INTRODUCTION

The Series 12 mixing console is designed for full stereo operation throughout, but can be used in a mono mode if required. It is primarily designed for use in applications where a simple operation is required such as in busy newsrooms, dubbing suites, and outside broadcasts. The mixer will fully meet IBA technical specifications and may be used in a number of different operational configurations. It is ruggedly constructed and its circuit design is of modular construction.

It consists of three components: the mixer, power supply, and optional Penny and Giles fader pack. Each component may be 19" rack mounted and mounting brackets with handles are supplied if specified. Alternatively, the system may be fitted in console woodwork or a flight case. The mixer and power supply may either be used on their own or linked with the fader pack which can be retro-fitted after purchase.

The mixer is equipped with two microphone inputs, one switchable mic/line input, a mono line/telephone input which generates a clean feed of mixer output, four stereo line inputs, and a stereo line input selectable between four stereo sources. It also has three separate monitor feeds: guest headphones, presenter headphones, and monitor loudspeakers, each with their own individual volume controls.

This manual gives a detailed description of the mixer and all the information necessary for the correct installation and maintenance of the equipment.

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issue 2

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CHAPTER FOUR - ELECTRICAL DESCRIPTION

The Series 12 mixer is constructed with a number of separate pcb's. An electrical description of each board is outlined in this chapter. The main busbar harness which links the boards is of a plug-on type. Inputs and outputs etc are on separate plug-on connectors. This enables easy removal and replacement of the pcb's without any desoldering being necessary.

4.1 MICROPHONE INPUT CHANNEL

Low level signals from a microphone are fed direct to the 1:6 ratio input transformer T1. The output of which is amplified by IC1. The gain is variable using the front panel gain control RV1. After the non-inverting amplifier IC1, the signal is fed to the DBX VCA, IC4; left and right 'PFL' mix via S2 (when latched-in), R16 and R17; AGC card and only in the case of Channel 1, to the talkback bus via R18 (Channel 1 provides the talkback source for the two talkback outputs and the telephone channel communicate facility).

VCA: The VCA (IC4) is used to perform logic controlled muting and automatic gain control (provides constant output level when AGC switch S1 is latched-in) and has a nominal gain of 10dB. The output of the VCA is fed to a current-to-voltage converter configured around IC2A. IC2B forms a summing amplifier for the control voltage of the VCA. The output of IC2A is fed to the main stereo mix bus via the fader (if fitted) otherwise directly by linking A36 and A39. The output is also fed to the clean feed mix bus unless link P2 is reversed.

PFL: The 'PFL' switch, S2, when latched-in pulls the PFL D.C. bus low thereby switching the monitor section of the mixer to the PFL audio bus.

Pressing the momentary on switch, S3, sets or resets the channel on/off latch formed by IC3 which in turn controls the VCA. IC3 derives its power from the 5.6 volt line formed by ZD1, R25, C18, and C19 which in turn is fed from the 24 volt power line.

To enable local muting to take place, a link is made on the audio input connector between pins A5 and A6 on channels 1 and 2. In the case of channel 3, the link is made via the channels mic line input switch situated on the connector panel. This is so that when channel 3 is selected for line input, no local muting takes place. If a 'distant' microphone input is to be used as a source and no local muting is required, this link should be broken on the input channel that is to be used. When channel 3 is switched to line mode, as well as local muting being de-activated, an attenuator is switched in circuit between the input socket and the input transformer T1. The attenuator consists of 3 resistors R5, R6, and R7 which are mounted on the mic/line switch.

4.2 AGC CARD

The AGC card plugs on to the microphone channel pcb via connector P1. The AGC function is activated by latching in the AGC switch S1. IC1A performs a precision rectifier function, the D.C. output of which is fed into the time constant network R3, C1, R4, C2, and R5 and is then buffered by IC1B. IC2A forms an anti-log amplifier which provides a linear output voltage. IC2B is a D.C. amplifier with a variable gain and D.C. offset which enables a 40 dB control range of the VCA.

4.3 STEREO INPUT CHANNEL

The stereo line level signal is fed to an input attenuator which has a preset to enable balancing of the two phases and to provide a good common mode rejection. This is then fed to a differential input stage formed by IC1A (left channel) and IC2A (right channel). The output is fed via the dual front panel gain control, VR2A and VR2B, to the left and right gain trim amplifiers, IC1B and IC2B. The two amplifier stages formed by IC1B and IC2B allow the left and right signals to be trimmed to an equal level. The prefade signal is taken from this point and routed via the 'PFL' switch S1 (when latched in) and R33 and R34 to the stereo PFL bus. The signals from the gain trim stages are then fed via the fader (if fitted) to the channel-on relay RL1; to the main stereo mix bus; and also the mono clean feed mix bus via R35 and R36 and links 2A and 2B.

Logic control of remote stop/start relays, RL2 and RL3, is performed by IC3 which is powered from the 5.6 volt line derived from the 24 volt aux supply via R52, ZD1, C28, and C29. If no fader is fitted, link plug 1A/B is plugged in 'GRAM' postion and link 3 is fitted in postion 'B'. The momentary 'channel on' switch S1 when first pressed, then performs the function of activating the channel on relay RL1 and the momentary remote start relay RL3 simultaneously. Pressing switch S1 a second time will deselect the 'channel on' relay and activate the momentary remote stop relay RL2.

When the fader is fitted, link 3 is in positon A and link plug 1A/B can be fitted in either of two positions: 'GRAM' or 'CART'. In the 'GRAM' position, the momentary channel on switch, S1, only controls the channel on relay RL1. The fader micro-switch then controls remote stop and start functions. With the link plug in the 'CART' position, the channel audio is always on (controlled by the fader only) and switch S2, when pushed, fires the momentary remote start relay, RL3. The remote stop relay, RL2, is activated by closing the fader. Remote stop/start functions can be disabled by breaking the link between pins A13 and A14 on the remote interfacing connector socket. This link is normally made on all stereo channels except channel 9 where the link is routed via the auxiliary select pcb. Then only when aux button 4 is selected will this link be made, thus enabling remote controlling of auxiliary equipment from channel 9 position 4 only.

4.4 TELEPHONE CHANNEL INPUT

The mono line level signal is fed to an input attenuator which has a preset to enable the balancing of the two input phases for a good common mode rejection. The signal is then fed to a differential long tail pair configured by Q1 to Q4. The gain of the stage is configured by VR1. The outputs from the long tail pair are then fed to a differential amplifier formed by IC1A which has a single ended output. The prefade signal is taken from this point via 'PFL' switch S2 (when latched-in) and to the PFL mix bus via R16 and R17. The output of IC1A is also fed via the fader (when fitted) through the channel on relay, RL1 (when activated) to the main mix busses via R20 and R21. An output is also taken to the clean feed break point output socket situated on the connector panel. The channel on relay, RL1, is controlled by the momentary 'channel on' switch, S3. Logic control for this function is performed by IC3 which is powered from the 5.6 volt line derived from the 24 volt aux supply via R30, ZD1, C14, and C15.

IC2A is a virtual earth summing amplifier for the clean feed mix and the talkback output for this channel. Its output is then fed via C21, R26, and edge connector pin A11 to an output transformer mounted on the connector panel. If a cue programme is required as an auxiliary output from this module, the clean feed break points situated on the connector panel should be joined together. This then also adds the telephone channel output on to the clean feed mix bus via connector pins A5 and A6.

IC2B is a talkback amplifier which receives its input via 'talkback' switch S1 (when held in) and channel 1 microphone input via connector pin A31. Its output is then fed to the virtual earth summing amplifier IC2A.

When 'PFL' switch, S2, is latched in, the clean feed mix from connector pin A24 is broken away from IC2A. This enables communication with the caller via talkback switch S1 to be free of mixer output.

4.5 MASTER OUTPUT BOARD

IC1A and IC1B are virtual earth amplifiers for the summing of the left and right PFL mix inputs from connector pins A20 and A21. The outputs of which are fed directy to the monitor board via connector pins A3 and A4.

IC2A and IC2B are virtual earth amplifiers for the summing of the left and right main mix from connector pins A22 and A23. The outputs of these are fed to the 'limiter switch' S1. When the 'limiter switch' is delatched, the signals are fed via the mono switch S2 to the main output amplifiers IC3 and IC4. These amplifiers have trim controls (VR1 for right and VR2 for left) which enable the main left and right outputs to be adjusted. The signal is then fed to the monitor board via connector pins A1 and A2. When the limiter switch S1 is latched in, the outputs from IC2A and IC2B are fed to the limiter pcb sub-assembly via connector B. The output of the limiter is then fed to the mono switch S2 and the main output amplifiers IC3 and IC4. When the mono switch, S2, is latched in, the left and right signals are mixed by joining the junctions of R21 and R22 to the junctions of R28 and R29. This performs two functions: the input impedance to the output stages is reduced halving the summed signal; and a mono output is now available on both left and right outputs.

Auxiliary relays RL1 and RL2 are perminantly latched in when a mute circuit is operated. The mute circuit connects connector pin A34 to 0 volts turning transistor Q1 off. Transistor Q2 is then turned on allowing current to flow through the relays.

4.6 LIMITER

The audio paths of both channels are identical in working. So considering the left channel only, IC1A is a unity gain buffer amplifier which provides a low impedance source to the following stage. The signal is then attenuated by R2 and R3 to ensure the F.E.T. Q1 operates in its linear region. Limiting action is established by the attenuator formed by R4 and the F.E.T. source to drain resistance which is controlled by the potential difference between the gate and the source. VR1 sets the threshold of the F.E.T. operation. C1 and R5 introduce a small component of the audio signal to the gate to minimise distortion caused by the F.E.T. IC1B is a signal level amplifier which ensures unity gain between the input and output of the limiter stage below limiting threshold. The output signal is fed via C5 to the master output board and is also coupled via C6 to a precision rectifier formed by IC3A and IC3B.

The outputs from IC3B (left channel) and IC4A (right channel) are combined via D6 and D3 and fed to the time constant network formed by R39, R40, R41, C13, and C14. The output from this network then controls both channel F.E.T's simultaneously to maintain correct stereo balance. It is also fed to the limit indicator circuit situated on the meter motherboard via R44.

4.7 MONITOR BOARD

Connector B is the main signal input from the master output board. The 'PFL' input from connector B is routed to the auto PFL relays RL1 and RL2 and via RL2 to the monitor speaker auto PFL switch S4. When an input channel's 'PFL' switch is latched in, the PFL D.C. bus is connected to 0 volts and transistor Q1 is turned off by edge connector pin A33 being held low. Transistor Q2 is then turned on allowing current to flow through the auto PFL relays RL1 and RL2. When these relays are latched in, 'PFL' is routed through RL1 to connector K to the meter motherboard. When RL2 is latched in, 'PFL' is routed to monitor speaker auto 'PFL' switch, S4, and through RL3 (talkback relay) to the presenter headphone level control VR2A/B mounted on the front panel.

IC5A and IC5B are op amps which drive the headphone output stages Q11, Q12 (left), and Q13, Q14 (right) which are complimentary pairs that increase current capacity for driving the presenter headphones.

Monitor speaker auto 'PFL' switch, S4, allows the option for monitor speakers to either follow the preselected input via 'monitor source' switch S3 (auxiliary input via connector C or desk output via connector B) or PFL, when an input channel's 'PFL' switch is latched in. Audio routed through S4 is fed via the front panel mounted monitor speaker dual level control VR3A/B to unity gain buffer stages, IC3A and IC3B. The outputs of IC3A and IC3B are then fed through monitor speaker mute relay, RL4, to connector D and then to the monitor output transformers which are mounted on the connector panel.

Mute relay, RL4, is activated when a microphone channel is operated. This pulls the mute D.C. bus low (pin A34) which turns transistor Q5 off and transistor Q6 on allowing current to flow through RL4. The mute bus is also pulled low when a talkback send button, S1 or S2, is pressed. This is to prevent any audio feedback occurring.

Talkback relay, RL3, allows a total interrupt function of the presenter headphones. When talkback is received on connector G, the audio is routed through the summing amplifier, IC1A to the talkback relay, RL3. The relay switches the presenter headphones to the talkback source when the base of Q3 is held low. This transistor is then turned off turning transistor Q4 on allowing current to flow through the talkback relay, RL3.

Provision is made on the monitor board for sending talkback to two independent sources. This is achieved by pressing and holding the non-latching talkback switches, S1 and S2. The talkback audio is received from channel 1 microphone input via connector pin A31. It is amplified by IC1B, the output of which is fed to the two talkback switches. When a talkback switch is pressed, the audio and a closed circuit loop is routed to connector H so that the talkback relay of the receiving mixer is operated.

IC2A and IC2B are differential amplifiers receiving a monitor auxiliary input from connector C, the output of which is routed to the 'monitor source' switch, S3, which allows an auxiliary input such as 'off-air' to be monitored in place of desk output.

The main 'desk' output from the master output board is fed directly from connector B to: connector F from which it is connected to the main output transformers mounted on the connector panel; through 'auto PFL' relay, RL1, to connector K and on to the meter motherboard; to monitor source switch, S3; and to the front panel mounted guest headphone level control, VR1A/B. From VR1A/B it is fed to IC4A and IC4B to the guest headphone output amplifier.

4.8 THE METER MOTHERBOARD

The audio inputs are fed on to connector A, pins 1 and 2. They are then routed to the PPM drive cards and also to IC1A and IC1B which rectify the audio signal so that it can be used as a D.C. control voltage to drive IC1D. This is a voltage controlled lamp drive circuit designed to give gradual transition above threshold. Threshold is set by RV2. An audio output from the limiter is fed on to connector A, pin 5. This is rectified by IC1C before being fed to the voltage controlled lamp driver circuit, IC1D. Limiter threshold to turn the lamp driver circuit on, to show when limiting is taking place, is set by RV1.

4.9 POWER SUPPLY

The mixer requires two D.C. supplies for full operation, these are:

1. +/-16 volts D.C. regulated and smoothed for audio electronics. 1.5 amp per rail.

2. +24 volts D.C. regulated and smoothed for relays and indicators. 2.4 amp.

The power outlet is on a multiway connector situated on the rear of the frame. A power cable is supplied for connection to the mixer. The A.C. power requirement can be either 220-250 volts 45-65 Hz or 100-115 volts 45-65 Hz. To change voltage inputs, taps must be moved which are situated on the transformers mounted in the two Lambda power supplies. There are three fixed tapping points on the transformer: AC1, AC2, and D. There are also two floating wires: A (black) and B (grey) which are moved to change the voltage input setting.

This is done as follows:

1. 220-250 volts: The AC input is connected to AC1 (positive) and AC2 (negative). Floating wires, A and B, are both joined to D.

2. 100-115 volts: The AC input is connected to AC1 (positive) and AC2 (negative). Floating wire A is connected to AC1 and floating wire B is connected to AC2. Connection D is not used.

Circuit diagrams for the Lambda power supplies are supplied loose-leaf with this manual. For rear connector details of the multiway power connector, refer to figure 9.

CHAPTER 5

M A I N T E N A N C E

CHAPTER FIVE - MAINTENANCE

5.1 GENERAL ATTENTION

The equipment should be kept clean and dust free to minimise any possibility of malfunction due to dirty electrical contacts. Relays are of the enclosed reed type and so should prove reliable in operation over long periods without attention. The front panel control knobs are of a plastic material and care should be taken not to use chemical cleaners which might cause damage or stain the front panel.

5.2 BULB REPLACEMENT

All bulbs located in the switches are removed by placing a flat edged screwdriver under the side of the switch lens where a small notch is situated. When the screwdriver is twisted, the lens springs free. The lens cap is then removed along with the defuser and the ident if one is present. The bulb can then be removed and the new one fitted.

5.3 REMOVAL AND REPLACEMENT OF PRINTED CIRCUIT BOARDS

All internal wiring between pcb's and the connector panel uses molex plug-on type connectors. When removing any pcb's for maintenance purposes, no desoldering is necessary.

All printed circuit boards are held in place by fixing screws and by nuts situated under the control knobs. To remove the channel boards, it is not necessary to remove the connector panel. By removing cover plates A and B (refer to figure 1), the channel boards can be lowered through the bottom of the mixer. To remove a channel board, the top cap of its control knob should be carefully removed and the collet tightening screw loosened. When the knob is removed, the potentiometer fixing nut is exposed. When the nut and fixing screws are undone and the molex connectors unplugged, the channel board will be free of the mixer.

5.4 CONNECTOR PANEL (internal connections)

If, for any reason, the connector panel has to be removed, no desoldering is necessary as all the leads plug on to the pcb's. The 20-way auxiliary input connector is removed by undoing the two fixing screws, tilting the connector and its grey skirt up at an angle, and pushing them through the panel.

The table below should be referred to to enable correct reconnection of all the flying leads. For location of plug identifications and numbering, refer to figure diagrams as necessary as the pcb's are reconnected. In most cases pin 1 of the 7-way molex socket on the flying leads connects to pin 1 on the pcb's edge connectors. However, in some cases, higher numbers are used (e.g. 9 - 15) and the molex socket pin 1 always connects to the lower number (e.g. pin 1 to 9 and pin 7 to 15).

CHANNEL INPUTS	BOARD REFERENCE	REFER TO FIG N^{o}.
MIC 1	CH 1 pcb 1A - 7A	22
MIC 2	CH 2 pcb 1A - 7A	22
MIC 3 (via mic/line switch)	CH 3 pcb 1A - 7A	22
PHONE 4 (+ C.F. break point)	CH 4 pcb 1A - 7A	23
LINE 5	CH 5 pcb 1A - 7A	24
LINE 6	CH 6 pcb 1A - 7A	24
LINE 7	CH 7 pcb 1A - 7A	24
LINE 8	CH 8 pcb 1A - 7A	24
OFF-AIR INPUT	MONITOR pcb connector C	26
CLEAN FEED OUTPUT (via transformer)	CH 4 pcb 9A - 15A	23
MAIN OUTPUT (via transformers)	MONITOR pcb connector F	26
MONITOR OUTPUT (via transformers)	MONITOR pcb connector D	26

CHANNEL INPUTS	BOARD REFERENCE	REFER TO FIG N^{O}.
CH 5 REMOTE	CH 5 pcb 9A - 15A	24
CH 6 REMOTE	CH 6 pcb 9A - 15A	24
CH 7 REMOTE	CH 7 pcb 9A - 15A	24
CH 8 REMOTE	CH 8 pcb 9A - 15A	24
AUX REMOTE		
(4 wire socket)	CH 9 pcb 9A - 15A	24
(2 wire socket)	AUX pcb 9A - 15A	29
TALKBACK INPUT	MONITOR pcb connector G	26
TALKBACK OUTPUT	MONITOR pcb connector H	26
MUTING RELAYS	MASTER OUTPUT pcb 9A - 15A	25
HEADPHONES	MONITOR pcb connector E	26
POWER INPUT	MONITOR pcb connector J	26

5.5 INTER-CONNECTIONS BETWEEN PRINTED CIRCUIT BOARDS

There are a number of inter-connecting, plug-on leads other than those from the connector panel that interface with the pcb's. Their location and inter-connection are described as follows:

5.5.1 MAIN HARNESS

This is a 16-way insulation displacement plug-on harness that carries common signals and power to the pcb's. It has 10 16-way sockets to inter-connect all the channel input boards with the master output board and a 10-way socket that connects to the monitor board.

The 16-way harness plugs on to the master output board and the channel input boards and parallels up all their connector connections from A19 - A34. The 10-way socket is located at the left-hand end of the mixer (looking from the rear) and plugs on to the monitor board connector A (refer to figure 26). Careful attention should be made that the reference point A25 (left-hand end of connector A on the monitor board) is connected to A25 on the master output pcb. This is the point where the harness has been cut off short. The top six wires, A19 - A24, have been removed as they do not interface with the monitor board.

5.5.2 PLUG-ON LEADS

There are five separate plug-on leads that inter-connect from one pcb to another as a single function. Their connection details are described as follows:

1) Inter-Connection Between Master Output Board and Monitor Board:

   This is a 7-way harness which is wired using all white wires. Pin 1 on one end of the harness cross-connects to pin 7 on the other end. This is so that all the molex sockets in the mixer remain constant in the way they plug-on to the pcb's. The reference point on the sockets on both ends of the harness is still pin 1. Therefore, pin 1 on one end of the harness connects to monitor board connector B pin 1 (refer to figure 26) and pin 1 on the other end of the harness connects to master output board connector A pin 1 (refer to figure 25).

2) Inter-Connection Between Stereo CH9 and Aux Select Board:

   This is a 7-way harness that is wired using different coloured wires. Brown to pin 1 and mauve to pin 7 (follows resistor colour code). This harness does not cross-connect, therefore, pin 1 on one end of the harness connects to pin 1 on the other end. The reference point on the sockets is still pin 1. One end of the harness connects to pins 1A - 7A on CH9 stereo input pcb (refer to figure 24) and the other end connects to pins 1 - 7 on the aux select pcb (refer to figure 29).

3) Inter-Connection Between Monitor Board and Meter Motherboard:

   This is a 7-way harness that is wired as in (2) above. Using pin 1 as the reference point, it connects the monitor board connector K (refer to figure 26) to the meter motherboard connector A (refer to figure 27).

4) Inter-Connection Between Meter Motherboard and PPM Drive Cards:

   These connections are made by a pair of identical harnesses with 5-way sockets at each end. The reference point is pin 1. Connector B on the meter motherboard connects to the left PPM drive card and connector C to the right PPM drive card (refer to figure 27 for connector locations).

5.6 ALIGNMENT AND IN-SERVICE TRIMMING

Over a period of time, due to ageing of components, temperature drift, and vibration, a small drift of alignment in the audio path may become discernible. Provided that performance remains satisfactory in respect of noise, distortion, and band width, the following procedure can be used to re-align as necessary:

1) PPM Alignment:

Inject 0dBm 1kHz (PPM 4) into a stereo input channel. Adjust VR1 and VR2 on the master output board (refer to figure 25 for preset locations) until 0dBm appears on the mixer output (this reading should be taken on external test equipment). Align PPM cards for reading of 0dBm (PPM 4) on the meter according to alignment sheet provided with a PPM card that is used.

2) Stereo Input Channel and Master Output Board Alignment:

Inject 0dBm 1kHz into a stereo input channel and latch-in PFL switch. With the operator gain set at mid-range, adjust VR3 and VR5 (refer to figure 24 for preset locations) until PPM reads 0dBm (PPM 4). Depress PFL switch, open fader fully and adjust VR1, VR2 on the master output board (refer to figure 25 for preset locations) until PPM reads 0dB. The master output board has now been set-up and should need no further adjustment.

3) Remaining Stereo Input Boards Alignment:

Inject 0dBm 1kHz into each stereo channel in turn. With the operator gain set at mid-range and the PFL switch latched-in, adjust VR3 and VR5 on each board in turn until PPM reads 0dBm.

4) Input Balance Alignment on Stereo Channels:

This adjustment should be carried out with channel fader fully open and external test equipment connected to the output of the mixer. Pins 2 and 3 on the XLR input socket should then be shorted together and an unbalanced 0dBm 1kHz signal should be injected into the channel input between pins 2/3 and 1 (earth). The output of the mixer should then be measured on the external test equipment and VR1 or VR4 (refer to figure 24 for preset location), depending whether left or right being set up, should be adjusted until minimum output reading is obtained, typically -70dBm.

5) Input Balance Alignment on Telephone Channel:

The same procedure as above should be followed using VR2 to adjust for minimum output (refer to figure 23 for preset location).

6) Microphone Input Level Adjustment:

Inject 1kHz signal level as required for 0dBm reading on PPM with microphone gain control at centre position and PFL selected. Deselect PFL and with fader fully open, adjust VR3 for 0dBm reading on PPM (refer to figure 22 for preset location).

7) Microphone Input Distortion Adjustment:

Inject 1kHz signal level as required for 0dBm reading on PPM with microphone gain control at centre position and fader fully open. With distortion meter connected to the output of the mixer, adjust VR2 for minimum distortion (refer to figure 22 for preset location).

8) AGC Card Alignment:

Inject 1kHz signal level as required for 0dBm reading on PPM with microphone gain control at centre position, fader fully open, and AGC switched out. Switch AGC in and adjust RV1 (refer to figure 28 for preset location) on AGC board until PPM reads 0dBm. Increase input signal level by 20dB and adjust RV2 on AGC card until PPM reads 0dBm. Reduce input signal level by 40dB and check that PPM reads 0dB. If not, re-adjust RV2. Increase signal level by 20dB and repeat above from when AGC was first switched in until points correspondence to approximately +/-1dB.

9) Limiter Adjustment:

Inject 0dBm 1kHz signal into left and right inputs of a stereo channel. Adjust gain control for an output level of +5dBm. Switch limiter in and increase signal level to right-hand channel by 10dB. Adjust VR2 (refer to figure 28 for preset location) so that right output level is +6dBm. Adjust VR1 for left output level to read -4dBm. Restore right-hand channel input level to +5dBm and increase signal level to left-hand channel by 10dB. Check that right output level is -4dBm. If not, re-adjust VR2. Check that left output level is +6dBm. If not, re-adjust VR1. Repeat complete above procedure until the best results are obtained.

10) Overload Lamp (limiter switched out):

Inject 1kHz signal into any input channel until PPM reads +6dBm. Adjust VR2 on meter motherboard (refer to figure 27 for preset location) until lamp has just started to glow.

11) Overload Lamp (limiter switched in):

Inject 1kHz signal into any channel, slowly increasing (watching PPM) until limiter is just beginning to operate. Adjust VR1 on meter motherboard until lamp has just started to glow.