# ARGIRIADIS
## ANALOGUE ELECTRONICS

# OWNER'S MANUAL

## AAE RECORDING
## PREAMP & REVERB

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# CONTENTS

# DESCRIPTION

This unit accepts various input sources and amplifies them for recording purposes. An extra facility is the spring reverb section and the amplified reverb drive signal. Both of these signals can be mixed with the original signal. It is a purely analogue design using valves and transistors working on relatively high voltages (up to 320Vdc) producing high level signals, a lot of headroom i.e. high dynamic range.

It can be adjusted to produce clean, mildly or even heavily distorted sound. The type of harmonics generated can be even and low in order (musical) but sometimes if so desired, odd too, for harsher sound (for instance bass, drums etc.) Harmonic distortion can be combined with frequency adjustment through careful use of the gain switches/controls and the inbuilt EQ. Overdrive distortion combined with the reverb sections can also produce a variety of different sounds. However, like most analogue sound processors **experimentation by the user** is necessary.

There are three main inputs ;

1a) Transformer microphone input with an impedance of 2K approximately. With dynamic microphones there is more sound colouration due to the way that the input transformer interacts with the inductive part of the impedance of these microphones.

1b) A transformer-less microphone input that has higher input impedance (10K approx.) They both have a 48V supply facility for condenser microphones and there is an optional 80/140Hz high pass filter on this input, which can be by-passed if not required.

2) A line input with an adjustable input level control that can accept anything from guitar instruments (including bass), keyboards (including synthesizers), computers, and other recorders etc. with an input impedance of well over 1M.

3) A standard musical instrument input – this will be for guitars and bass (but if it sounds good other instruments as well). The input impedance here is 1M, and the overdrive will come directly from the instrument and that depends on the the player too. No bass or guitar preamp has a input level control. It somehow sounds more natural to feed the input stage directly.

## *THE MAIN INPUT STAGE*

The selected source to be processed is fed to a valve amplifier stage that is employing two valves which can either be 12AX7 or 12AY7 i.e. they are interchangeable. It is possible to remove one valve and fit the other without any circuit changes. The 12AX7 is a very high gain common preamp valve, the 12AY7 has got slightly less gain and is a lot cleaner, it was used a lot in 1950's guitar amps of the tweed era. I personally found through experimentation and bench testing the best combination is to have the 12AY7 as the first i.e. the very far left, followed by a 12AX7.

This stage is switch-able to function either in triode or cascode mode. Cascode is a low noise pentode made out of two triodes, the EF804 used on the V72 is a pentode. Both triodes inside the envelope on both valves are connected in parallel for higher transconductance and better noise performance.
The triodes are working under such operating conditions that they normally give a clean signal.

When over-driven by a small amount they produce low levels of mainly even harmonic distortion which may also be expressed as 'warmth'. Many microphone/instrument pre-amps use triodes as input amplifiers and sometimes they cascade them for more gain. Pentodes are also used for this purpose but if over-driven they produce more distortion both odd, even and of higher order. In fact they can sound quite 'dirty'. The cascode is very similar to the pentode but less noisy and it does have its own little sound.

In triode mode, the stage has got enough headroom for a clean sound, for most guitars including bass. In cascode it will sound a bit crunchy and dirty but this also depends on a) what kind of guitar pick ups the instrument has and b) the player. For synthesizers, keyboards etc. that have a high output, the line input control can be used to reduce the input level if clean sound is a requirement.

As far as microphones are concerned the triode/cascode choice is not just for sound coloration but also a matter of gain. This means that with low level microphones the cascode can be used. For high level microphones and clean sound triode mode operation will provide enough gain.

**Therefore this triode/cascode choice does not only alter the sound but the gain too.**

The output of the valve stage enters a high voltage transistor buffer to produce a signal current high enough to drive the passive inductor based equaliser (EQ) that follows.

This EQ is important because it can emphasize certain harmonics, eliminate others and give a certain character and "colour" to the sound if required . The EQ interacts both with the stages before it and the stages after it.

For instance you can a) overdrive the input valve stage (before the EQ) and use the EQ to boost or attenuate harmonics that have been generated by this stage. Similarly you can b) keep the input stage clean and cut or boost certain frequencies of the signal to be over-driven by either or both the solid stage gain that follows and the medium power valve amplifier that comes after it. Of course a combination of a) & b) is possible too.

There are four frequency regions to boost/attenuate – bass, low mid, high mid & treble; the low , high mid and treble sections are also divided in two. The bass and treble are shelf type filters.

The low mid can be split to either 380Hz or 530 Hz and the high mid to 1kHz or 2kHz. Boost of the low mid frequencies introduces warmth, however too much boost also reduces clarity. For this reason when mixing down a few musical instruments including voice, some engineers prefer to attenuate these frequencies. Boosting the high mid frequencies produces a lo-Fi type of sound especially over the 1kHz frequency region, a slight cut on the other side emphasises transparency. A boost on the 2kHz introduces brightness, presence, sharpness, fast attack, grungy dirty sound. Again an attenuation cleans it all up.

The input tubes that amplify the signal that drives the EQ and the tubes that follow the EQ are working freely within the designed operating space without any tightly controlled negative feedback (NFB). This design, in terms of sound, produces interesting tonal variations due to the way the inductors and capacitors in the four tone control sections interact with the harmonic distortion that the tubes are producing. Most contemporary EQs and amplifiers use NFB, this makes them very accurate but not particularly good sounding. NFB is also responsible for hard clipping that doesn't sound good either.

As with all passive EQs (and to a lesser degree most active ones too) there is some interaction between the two mid range controls and also the low mid range interacts with the bass control and the high mid range interacts with the treble control.

A switchable high voltage transistor gain stage follows to compensate for the attenuation that the passive EQ introduces. For most input signals this stage is necessary unless the line/mic input levels and/or EQ controls are set relatively high. This stage contains a soft clip limiter that produces low order odd harmonic distortion when this stage is over- driven The sound will be kind of 'dirty' similar to a guitar fuzz pedal but a lot milder possibly good for modern percussion and bass especially if the input stage operates in cascode mode. This stage can be switched out the signal chain if it is not required.
Over-drive conditions in various stages are indicated by various LED's on the front-plate.

The DRY control (DRY = non-reverberated signal) sets the amount of signal that enters the mixing stage of the system.

## *THE REVERB DRIVE*

The signal for the reverb enters a frequency selector stage which offers the following signal options for reverberation ;

a) the signal after it has gone through the main system i.e. post EQ which means that the dry signal EQ settings effect the reverb send and amplifier signal too.
b) the signal which has gone through four different frequency fillers : low, low mid, high mid & treble.
c) the signal before it enters the dry section EQ

These six frequency modes can be selected through a rotary switch to create different **reverb** and reverb **amplifier** sounds.

A gain stage then follows (PCL 86 triode section) and the reverb drive control plus a gain switch set the amount of signal that enters the **reverb drive amplifier.** This is a medium power stage using the PCL86 pentode section in single ended mode which is transformer coupled to the spring input transducer. There are two spring drive options ;
a) ultra-linear (UL) which is transparent and frequency balanced
b) pentode for sharp, bright and more aggressive sound

As the reverb drive level increases you can create grungy, distorted reverb not only due to amplifier distortion but the springs themselves also create their own distortion.

## *THE AMPLIFIER SIGNAL*

This control sends the reverb drive output signal **before** it reaches the spring input transducer inside the tank. This is an electro-magnet and the way that it interacts with the PCL86 valve/output transformer plays an important role in the sound here.

Obviously, the UL/PE switch effects the 'amplifier' signal too. When set in UL the sound is clean whilst in PE harsh, and it gives a serious boost in presence. Very good for 50's type of guitar sound. Essentially this section is very similar to the ECC99 output section (explanation follows shortly) but a lot more low-fi.

a) UL distortion is warm & musical
b) PE sounds aggressive, bright, good sound for guitars and percussion.

The **'amplifier' control** adjusts the level of this signal before it enters the main mixer. It can be switched on/off by a miniature toggle switch.

## *THE REVERB RETURN*

The signal coming out of the reverb tank is fed to a low noise transistor stage in order to bring it to a level high enough for the mixer of the system, and its level can be adjusted by the **reverb return** control.

## *THE MIXER STAGE*

The three signals i.e. **dry, amplifier,** and **reverb** return are fed to a high voltage solid state mixer with an in-built soft clip limiter which drives the **master** control that sets the amount of signal that enters the output tube.

By turning down the **dry** and **amplifier** controls all you get is reverberated signal coming out of the unit, an interesting but extreme kind of sound. Similarly you can have only dry or amp signal only. Obviously, you can have all of these three controls in various settings to achieve a variety of reverberated and non-reverberated sound.

There are DRY/REV **kill** and AMP **on** switches that can remove anyone of these sections without changing any level settings.

## *THE OUTPUT STAGE*

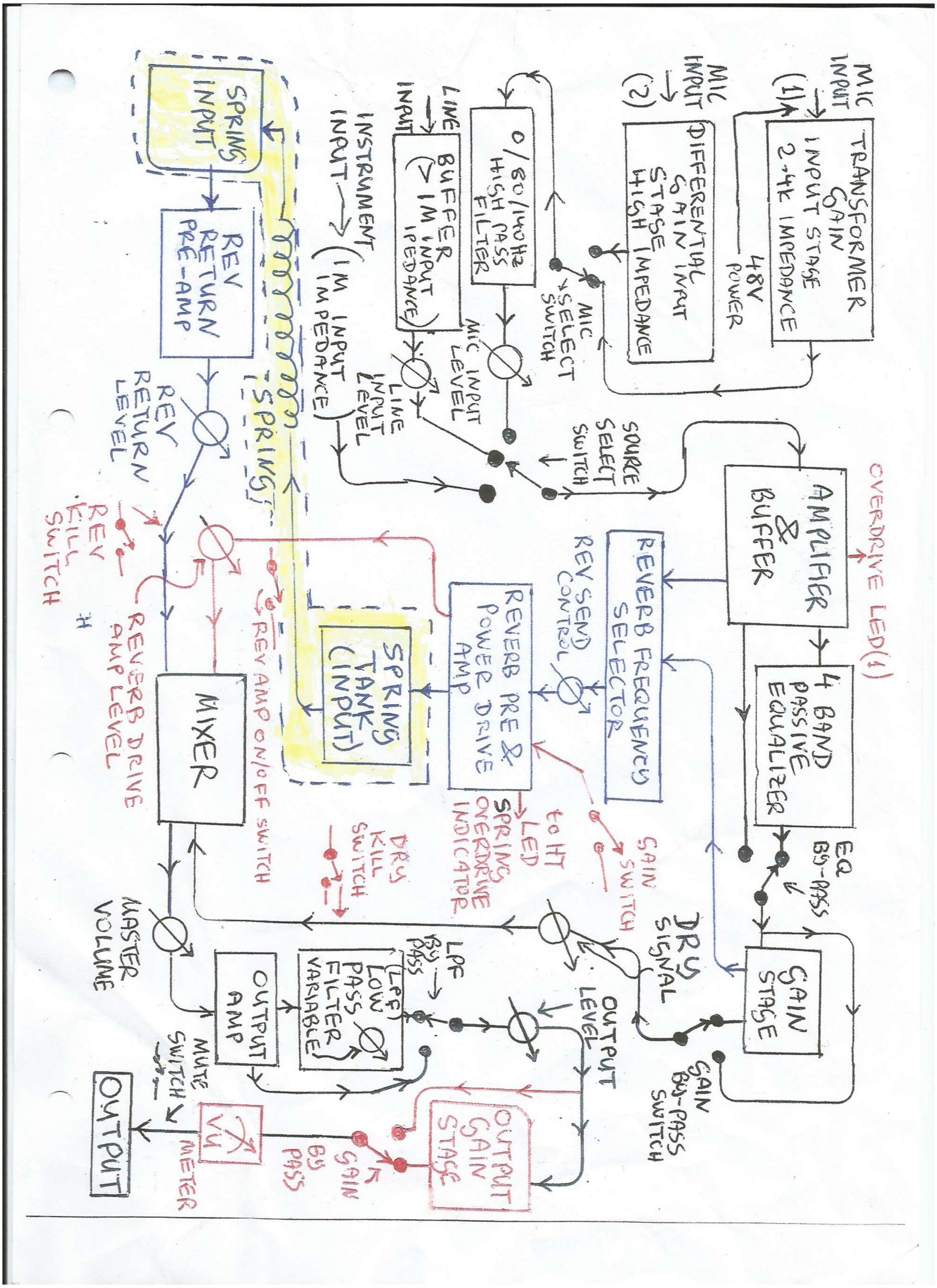
This is a parallel single ended medium power triode (ECC99) stage which uses a high quality output transformer and no negative feedback (NFB) to sound as natural as possible. NFB removes distortion by comparing the output with the input signal in an amplifier and then subtracting the distortion from the input of the stage. Triodes such as the ECC99 under the correct operating conditions produce low levels of second harmonic distortion which is proportional to the level of the input signal. When the input to this stage is set low through the use of the **master** control the output will be very clean with hardly any distortion. If by doing this the main output of the system is very low too the **output** control may have to be set higher. There is also an extra output gain stage can be switched in if necessary.

As the signal level to the ECC99 stage increases slightly more even harmonic distortion will be produced especially on the bass frequencies, creating a sound coloration often referred to as warmth. The sound will still be clean but more musical. What makes this stage different to the input triode stage is the way the inductive properties of the output transformer interact with the harmonic distortion of the ECC99 output tube.

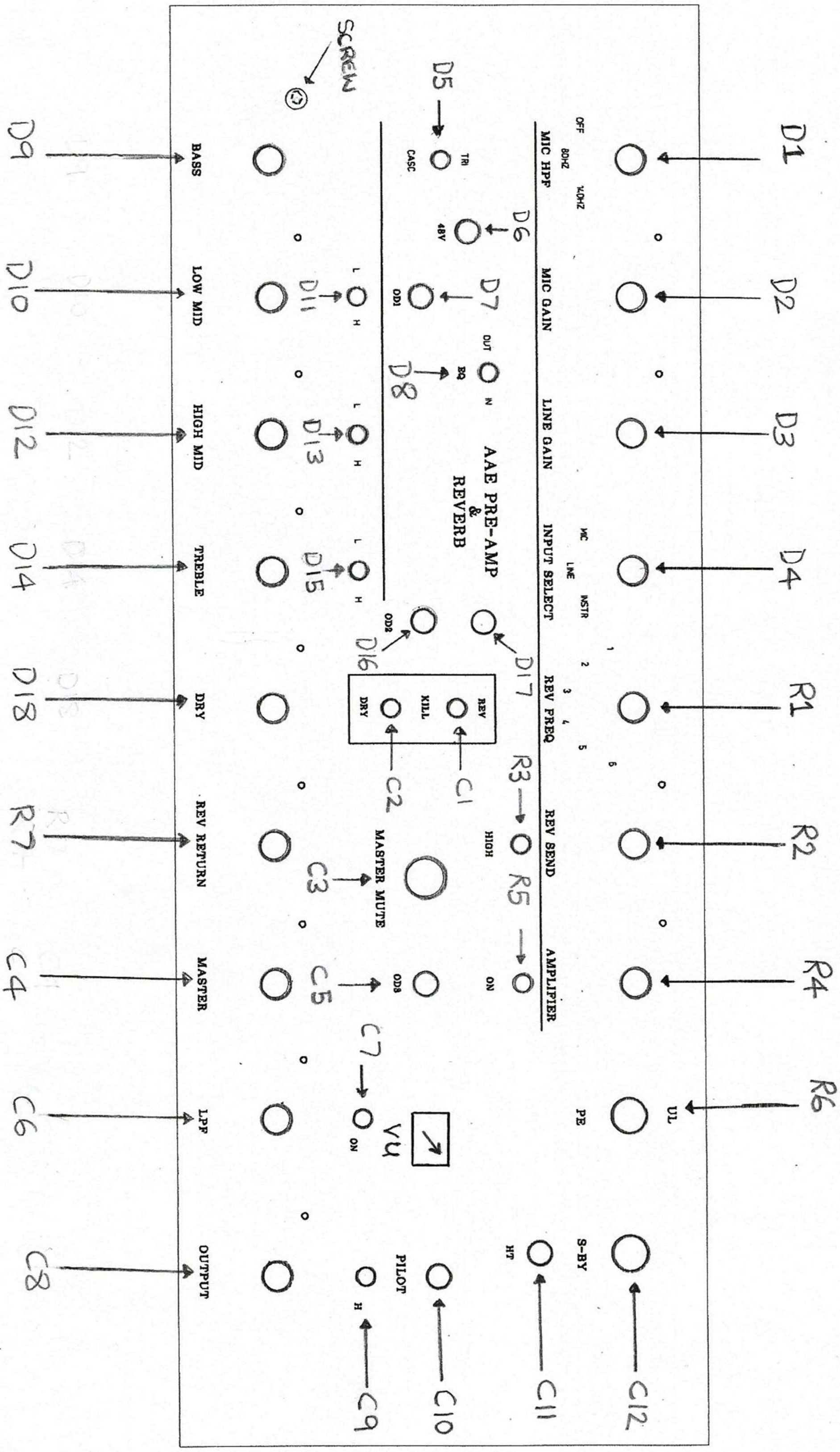
As the input signal increases even more the distortion will become more pronounced and harsh. Note that single ended triode stages clip very softly creating a very characteristic sound.
There is also a second order low pass filter (LPF) at the output that can be switched in or out. The **LPF** control sets the cut off frequency from where the attenuation begins.

**ABREVIATIONS**

SW = switch
OD = overdrive, often referred to an LED indicator showing when a stage is being over-driven
BFFR = buffer, a current amplifier to match high output impedance circuits to low input impedance ones such as the equalizer sections.
EQ = equalizer, also refers to a small section of an equalizer
FREQ = frequency
CTR = control, potentiometer with a knob
UL = ultra-linear
PE = pentode
TR = triode
CASC = cascode
REV = reverb
RET = return, such as reverb return
NFB = negative feedback
LIM = limiter
MID = mid range




**FUNCTION DIAGRAM**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&
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C12

**FRONT PANEL**

# FRONT PANEL CONTROLS

## *DRY SECTION*

D1 : MIC HPF – rotary switch. Microphone High Pass Filter (0/80/140 Hz) selector. In the '0' position this filter is bypassed.

D2 : MIC GAIN Linear (Lin) potentiometer (pot) – controls the microphone input level i.e. the signal coming out of the MIC input pre-amps.

D3 : LINE GAIN Lin pot – controls the line input level.

D4 : INPUT SELECT rotary switch - selects the source (MIC, LINE or INSTR.) that enters the input “amplifier & buffer” stage.

D5 : TRI/CASC (Triode/Cascode) miniature toggle switch – selects the operating mode of the valve input “amplifier & buffer” stage.

a) triode : clean, moderate gain, transparent sound
b) cascode : more coloration, higher gain and if overdriven it produces more 'dirt' & distortion.

D6 : 48V Power LED - lights up when the 48V (phantom) power is applied to the MIC inputs

D7 : OD1 LED – lights up gradually when the input stage valve amplifier starts being overdriven, the overdrive here is very smooth and soft.

D8 : EQ IN/OUT min switch – in the 'out' position the EQ is bypassed.

D9 : BASS Lin pot– bass level control for frequencies below 250Hz, approx. +/- 9dB's on 30Hz, when the low mid control D10 is flat i.e. set at 5. The boost/cut increases when D10 is at lower settings and decreases when it is at higher settings. The maximum variation on 30Hz is approximately +/- 16dBs.

D10 : LOW/MID Lin pot - low midrange control, works in-conjunction with D11 switch.

D11 : LOW/MID select min toggle switch.

a) in the L position D10 affects frequencies around 380Hz giving approx. +/- 7dB when the bass control D9 and the high mid control D12 are flat i.e. on 5. The boost/cut increases when D9 & D12 are at lower settings and decreases when D9 & D12 are at higher settings.
b) In the H position D10 affects frequencies around 530Hz giving approx. +/- 6dB when the bass D9 & the high mid D12 are flat i.e. on 5. The boost/cut increases when D9 & D12 are at lower settings and decreases when D9 & D12 pots are at higher settings.

D12 : HIGH/MID Lin pot – high midrange control pot, works in-conjunction with D13 switch.

D13 : HIGH/MID SELECT min toggle switch

a) in the L position D12 affects frequencies around 1kHz giving approx. +/- 9dB when the low mid control D10 & treble control D14 are flat. The boost/cut increases when these controls are at lower settings and decreases when these controls are at higher settings.

b) In the H position D12 affects frequencies around 2kHz giving approx. +/- 7dB when D10 & D14 are flat. The boost/cut increases when these controls are at lower settings and decreases when these controls are at higher settings.

D14 : TREBLE Lin pot – high frequency control, it works in-conjunction with D15 switch.

D15 : TREBLE Select switch min toggle switch

a) in the L position D14 controls the lower treble frequencies giving an approx. +/- 7dB at 3kHz & +/- 11dB at 10kHz when the high mid control D12 is at 5. The boost/cut increases when D12 is at lower settings and decreases when D12 is at higher settings.

b) In the H position the control shifts onto slightly higher frequencies giving an approx. +/- 6 dB at 4.5 kHz & +/- 10 dB at 10kHz. The boost/cut increases when D12 is at lower settings and decreases when D12 is at higher settings.

D16 : OD2 LED - lights up when soft clipping i.e. limiting takes place in the post EQ solid stage “gain stage”.

D17 : POST EQ gain switch. When this switch is pressed down it gives an extra 9dBs of gain to the signal before it is fed to the D18 DRY level control.

D18 : DRY Lin pot – controls the dry (i.e. non-reverberated) signal that enters the mix.

## *REVERB SECTION*

R1 : REV FREQ rotary switch – selects the frequency range of audio signal that enters the reverb spring drivers.

### Positions

1. Signal is picked up after the main EQ. i.e. the EQ also affects reverb amplifier and reverb spring sections.
2. High frequencies, over 2kHz
3. High mid, around 2kHz
4. Low mid, around 850Hz
5. Low frequencies, less than 150Hz
6. All frequencies, that means that the signal to be reverberated is picked up before the main EQ.

R2 : REV SEND Log pot – controls the level of the signal that drives the spring reverb amplifiers.
R3 : HIGH min toggle switch – when set on HIGH it gives an extra gain on the reverb drive amplifiers. **Press mute switch C3 whilst operating R3.**
R4 : AMPLIFIER Lin pot – mixes the output of the reverb signal **power amp before** it enters the spring with the **main output** signal.
R5 : ON min toggle switch – when pressed down the AMPLIFER signal coming out of R4 enters the mix.

R6 : UL/PE large toggle switch – selects the reverb output amplifier operating mode.

UL – ultra linear, clean sound. Most 1950s Hi-Fi operated in this mode.
PE – pentode, harsh more presence & distortion.

**IMPORTANT : Please switch off the high voltage via the standby switch before operation of the UL/PE switch, choose mode of operation via the (UL/PE) R6 switch, then reconnect the high voltage via the standby C12 switch.**

**Failing to do so can cause serious damage to the reverb output valve and/or output transformer.**

**Also, whenever using these two switches R6 & C12 press and hold the MUTE button, C3 to avoid loud pops in the units output see C3 below.**

R7 : REV RETURN Lin pot : controls the level of the reverberated signal entering the mix.

## *COMBINED SECTION*

C1/C2 : REV/DRY KILL min toggle switches
a) C1 removes the reverberated signal from the mix when pressed down
b) C2 removes the dry signal from the mix when pressed down

C3 : MUTE red momentary push button switch (non locking type) – this is a **very important** button because when changing the gain settings through switches, switching from triode to cascode, changing from ultra linear to pentode, and vice versa ; loud pops will appear in the units output. To avoid, press and hold the MUTE button whilst operating these switches.

C4 : MASTER Log pot – controls the signal entering the output valve amplifier and the amount this amp is overdriven if required.

C5 : OD3 LED – lights up when the output valve amplifier enters overdrive.

C6 : LPF Lin pot – continuously varies the cut-off frequency of the Low Pass Filter

| | | |
|---|---|---|
| at | 0 | it is 1kHz |
| | 5 | 3.3 kHz |
| | 10 | 14kHz |

C7 : ON min toggle switch – LPF on/off switch

C8 : OUTPUT Lin pot – controls the output level of the whole system.
**Always start operating the unit with C8 on 0.**

C9 : H extra gain output min toggle switch – on H i.e. high setting inserts an extra gain (+12dBs) stage onto the output. Always start operating the unit with this switch on Low i.e. upwards. Only use on H if the input gain levels are low. See text later.

C10 : PILOT LED
a) lights up when the power supply (separate unit) is switched on. Indicates that the heaters of the valves are getting hot.
b) goes dim when the MAIN OUTPUT LIMITER operates indicating a very high signal coming out of the unit (approx. +/- 10V peak to peak).

C11 : HT LED

a) lights up when the standby switch C12 is on, the high voltages applied and the unit is ready to operate.

b) goes dim when the REVERB DRIVE AMP/SPRING is being overdriven.

C12 : S-BY large toggle switch – switches on/off the operating voltages of the unit.

## *REAR PANEL*

MIC 1 : XLR input socket, **Microphone input** for the 2.4k impedance transformer input stage, this stage also has a 48V phantom power facility.

48V : rocker switch, switches on the **48V** power supply for **condenser microphones**

SELECT : rocker switch , MIC 1/ 2. selector : **Select each microphone by pressing toward MIC 1 or MIC 2.**

MIC 2 : XLR input socket transformerless **high impedance microphone input.**

LINE : ¼ inch jack socket **line input**

REV : **(phono) RCA**/Input socket **reverb return input**

INSTRUMENT : ¼ inch jack socket **instrument input, guitar, bass etc.**

OUTPUT : ¼ inch jack socket **units main output**

REV DRIVE OUT : ¼ inch jack socket the output of the medium power amp that **drives the input of the spring tank. Do not operate the unit without a load (spring input) connected to it. You can also connect a speaker to this output instead of the spring.**

# NOTES

1) This unit operates on high voltages and uses valves it is therefore capable of producing high output levels. These high level signals may generate enough of an output to damage mixers, speakers or even ears. The output of the unit **contains protection circuits** to avoid these problems but it is better to be cautious. This is why it is a good idea to **start quietly and gradually turn the output level up** so you can keep better control of the signal that comes out of the unit.

2) **Always** :
a) **press MUTE C3** switch first to mute the system before you use ( i ) Reverb send gain switch R3 and (ii) S-BY switch C12
b) Temporarily **switch off the high voltage** from the unit via standby switch S-BY C12 first **before you use UL/PE** R6 switch.

3) Under normal conditions the mute switch must be pressed when the triode cascode switch D5 is operated unless you choose not to. You can use the triode/cascode switch without muting the system as an effect to create the **reverb 'beat' sound**. This happens if you operate the triode/cascode switch while the reverb return is on, if you do s**o keep the output level low.**

4) **If the sound is missing from one section remember :**
a) for the DRY channel to work the C2 DRY KILL switch **must be off i.e. up.**
b) for the reverb channel to work the C1 REV KILL switch **must be off i.e. up**
**However,** for the AMPLIFER signal to enter the mix the AMPLIFIER switch R5 (below AMPLFIER control R4) **must be ON i.e. down.**

5) There is a **dot** on the top left of each controls to indicate the numbers i.e. settings. On the **Bass control D9** there is no dot but a **black screw** which serves save purpose.

6) LINEAR POTENTIOMETERS : These are used when proportional levels are important. For instance the OUTPUT knob (C8) that controls the level of the signal that comes out of the whole system to ensure that undesirable overloading/hard-clipping does not occur in the units that the preamp is driving. When C8 is set at 6 for example the level coming out of the unit is twice as high as when C8 is set at 3. Linear pots are also used in the input sections, mixing sections and the equalizer (EQ) but the EQ controls are not as proportional due to the nature of the passive EQ design.

7) LOGARITHMIC POTENTIOMETERS : The controls REV SEND R2 and MASTER C4 are logarithmic because it is easier to obtain accurate control of gain at low settings so that the best possible smooth and coloured “warm” sound can be obtained. When for instance the MASTER C4 is turned up, valve distortion takes place gradually at first until it gets to approx. 5, and then at higher settings the distortion increases quicker.

# OPERATION

**PLEASE DO NOT LEAVE THE UNIT ON IF IT IS NOT IN USE**

**ALWAYS SWITCH IT OFF AFTER OPERATION!**

## *POWERING UP*

### PREPARATION BEFORE TURN ON

Ensure that the stand-by switch (C12) on the front panel is off (the up position). This is because no high voltage must be applied to the tubes until they warm up.

This unit is capable of high output levels and even though its output is limited internally by a clipping circuit, if you send a high signal noise through it such as a loud buzzing caused by a faulty input lead, it may generate enough of an output to damage amps or speakers. It is therefore best to start quietly and gradually turn levels up so you can keep control of the units overall output levels.

**1) ON REAR PANEL :**
a) Connect the REV DRIVE OUT to the INPUT of the tank, **REMEMBER this output must never be left open (i.e. with no load connected to it) when the unit is operating.**
b) Connect the OUTPUT of the tank to the REV phono RCA/input socket i.e reverb return input.

For a first test it is recommended to use a simple monitor amplifier so connect the input of this amplifier to the ¼ inch OUTPUT jack socket i.e. the units main output.

2) SET :
a) the DRY signal control D18 at 10.
b) press down switch D17 (which is situated on top of OD2 LED) to insert the extra gain stage between the EQ and the output stage.
c) MASTER control C4 on 8.
d) units OUTPUT level control C8 on zero.
e) extra OUTPUT gain SWITCH C9 on the OFF position (i.e. up)

3) SET :
a) EQ IN/OUT switch D8 on IN
b) For flat frequency response :
BASS D9/MID 1 D10/MID 2 D12 controls on 5 & TREBLE D14 on 4 & ignore L/H (low/high frequency) D11, D13, D15 switches for the time being.
c) switch off the output low pass filter (LPF) through switch C7 (i.e. up)

4) The best input to try first is LINE because its level can be adjusted through LINE GAIN control D3
SET :
a) this control (D3) on 7
b) TRI/CASC switch D5 on triode mode i.e. TRI
5) Due to the complexity of the system it is recommended to try one section at a time i.e. the DRY section, get used to the gain/level controls and equalizer first before you try the reverb and finally

amplifier sections so

SET :
a) Mic gain control D2 on 0
b) REV SEND control R2 on 0
c) SEND/GAIN switch R3 on LOW (i.e. up)
d) REV RETURN control R7 on 0
e) AMPLIFIER control R4 on 0 & AMPLIFIER switch R5 OFF (i.e. up)

Ensure that the reverb KILL switch C1 is on (i.e. down) & the DRY KILL switch C2 is OFF (i.e. up)

6) The best source to use for the first test is a pre-recorded piece of music to understand the gain levels and EQ controls, filters, reverb & amp section. Next perhaps try a guitar or keyboard and finally microphones.

## *SWITCHING ON*

1) Connect the unit to the power supply – keep the stand-by switch off.
2) Switch the power supply on. The pilot LED will light up

***After*** approximately **40 seconds** press the S-BY switch C12 down. The HT LED will light up. ***Wait*** another **30 seconds** before you operate, to avoid any initial hum/noise which will quickly disappear. The is because the circuits need a few seconds to settle.

## *OPERATION*

1) With an audio source into the LINE input, gradually turn up the OUTPUT control C8 to get some sound. Observe the VU metre, and ensure that it doesn't register much higher than 0VU. Also, the PILOT LED must not go dim.

2) Now set the C8 (output control) very low (say at 2) and press down the output gain switch C9 to increase the output level. Get a 'feel' on how C8 and C9 interact on various C8 settings.
**Again observe the VU and the pilot light to make sure you do not exceed the maximum output limit.**

3) Reduce the output C8 a bit, switch off C9 (less output gain) and increase the line gain control D3.As the input line level increases LEDs OD1 and OD2 might light up indicating soft overload on the input stage and the post EQ stage.

4) TESTING THE EQ. Remember the EQ controls are highly interactive with one another, so the best way is to investigate what each control does separately.
a) SET all EQ controls on 0, then turn up each control on its own all the way up to 10 to discover how it affects the tone. Also try the L/H frequency shift switches D11, D13 & D15
b) Experiment by first mixing the BASS TREBLE controls only, whilst the MID RANGE MID 1 & MID 2 are on 0.
c) Experiment by first mixing the MID 1 & MID 2 controls only, whilst the BASS & TREBLE are on 0.
d) Finally mix all controls & switches to investigate the interaction effect on the tone.

You can hear the tone variations that each control offers by by-passing the whole EQ through IN/OUT EQ switch D8 every now & then.

5) Testing the output low pass filter.
a) Set the LPF control C6 at 10
b) switch on the filter through switch C7 (by pressing on to the ON position, on top of the LPF control)
Listen to the effect of this filter on various frequencies by using the C6 LPF frequency sweep control.

6) Now you can experiment using the LPF section with various EQ control/switch settings.

7) GAIN SETTINGS AND OVERDRIVE.
Set the EQ flat, switch off the LPF (C7 switch up) & experiment with one gain control at a time
a) reduce the DRY D18 control & increase the LINE GAIN D3 to hear how the input triode stage gradually overloads. A high output signal source such as a guitar with humbucker pickups, synths, high output CD player or mixer channel may be necessary in order to distort the input triode stage. As LED OD1 starts lighting up this is when second and higher even harmonics are being added to the original signal.
b) CASCODE MODE **:**
**IMPORTANT : USE THE MUTE SWITCH C3 EVERY TIME YOU USE SWITCH D5 TO GO FROM TRIODE TO CASCODE & VICE VERSA**
**press the mute switch C3, switch on to CASC using D5 & release mute switch C3**
This is in order to avoid any loud clicks coming out of the system.
The tone will be 'dirtier' & harsher with more gain and plenty of odd harmonic distortion especially on high level input signals which will show as you increase the LINE GAIN D3.
c) Repeat a) & b) above and increase/decrease one EQ control at a time. Finally experiment with various different EQ settings.
d) POST EQ EXTRA GAIN STAGE
Repeat all of the above but with & without the extra gain circuit. Use min. toggle switch on top of OD2 (D17) to switch it in/out of the system. This gain stage clips very softly and that happens especially when the EQ controls D9, D10, D12, D14 are set high, LED OD2 then lights up.
e) OUTPUT STAGE OVERDRIVE
Note that this is a very mild overdrive in terms of sound and it is more apparent in the low frequencies. Reduce the OUPUT C8 control and ensure that the output gain stage is off i.e. by-passed by setting switch C9 (on top of the output control) UP, Set LINE GAIN D3 on 6 & EQ controls on low settings for a clean sound. You may need the extra POST EQ gain stage.

Start increasing the DRY D18 & the MASTER C4 controls to overdrive the ECC99 single ended valve stage. When this happens LED OD3 will light up. If your audio source is not high enough, you may also have to increase the LINE GAIN control D3, and the EQ controls too in order for this to happen.

The distortion here is mainly even harmonic, primarily second. This stage also clips softly but it has its own tone because it operates in power mode.
If you want a bit more 'dirt' & distortion, you can turn up gradually all the other levels (for instance you can press switch D5 to have the input stage operating in cascode too. Then you will have a blend of sound between the input stage and the output stage as far as distortion is concerned. **Make sure that the OUTPUT level is low and switch off the OUTPUT GAIN stage C9.** Generally speaking **the VU meter must never show higher readings than +3VU** unless you are driving a very

insensitive piece of equipment.

f) CLEAN & TRANSPARENT SOUND
Keep levels and gains low LEDs stay on triode mode at the input, LEDs OD1, OD2, OD3 must hardly ever light up & turn up the OUTPUT control C8. If the output is still low, switch on the extra output gain C9. **Remember pilot LED C10 must never go dim because this indicates a +/- 10V peak to peak at the output.**

# THE REVERB SECTION

## *PREPARATION*

1. Make sure the reverb tank is connected properly see 1) on section "PREPARATION BEFORE TURNING ON".
2. To understand the tone of the reverb it may be a good idea to start on triode mode at the input stage (switch D5 on TRI) & UL mode (switch R6 on UL, ultra-linear), in the reverb drive amp.

a) **Always use MUTE C3 switch when you operate the TRI/CASC D5 switch.**
b) **Always use MUTE C3 and S-BY C12 (stand-by) switches when you operate the UL/PE R6 large toggle switch this is in order to avoid failure of the output valve/transformer.**

**PROCEDURE FOR OPERATING UL/PE SWITCH.**
**i) press the red mute C3 switch**
**ii) whilst holding C3 go to S-BY by turning switch C12 upwards. HD LED will go dark.** The high voltages are then removed from the unit.
**iii) still holding the MUTE switch choose UL or PE by using switch R6.**
**iv) still holding the MUTE switch switch on the HT (high voltage) again by pressing down the S-BY switch C12. The HT LED will light up again.**
**v) now release the MUTE switch.**
3a) Set the reverb gain switch R3 on low (i.e. up).
b) Remove the dry signal by pressing down the dry signal kill switch C2 so you can just test the reverb first.
c) Turn the reverb kill switch C1 up to enable the reverberated signal to enter the mix.
d) Set the MASTER C4 on 7 and set the output C8 control high but keep the output gain low via switch C9.
e) Set the REV FREQ selector switch R1 on 6 to drive the reverb without any equalisation first.

## *OPERATION*

1) Turn up the reverb return control R7 all the way up to 10 and then gradually turn up the reverb send R2 control. The mini power PCL86 valve amp will then drive the spring and the reverberated sound will be heard.
2) Turn on/off the REVERB GAIN switch R3 (under REV SEND control) in conjunction with the R2 REV SEND control to hear the effect of the increase of the spring drive. The **more drive** you introduce the **more** you may have to **reduce** the **return** signal by **turning down** REV RETURN control R7.

When the reverb drive **amp** and **spring** get **overdriven the HT LED will go dim.**

Adjust the tone by using the REV/FREQ rotary switch R1.

3) You can now reintroduce the dry signal (via kill switch C2) and try to mix both dry and reverb signals via R7 (REV RETURN) and DRY D18 controls.

4) THE PENTODE MODE
**Always use procedure 2b in reverb section – "preparation" when operating the UL/PE switch R6.**
Switch on to pentode mode. You will immediately notice the pentode tone is a lot 'sharper, brighter and bouncy' especially on the high frequency end and the reverb is a lot louder.

5) You can now try the frequency selector switch REV FREQ rotary switch R1 in both UL & PE. Remember on position 1, the reverb drive comes from the **output of the main units (dry signal) EQ.**
As far as the reverb is concerned the **most responsive EQ sections involved are MID 2 & TREBLE.** However, boosting the bass may also produce an interesting reverb tone.

6) THE AMPLIFIER MODE
a)To bring this into the mix just press switch R5 on the ON position under the AMPLIFIER control R4 and gradually turn up this control to get the desired sound.
b) To isolate it use KILL switches C1& C2 to remove the reverberated and dry signals.

When you use the AMPLFIER mode as an amplifier signal only or the REVERB (or both) and the UL/PE switch R6 is set on **pentode** mode, the **distortion can sometime be high,** especially if a guitar is used on the instrument input on certain guitar riffs. If you want to avoid this ;

(i) Keep the REV SEND control R2 on a low setting i.e. 2...4
(ii) Set the REV GAIN switch R3 on low i.e. up.
(iii)If you need to increase the AMPLIFIER signal turn up the AMPLIFIER control R4 up to 10, and use the REV SEND R2 as a AMPLIFIER/REVERB DRIVE level control.

This signal shares the same main EQ as the dry signal when the REV FREQ switch R1 is in position 1. One important point to emphasise here is that in all other frequency settings of the selector switch R1 i.e. 2,3,4,5,6 the main EQ only effects the DRY section and not the AMPLIFER and REVERB sections. Therefore frequency selector switch R1 can be used as an extra EQ in these positions for the AMPLIFIER and the REVERB.

## *MICROPHONES*

1) Plug in a microphone into one of the XLR MIC inputs on the **rear panel**. Use the one that matches the output impedance of your mic, you may have to try them both.. Also if you are using a **condenser** microphone press the **48V** rocker switch down **(rear panel again)** ,the 48V **green LED** on the **front panel** will light up.
2) Select this particular input via the SELECT rocker switch **(rear panel)**. This switch must be pressed towards the XLR socket that you are using.

3) Use the settings in EXAMPLE (1) below first, you may have to re-adjust later, it depends on the microphones and the recording room.
4) Ensure the MIC gain control D2 is on zero.
5) Set the rotary INPUT SELECT switch D2 on MIC
6) Now gradually turn up the MIC GAIN control.
7) Re-adjust settings, experiment with AMPLIFIER and REV RETURN according to taste/requirements, see EXAMPLE (1) below.

### *SPEAKER CONNECTION*

The REV DRIVE OUT output can also drive a speaker (8 Ohm) instead of a spring .The unit then can be used as a low power amplifier (a guitar amp) for room recording purposes. The REV SEND control R2 then serves as a volume control and the gain switch R5 function still applies.

The following controls will have no effect on the speaker sound DRY (D18), REV RETURN (R7), MASTER (C4), LPF (C6), OUTPUT (C8), AMPLIFIER (R4), MUTE (C3), and REV & DRY KILL switches (C1, C2). These controls will still affect the main output of the unit.

When the REV FREQ switch R1 is set on 1. the main EQ will affect the speaker sound.

Since you have removed the spring from the output the reverb will not be there.

**Make sure that there is either a spring or a speaker connected to this output or you may damage the reverb output valve/transformer!**

# EXAMPLES

1) Various instrument samples on the line input :

SET BASS D8 on 8
LOW/MID D10 on 4
LOW/MID switch D11 on H
HIGH/MID D12 on 6
HIGH/MID select switch D13 on H
TREBLE D14 on 8
TREBLE select switch D15 on H
LPF C6 on OFF by setting LPF switch C7 UP
DRY LEVEL control D18 on 7
MASTER control on 10

**These settings are also good for microphones too** but you will have to adjust the microphone gain to the correct level, it all depends on the mic and the room.

To generate an image of space and transparency switch off the AMPLIFER signal by setting AMPLIFIER switch R5 off i.e. UP. Also, operate the input stage on TRIODE mode.

For reverb :
SET REV SEND control R2 on 5.
REV SEND GAIN switch R5 off i.e. UP
Adjust REV RETURN control R7 according to taste.

To add a bit of 'low-fi', and if you're using **mics to add brightness and presence**
SET AMPLIFIER control on 5 and bring onto the mix by switching ON switch R5 i.e. press down..
You can try this on ultra-linear for cleaner sound.
Pentode operation will add a bit of edge.

2) For a clean 50's guitar sound you can plug a guitar in to the INSTRUMENT INPUT.
Switch onto TRIODE mode for high output guitars, and on to CASCODE mode for low output guitars via switch D5

SET BASS D8 on 5
LOW MID control D10 on 3
LOW MID switch D11 on H
HIGH MID control D12 on 8
HIGH MID select switch D13 on H
TREBLE control D14 on 10
TREBLE select switch D15 on L
Set post EQ gain switch D17 (on top of OD3 LED) ON i.e. down
Switch on Low Pass Filter LPF via C7 switch i.e. down
Set LPF control C6 between 5...10 (depends on guitar type)
DRY control D18 on 6
MASTER control C4 on 10
REV FREQ rotary switch R1 on 6
REV SEND gain switch R3 off i.e. UP
REV SEND control R2 between 2...7 (depends on guitar type)
UL R6 on pentode i.e. PE
AMPLIFIER control R4 between 8...10
Ensure that the amplifier signal enters the mix by pressing down the ON/OFF AMPLIFIER switch R5.

Insert reverb onto the mix through REV RETURN control R7 according to taste.
Experiment with the reverb select switch R1 too.

This way you boost high/mid & presence frequencies & cut all frequencies above them via the LPF to emulate the guitar speaker tone.

3) Another example for various instrument samples – bass, guitars, synths, percussion etc.
Kill reverb first via switch C1 (can be reintroduced later).

SET BASS D8 on 10
LOW/MID D10 on 5
LOW/MID switch D11 on H
HIGH/MID D12 on 5
HIGH/MID select switch D13 on H

UL/PE switch on PE ie.pentode

TREBLE D14 on 4
TREBLE select switch D15 on L
LPF C6 on 2
Keep reverb gain switch R3 UP (i.e. low). You may have to experiment with the reverb send control R2 to the get the right sound. Remember this control can be set higher if the R6 UL/PE switch is set on UL mode.

Now set the REV FREQ select switch on 1 so that the EQ affects both DRY and REVERB sections. Using the DRY KILL switch C2 and the AMPLIFIER ON/OFF switch R5 listen to either DRY or AMPLIFIER signal. Remember when C2 (DRY KILL) is turned down it stops the dry signal. When the AMPLIFIER ON/OFF switch R5 is up (i.e. in the opposite direction to the DRY KILL switch), it stops the AMPLIFIER signal.

Set the DRY D18 control and the AMPLFIER R4 control so that the level of the signal coming out of both sections is the same.

Compare the tone of both sections separately instead of blending them.

Now bring the reverb back into the mix by turning up the REVERB KILL switch C1 and use the REV RETURN control R7 according to taste.

For bass guitars and bass synth sounds, you'll find out the dry signal section sounds more complete, most other samples can just go through the amp section, which sounds more transparent on UL mode.
Now you can blend all the sections, and use the D18/R4/R7 controls to achieve the correct mix.
Finally, experiment with the TRI CASC switch D5.

# TECH NOTES

### *MAINS VOLTAGE*

In the UK the mains voltage is 240V AC whilst in continental Europe is 220V AC. At the rear panel of the power supply unit there is a slide voltage selector switch to enable a choice between 220V & 240V AC. The sliding point of the switch is RED. **When it shows 230V it is actually 240V AC. When it shows 115V what it actually means is it is set on 220V AC.**

**In continental Europe** always operate on 115V**, i.e. 220V AC**

Non of my suppliers could sell me a good quality switch with the correct label i.e. 240/220V at the time that this unit was built.

### *EARTH SAFETY BOND*

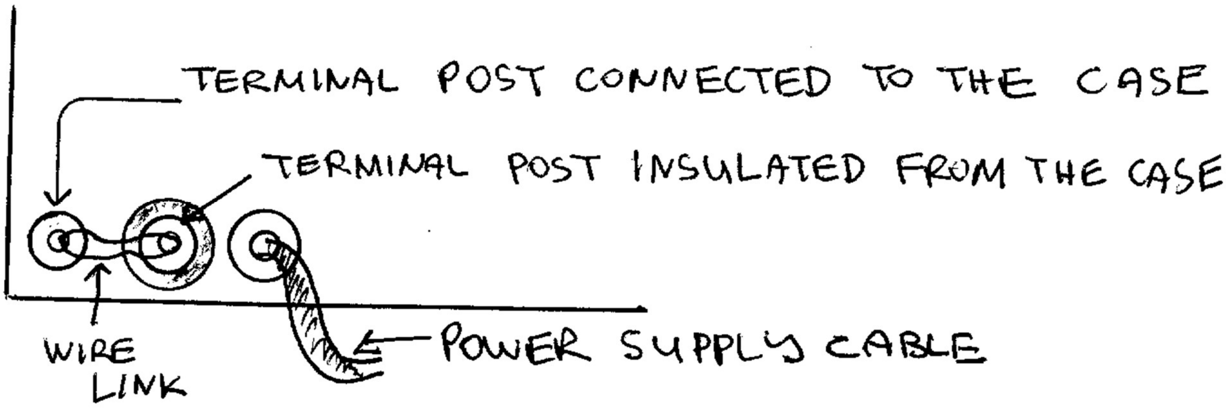
It is a safety requirement in the UK that in the power supply unit (PSU) the **mains earth (or ground)** is **permanently and securely connected** to the **metal case**. The mains earth is the centre pin on the IEC fussed chassis EURO PLUG. This plug is fitted at the rear panel of the PSU.

If for instance the mains lead is pulled by accident, breaks up and a live wire comes into contact with exposed metal work, the **mains fuse** will blow. This prevents electric shock to anyone who might be touching the metal case at the time. It is absolutely essential and necessary for **safety** that the EURO LINE SOCKET lead (what here in the UK we call 'kettle' lead) has a **mains earth** connection . This lead must always be plugged into a **mains outlet** with an **earth** connection to the mains power supply ground.

For similar safety reasons, and power supply unit noise reduction purposes, the negative terminals of DC high and low voltage lines are also securely and permanently connected to the metal case of the PSU. Therefore, the negatives of the DC voltage lines are also connected to the mains safety ground, through the PSU metal case.

### *SIGNAL GROUND*

On the main unit, at the rear panel, next to the power supply cable entrance, there are two TERMINAL POSTS. The one nearer to the power supply cable entrance is **screwed** on to a **black nylon insulator,** which is internally connected to the negative side of the power supply lines, which are also connected to the **signal ground**. These two terminal posts are securely connected by a WIRE LINK, so that the main units' case acts as an **electrostatic metal shield, to avoid hum, noise & interference.**




**For most applications the unit can be used with these two posts linked.**

## *GROUND LOOPS*

Sometimes, if this unit is connected to another piece of equipment, and they are both safety grounded, a **ground loop** may be developed between the safely ground and the signal grounds through the signal and mains leads. This ground loop will manifest itself as **hum.**

This situation is more likely to arise when the two units are powered from different mains sockets, especially if the sockets are far apart. 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 case of the PSU, the safety ground, and finally the yellow/green ground wire of the mains leads from one unit to the other.

## *GROUND LIFTING*

If this problem occurs, it is possible to brake the loop by removing the wire link between the two terminal posts, and disconnect the signal ground from the case in the PREAMP REVERB UNIT. This must eliminate the hum. Both units are then safety grounded, and only one has the ground signal connected to the case. The PREAMP REVERB UNIT is still electrostatically shielded for noise purposes. This is because **both cases are still connected to the signal ground through the mains safety ground wiring**. This is called **ground lifting** and it must **only be applied** to the signal ground/case connection.

***THE EARTH BOND BETWEEN CHASSIS (I.E. METAL CASE OF THE POWER SUPPLY UNIT) AND EARTH MUST NEVER BE BROKEN TO AVOID DANGER OF ELECTROCUTION!***

Ground lifting must always be done carefully and the signal level/volume controls must be kept low at first (during the test) and then gradually turned up. This is because the reason of the hum may not necessarily be a ground loop, and by removing the link the hum may become a lot louder.

The situation becomes more complicated of course if more pieces of equipment are connected together which means more ground loops may exist.

## *DEALING WITH HEAT*

The unit contains tubes, two of which are medium power and various medium power transistors (to produce low impedance drive for the passive EQ's etc.) mounted on heat sinks. The heat generated must be able to escape.

Both bottom and top plates are vented, and 3/4 inch sturdy rubber feet have been fitted on the bottom plate to ensure ventilation when the unit rests on a flat surface.

Remember, cool air enters the unit from below, travels through it upwards and leaves the unit from the vent of the top plate. Provided no heat is trapped anywhere, on a flat surface except for the power tube, no other component gets too hot to touch.

Similar rules apply to the external power supply unit (PSU) and the vents are on the rear panel. The PSU must also be kept in a cool place with plenty of air flow. The chord that connects the unit to the PSU is approx. 5 foot 10 inches long.

Now, in my experience, stacks of rack-mounted gear tend to accumulate heat. Most faults in repairs, I have made in the past, in equipment of various different makes were caused by excessive heat concentration or so called 'hot spots'.

If the unit is to be mounted in a rack then 4inches of space below and above is recommended.

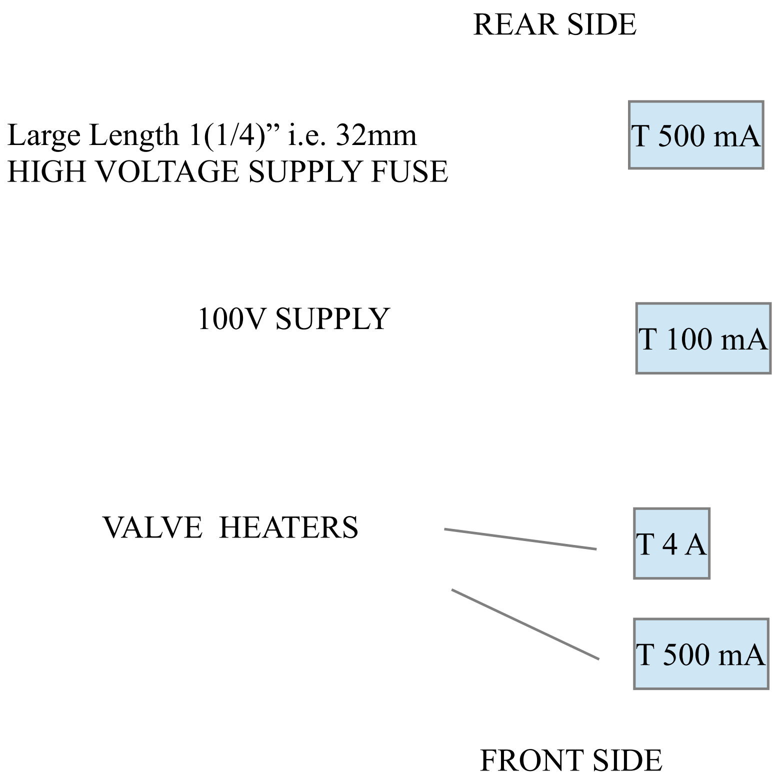
Also, if it is not operating it must be switched off.
Of course the ease of operating the controls and switches must be taken into consideration here as regards the position of the unit.

## *FUSES*

Fuse value choice is based on estimates this means that some of the fuses have slightly low ratings, to make sure that they blow if something goes wrong and protect the circuits that they are connected to. Occasionally, they may blow for no apparent reason, in this case they have to be replaced with the same type and value provided the blown ones are not coloured black. If this is the case then maybe there is something wrong with the circuit, which has to be rectified first before fuse is replaced.

The MAINS FUSE is a 20mm (T) 1A/250V type (T) stands for ANTISURGE. The FUSEHOLDER of this fuse is part of the IEC EURO LINE plug at the rear of the PSU case.

**POWER SUPPLY FUSES (INSIDE THE PSU)**
as viewed from above when the case is open.




**NOTE : RISK OF ELECTRIC SHOCK, DO NOT OPEN UNLESS YOU ARE QUALIFIED.**

**MAIN UNIT FUSES**

There are two quick blow (fast) fuses mounted on the top of the rear panel. They are both (F) 100mA/250V. One protects the spring reverb drive amplifier and the other the output stage amplifier..

## *AGE, NOISE & MICROPHONY PROBLEMS IN VALVES*

All valves generate noises, some do it more than others. Pre-amp valves such as the ones used at the input stage of the unit have been designed and manufactured for **low noise/microphony operation.** They are also specifically selected for these purposes after the manufacturing process. Some factories for instance may select 100 valves out of a batch of 1000 that exhibit the best low noise and low microphony characteristics, after 'burning in' & testing.

Some valves however can become noisy and microphonic with time.

Another thing that happens with valves is that **as they age,** gradually ,they also **loose the ability to**

**produce the same level of audio signal that they were capable of giving in the beginning of their 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 as the valve ages, the emissive material coating on the cathode runs out, fewer electrons are emitted and the signal level drops. There are other reasons for a drop in transconductance due to age such as air leakage etc.

Unfortunately some of the above noise & microphony problems can sometimes exist in brand new valves if they are not 'burnt in' and tested fully.

With emission loss due to ageing, transconductance fall can make a valve produce more and more hiss which is continuous and gets worse with time. This becomes more serious if the **noisy valve is the first one on the input stage, because any noise generated by it will be amplified by the rest of the system.**

## *HOW TO TEST PRE-AMP VALVES FOR NOISE & MICROPHONY*

1) By substitution - If the problem disappears when a suspected valve is replaced with another one of the same type, then the first valve is no good. It is important that a) the new one is known to be working (to avoid damaging the equipment, b) it is a low noise, low microphony type.
2) Microphony test – **Please keep the volume down when you perform this test.** Gently tap the valve and listen for noises. A slight knocking is to be expected but a loud bang is not acceptable. Also, if the valve starts 'ringing' when it is tapped or it was ringing before and after tapping and it temporarily stopped – it is microphonic.

Another problem and not necessarily due to ageing is **crackling noises** which can either be continuous or intermittent due to the irregular arrival of electrons at the anode.

**Microphony** can manifest itself a) the equipment makes noises when it is moved or slightly knocked. b) Ringing/whistling noise similar to mic/PA audio feedback which may start or stop at random.

Even new and high quality New Old Stock (NOS) valves can be seriously microphonic, some NOS 6SN7's are particularly prone to this problem. A valve is a mechanical device and the various parts are welded to one another and they can become loose with time. Age only makes things worse combined with heat and thermal cycling (the equipment switched on & off) deteriorate the adhesion of the mechanical elements.

Other serious problems associated with ageing in valves are ionisation, and gas currents as well as electrical leakage between different elements. These type of problems are more likely to effect the medium power ECC99 & PCL86 valves in the unit.

To avoid any premature failure of any kind when I designed the circuits I made sure that all the valves are operating conservatively and well below their maximum ratings.
As I already mentioned for best reliability and longevity, the unit must :

1) **Only be switched on when it it being used.**
2) **Always switch on the power supply first, wait 40 seconds, then press the standy-by**

**switch to operate.**

3) **Always ensure that the REV DRIVE OUT output is connected to the SPRING (tank) input.**
4) **Always go to S-BY mode when you switch on from UL to PE (switch R6)**
5) **Allow plenty of ventilation around the unit.**

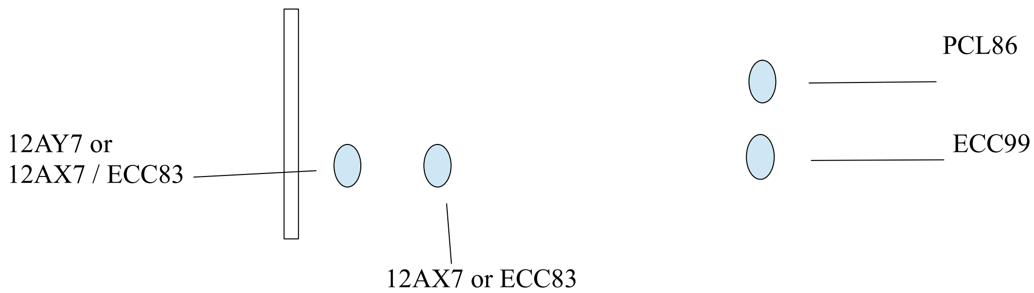
Brand new valves can be faulty or become faulty after a few hours of operation and especially ones not 'burnt in '& tested properly at the factory. They can damage other components in the equipment. **Please buy valves from reputable sources.** I keep spare new valves for all of the units I've designed and built. These valves have been fully tested in my workshop.

## *WHEN IS IT NECESSARY TO CHANGE A VALVE AND HOW OFTEN ?*

There is no definite rule or answer. It all depends on how long per day and how often the equipment is being used, the valve itself and what it does. The ECC99 mains output and the PCL86 reverb drive output valves must be replaced more often due to their higher power dissipation.

As a rough guide :
if you operate the unit say for instance 25 hours a week every week for a whole year both PCL86 & ECC99 may need replacing every 12...18 months whilst the 12AY7, 12AX7 (ECC83) every 3 years (unless some of then become noisy). Occasionally but not very often the ECC99 and PCL86 valves may become microphonic and noisy too.

**VALVE TYPE**




## *THE REVERB TANK*

The REVERB TANK has an INPUT and an OUTPUT as labelled on their metal case. The INPUT goes to the transducer which works in a manner similar to a tiny loudspeaker. It is a coil that sends the audio signals through the springs. It does this by converting an electrical audio signal current into a magnetic field that expands and collapses in proportion to the signal. This sets the springs into vibration. A small amount of electrical power is necessary to do this, and this is transformer coupled to this coil.

For this reason the REV DRIVE OUT MUST ALWAYS BE CONNECTED ON TO A LOAD when the unit is operating even though the reverb section is not in use. This load is obviously the TANK INPUT (or perhaps an 8 ohm loudspeaker).

The lead for this connection does not need shielding because it does not pick up hum, but for reasons mentioned; the wires must not to be too thin so that they snap.

The output of the tank is a pick-up type of transducer similar to a microphone which is susceptible to hum, so the lead for this connection must be shielded. This input must be treated like a high gain pre-amp input.

## *POSITIONING THE TANKS*

As far as unique 'dub' reverberated sound is concerned the best place for them to be, is inside a plastic tube, i.e. .a gas pipe of the kind you find in hardware or builders merchant. Some people actually take the tank to the store to try out different size pipes to fit. The tube must be longer than the tank.

Some recording engineers like to be able to use them in various different positions, i.e. one vertically while other horizontal etc. Some engineers occasionally like to tap the springs lightly for this 'reggae' tapping noise. You can also dampen the spring oscillation by inserting a bit of foam in the middle of the tank. The springs then will still wobble but without the more diffused reverb 'tail' at the end. Some like to softly run a coin along them to get an interesting sound too.

Note that reverb tanks are very susceptible to hum and interference and can be noisy too.