set the stronger the signal propagating through the springs will be. As CTR8 is turned up **higher**, reverb drive **amplifier and spring distortion will occur**.

(a) For a clean reverb set the CTR8 low, how low it depends on:

\*the signal level that is coming out of the EQ MIXER

and

\*whether you are in triode or pentode mode.

Reduction of the reverb drive in order to get a cleaner and more transparent reverberated tone will result in a lower signal level propagating through the springs. This will also result in a lower signal level received at the output transducer inside the tank.

Therefore to compensate for **reverb volume loss** you may have to set the reverb return (**REV RET**) **CTR7** control **higher**.

- (b) For spring **overdrive** you can:
  - \* Increase the **reverb drive** by turning **CTR8** higher.
  - \* Increase the **input section gain** through the CTR1, SW1 and TRI/CASC SW2 control/switches. and/or
  - \* Increase the **EQ** level control settings and/or
  - \* Increase the **EQ MIXER gain** by setting the EQ gain SW10 switch in the H position (down) and/or
  - \* Operate in **pentode** mode.

The **more drive** you introduce the more you you may have to **reduce** the **return** signal by turning down the reverb return control CTR7 in order to keep the output at the same level.

I recommend **experimenting** with these two controls (**CTR7&CTR8**), using them simultaneously to get a "feel" of the drive and return levels plus the overdrive spring distortion.

### (6) Pentode versus triode on reverb

In **pentode** mode the reverb driver produces a **higher signal level** than in **triode** (mode) in **all frequencies**.

However, the signal level is frequency depended, it gets higher at frequencies above 100Hz, and and it increases even further in the mid and high frequencies.

\* As already mentioned the springs **become very responsive and efficient** (in propagating the signal) **at around** 

1...2 kHz.

\*The reverb driver in **pentode** mode is a **high mid and treble booster** so it is a faster amplifier that **performs better** when transmitting **transient pules through the springs like sharp knocking sounds or beats.** 

\*In pentode mode it is easier to overdrive the springs and get the characteristic 'reverb distortion'.

Therefore, for a certain type of 'tone' a pentode output stage is perhaps a better, more natural driver, and when it comes to the high mid and high frequencies pentode mode it is the best.

To hear the difference, try this in both **triode** and **pentode** mode:

**Isolate the high mid** and the **high** frequencies by switching off the LOW FREQ & LOW MID (SW3&SW5) switches, **do a dry/reverb mix**, and simultaneously use the **CTR5** control to **sweep the HIGH MID** frequencies.

Therefore:

- In Triode mode
- \* The signal that is sent to the springs is **cleaner/** more **transparent**

#### \* The frequency response is more uniform.

\* The driver produces less distortion in the low frequencies.

\* The reverb drive (CTR8) and the reverb return (CTR7) controls must be set higher than in pentode mode.

As I have also already mentioned, **below 150Hz the springs become less efficient in propagating the signal**, so in some low frequency applications triode mode may be preferable.

### (7) The MO/(OFF)/ON switch SW8

To trigger **bursts of reverberation**:

- (a) Set this switch which is located under the reverb drive control in the middle position
- (b) Set the reverb drive control as high as possible, set the REV RET (CTR7) to around 8
- (c) Momentarily press SW8 in the up (MO) position, it will return to the middle position on its own.

(d) Another way of doing this is by turning the REV DRIVE control very fast from 0 to 10 while SW8 is in the ON (down) position.

(e) For an interesting sound effect, **switch off** LOW-FREQ (SW3) and LOW MID (SW5) and repeat (c) or (d) to trigger the reverb with **only the HIGH MID and HIGH FREQ EQ** bands, while **sweeping** the HIGH MID frequencies.

### (8) DRIVE RETURN

This is the 'power drive' signal that is applied to the input transducer inside the spring tank not the reverberated signal

Press SW9 down (ON) and gradually turn up the CTR9 control to bring the drive return signal into the mix.

Try the drive return sound in both triode and pentode modes.

### Switch off the high voltage before you operate the SW11 TRI/PE switch, see PRECAUTIONS.

(9) Experiment with mixing the three signals EQ out, reverb return and reverb drive return.

### STEREO

Test the **second channel on its own** and then **both** channels for **stereo** operation.

When both channels are operating use the **MONO reverb return switch at the back** to compare the reverberated sound in mono with equal reverb return on both channels and **stereo** with its **random reverb return signal level variations** (see notes on REVERB TANK AND THE SPRINGS).

Sometimes if you press the **mono switch at the back**, the reverberated signal **output level may reduced**, especially at frequencies around 1...2kHz when the springs are very efficient. This is because the springs may be **oscillating in out of phase** with one another. This is normal and to increase the output turn up the REV RET (reverb return) CTR8 control.

# **TECH NOTES**

### SAFETY

The **chassis** and **metal case** of every electrical appliance must always be securely and permanently **connected to the mains ground**. This is the **middle pin of the IEC** mains input socket, and the connection between the chassis/metal case and mains ground is the **safety earth bond**. This bond (connection) provides a very low resistance path to earth, so that if a live wire comes into contact with the exposed metal work the resulting live to earth current will **blow the mains fuse and avoid an accident**.

### THE SAFETY EARTH BOND MUST NEVER BE BROKEN TO AVOID DANGER OF ELECTROCUTION

Always use a three conductor IEC mains cord and make sure that it is in perfectly good condition. Also ensure that the mains outlet ground connection (in the room/studio where this cord is plugged) is regularly checked by a qualified electrician.

### GROUND LOOPS

If two or more units are connected through unbalanced audio leads and these units are safety earthed, **ground loops** will be developed **between** the electrical safety grounds (**earth bonds**) and the signal grounds of the **interconnecting cables**.

This is because part of the signal (the signal ground) takes two routes:

### a) Through the audio leads

b) Through the negative power supply lines (since it is connected to them) the metal chassis & case of each unit or its power supply safety ground connection (earth bond) and eventually mains earth wiring.

This could result in **hum that sounds more like buzzing**, especially if some units are powered from different sockets far apart, and at least one unit contains a noisy power supply.

#### GROUND LIFTING

If this problem occurs, it is possible to brake the loop by **disconnecting the units case from the signal ground**.

First turn the units output level control to zero, remove the tape and press the ground lift switch in the opposite direction. Then you can gradually turn up the output level control to test if the hum has gone.

It is very important to turn the output control to zero first, because the are many reasons why hum appears when two units are connected to one another, badly shielded interconnecting cables to name one. If a ground loop is not the reason of the hum **operating the ground lift switch for the wrong reasons could increase the hum/noise to a much higher level.** 

If a ground loop is the reason, then ground lifting can eliminate (or at least reduce) the hum.

If **both units** are **safety earthed** and only **one** has the **signal ground connected to its chassis/case** the other one is still shielded for noise. This is because **both cases are still connected to the signal ground through the mains safety ground wiring.** 

The situation becomes more complicated of course if more than two units are connected together because more than one ground loop may exist, and as I already mentioned there may be other reasons for the hum too.

One way to **eliminate hum** is to use the **balanced inputs and outputs**, because no part of the signal is going through any ground connection outside the unit. That means that the various unit metal cases are not connected through the interconnected cables.

### BALANCED AND UNBALANCED LINES

A **balanced line** is made out of two wires, one wire caries a signal which is out of phase with respect to the signal in the other wire so you need both wires to get the full signal, thus **both wires are live. The ground is not part of the signal and this results in eliminating ground loops, a great advantage.** 

Also, because the input of the equipment that these two wires are going to only accepts out of phase signals, it **rejects** in-**phase signals like noise, hum and interference.** This is because the input of the receiving device is only sensitive to the signal difference between the two wires and not to something that is added to both wires in equal amounts during propagation.

An **unbalanced line** consists of only one live wire electrically referenced to ground and the **ground is also part of the signal.** This can create **ground loops** and since only one wire is carrying the signal (with respect to ground) the input of the receiving device **does not reject noise/interference.** 

Now one thing worth mentioning here is that hum/noise and interference picked up from the surroundings is not an issue at the output because the output impedance of the unit is low and the signal is relatively high. It can only be a problem at the input because the impedance is higher.

# TUBE ISSUES

### NOISE, MICROPHONY & AGE

Tube noise often manifests itself as excessive hiss and intermittent crackling noises a condition that gets worse with time, but this type of noise (unlike microphony) is not caused by vibration. Excessive hiss is often caused by emission loss due to age.

Microphony is a noise that manifests itself as:

(a) the equipment makes noises when it is moved around, sometimes this noise can be loud.(b) Ringing and whistling noises similar to mic/PA audio feedback which may start or stop at random.

A tube is a mechanical device and its various **component parts** are welded to one another and can become **loose** with time. **Ringing** is due to mechanical resonance.

Often in very noisy tubes you get a combination of these noises like for instance a **gradual build up of** rumble which may develop into a **loud crackling** sound which may stop or get worse if the equipment moves or receives a gentle knock.

All tubes generate noises, some do it more than others and **tubes** become **more noisy with age**. Some tubes however are noisy even if they are **new**. New Old Stock (NOS) **tubes** can be **microphonic** even though **perfect in all other aspects**.

Age makes minor noise problems worse, and **thermal cycling** (the equipment is switched on and off) can deteriorate the adhesion of the mechanical elements. However, you must **never leave the equipment on** when not in use because a lot more damage can be done this way and it is not safe when you leave it on for long periods of time and you are not around.

### MICROPHONY TEST

### Please keep the volume low and then turn it up gradually when you perform this test:

Tap the suspected tube **very** gently and listen for noises. A slight knocking is to be expected but a loud bang/crackle is not acceptable. Also, if the tube starts 'ringing' when it is taped or it was 'ringing' before and after taping it temporarily stopped- it is microphonic.

Pre-amp tubes such as the 12AX7, have been designed and manufactured for **low noise/microphony operation**. At the factory the ones branded for 'low noise' are specifically selected through testing after the manufacturing process. Some factories for instance may select 100 out of a batch of 1000 after 'burning in' and testing, and a further selection of 10...20 out of the 100 batch may then be chosen as very low noise types.

I always buy low noise tubes and then through testing here in the workshop I further select 1...2 out of 5 for **extra low noise** operation to use in noise sensitive circuits. The nearer a tube is to the input of a preamp/processor, the louder its noise will be. In an output stage even a noisy tube may appear to be quiet. If the **first tube** of the **input stage is noisy, its noise will be amplified by the rest of the system.** If this tube is also in an input stage that accepts very low level signals (like for instance the output of a microphone) the noise level will be higher. If a tube of the **same type** that happens to be less noisy exists somewhere else in the unit, further away from the input in the signal chain, and it is **swapped with the first one, the noise may be substantially reduced.** 

The **first tube** in the EQ\_REVERB is the **12AY7/6072 (V1)** but because the unit is designed to accept studio line level signals (0dBu, 0.775Vrms) if it becomes noisy its noise will be low in low gain level settings, but will **increase if the input gain is set higher.** The 12AY7 noise will **disappear** as soon as you set the **GAIN control to zero**, if it does not that means that the problem **lies elsewhere**.

The REVERB OUT PRE-AMP tube 12AX7/ECC83 (V5) at the very back is right at the **input of the reverb** return chain, and it is far more susceptible to **noise**, because it amplifies the very low signal levels coming out of the spring tank output transducer, this stage is very similar to a mic pre-amp. However, this noise will only appear if you turn up the REV RET (reverb return) control. Note that because this tube is shared between both channels, the noise will appear in both channels too.

The high mid booster (12AX7/ECC83) **V4** is further away from the input in the signal chain and it is not really susceptible noise issues.

Noise is not a serious problem with the two PCL86/14GW8 (V6, T&B) reverb driver output tubes unless they are so noisy (...nearly faulty). The 12AU7/ECC82 EQ DRIVER V2 (T&B) is somewhere in between as far as noise is concerned but V2 and V3 T&B (T&B=top and bottom channels) are more likely to generate noise in cascode mode.

### EMISSION LOSS

As a tube **ages**, it gradually loses its ability to produce the same level of audio signal that it was capable of in the beginning of its working life. This is due to gradual fall in transconductance (gm), a parameter that quantifies the ability of the cathode to emit electrons. In simple terms the emissive material coating on the cathode erodes with age, fewer electrons are emitted and the signal level drops.

There are other reasons for a drop in gm due to age such as **air leakage, and/or electrical leakage between components inside the tube.** Emission loss (due to ageing or whatever reason) can cause a tube to generate more and more **hiss with time,** which may appear intermittent at first but then it becomes continuous.

Noise, microphony and leakage problems, can sometimes (but rarely) exist in new tubes especially if they are bought from places where they are not 'burned in' and tested properly.

Another problem which may not necessarily be due to age, but it is a fault that usually appears with age is **crackling noises which can either be continuous or intermittent without any vibration**. *This is due to the irregular arrival of electrons at the plate of a tube.* 

### IONIZATION

**Ionization** is a serious problem as well as gas currents and electrical leakage between different tube elements which cause it. These problems can develop with age but can also happen to new tubes especially if they operate too close to their maximum ratings. They are not common in pre-amp tubes but they occur in power tubes and can damage other circuit components too, because ionization can cause internal and external arching plus high current surges.

lonization can occur in medium power tubes like the PCL86/14GW8 in the REVERB DRIVER if such medium power amplifier circuit is not designed and built with safe and reliable operation in mind. To avoid premature failure of any kind when I design and build any circuit I make sure that all tubes are operating conservatively well **below their maximum ratings**.

As I already mentioned **newly bought tubes can occasionally be faulty too or become faulty after a few hours of operation** if they are not 'burned in' and tested correctly. **Such tubes can damage other components inside the main and/or the power supply unit.** 

New tubes can also be damaged during transport if they are not packed properly.

Please buy tubes from reputable sources. I keep spare new tubes for all units I have designed and built, they have been fully tested in my workshop.

For best reliability and longevity the unit must:

1) Only be switched on when it is being used.

2) Switch on the power supply first, wait 40 seconds then press the stand-by switch (S-BY) to operate.

3) Always go to stand-by (OFF, that is UP) when you change from Triode to Pentode and vice versa and then press the S-BY switch down again to operate.4) Allow plenty of ventilation around the unit.

HOW OFTEN AND WHEN IS IT NECESSARY TO REPLACE TUBES

There is no definite rule or answer. It I depends on how often, how many hours per day the equipment is being used, the tube itself and what it does.

The two **PCL86/14GW8 reverb driver outputs must be replaced more often** due to their higher power dissipation.

As a rough guide : If you operate the unit say for instance 5 hours a day for a whole year, the PCL86/14GW8s may need replacing every 12...18 months whilst the rest every 2...3 years, unless they become noisy.

### FUSES

Fuse choice is based on estimates and some of the fuses in the power supply unit have slightly low ratings on purpose to ensure that they blow if something goes wrong and protect the circuits that they are connected to.Occasionally, a fuse may fail like any other component (it is a thin piece of wire after all) in which case it must be replaced with the same type and value. A fuse that is faulty must not be colored black, if it is then the fuse is blown, in which case the reason why it is blown must be investigated before it is replaced in order to avoid further damage.

Every fuse in the power supply unit has an LED in its vicinity which is lit during operation to indicate that the fuse is not blown/failed. If an LED is off and you want to replace its fuse please:

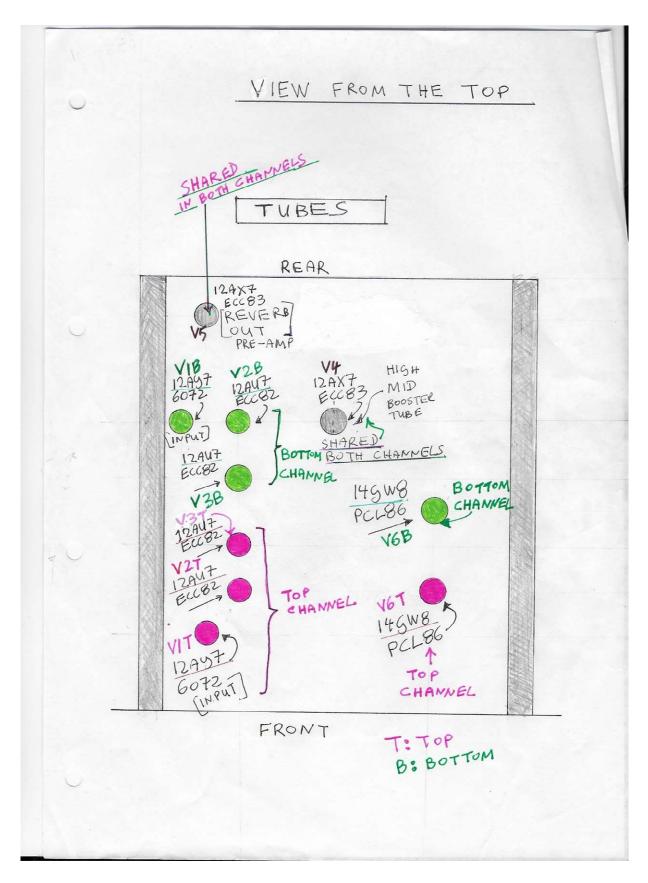
### (1) Unplug the power supply from the mains

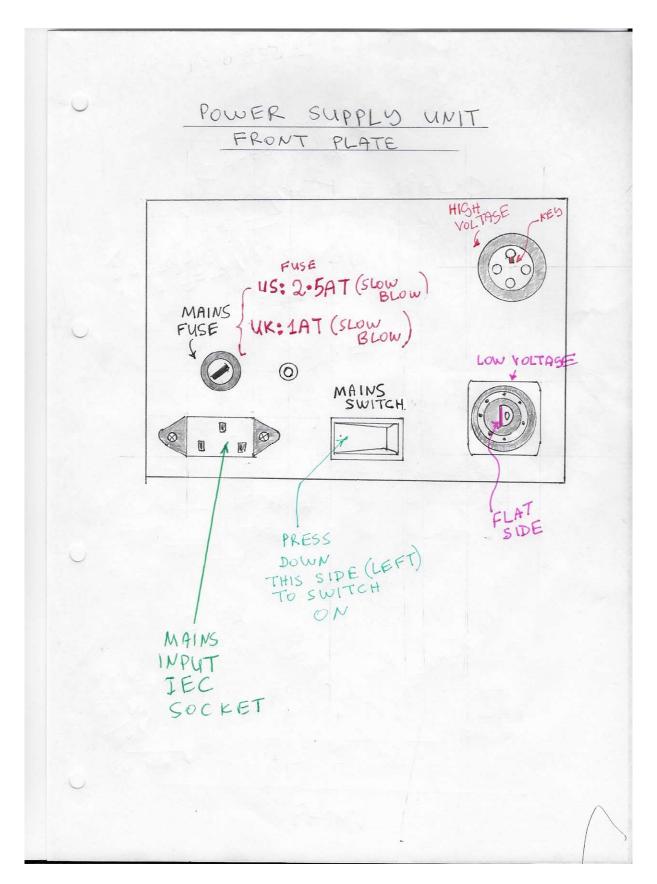
### (2) Wait approx. two minutes for the power supply capacitors to be discharged

(3) Carefully replace the fuse.

# ABREVIATIONS

OD=overdrive SW=switch CTR=control LIM=limiter FREQ=frequency REV=reverb MO= momentary on, as regards to a switch in reverb driver section which has the option to switch on the reverb momentarily when pressed up... it switches it on properly when pressed down... switch in the middle the reverb is off. TRI=triode PE=pentode RET=return BAL=balanced UN-BAL= unbalanced LOG=logarithmic LIN=linear SE = single ended NFB= negative feedback, a distortion reduction mechanism





Page **37** of **38** 

