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## SCENARIO

Time was when it became necessary to update the ubiquitous AD145 PICO mixer. Our survey showed two main requirements: as well as Audio Developments' build and audio qualities, narrower and shorter modules were required, and the simple but effective equaliser from PICO had to be retained. By this time, 4-track recording equipment had begun to trickle into the marketplace (Nagra D etc), so we decided to give the new mixer four outputs. Because of the lower noise-floor of modern digital recorders, the level of input signals may be reduced. We have taken advantage of this fact and lowered the slope ratio of the limiters to 7:1. By so doing, not only is signal distortion greatly reduced, but also the artefacts associated with limiters become much less noticeable or objectionable. Also, sixty years after the event, sound engineers have come to realise the potential of Blumlein's M-S techniques; to this end, facilities have been incorporated to take advantage of these techniques. The result is AD146 - a four-output mixer.

This was followed by AD148 - edit mixer. Based on AD146, two comprehensive left and right monitor modules have been added for editing purposes. If a mic/line module is included for commentary or voice-over purposes, its input-gain switch can be changed to a potentiometer, and its gain structure changed to ensure consistent level matching with a fixed, mechanical point of reference. Unfortunately, this way of working does reduce headroom and also compromises a mixer's noise performance.

After AD146/AD148 it was back to the drawing board. Our customers were still demanding a two-output mixer - as a true replacement for the PICO, and with the PICO's simplicity. Despite past assurances to the contrary, T powering is still required. And could we incorporate auxiliaries? And could we possibly bring it to market at 'entry level'? We have, and we have and that's MERCURY (AD147).

AD149 completes the 140 series of mixers, and has a repertoire of party tricks not to be found in any other mixer - not even for 'ready money'. The design team took, as its starting point, our list of all the ideas and suggestions presented to us over the past few years. Many of these requests came from film-sound recordists - a sub-set of recordists we have unintentionally neglected in the past.

Since the days of R & D for AD146, channel insert points suddenly became de rigueur. It transpires that many sound engineers have experienced the power of FLEX-EQ, our industrial-strength equaliser, and wish to be able to use it when making original recordings.

Being latter-day converts, we have included circuitry to take full advantage of all M-S techniques - even shuffling - in both production and post-production.

AD149 is Audio Developments' tribute to, and celebration of the genius of Alan Blumlein.

We appeal to all AD149 users to break with tradition and, just on this occasion, study the manual to reveal all the mixer's secrets.  
THIS IS ESSENTIAL.

Well - we HAD every intention that AD149 would complete the 140 series.  
We'd reckoned, however, without the persistence - nay, insistence - of our customers for a mains-operated version of the AD146 with four auxiliaries ...  
we've called it AD144.

## INTRODUCTION

### Unpacking

If there are any signs of damage to the outside of the carton, please notify us or your supplier immediately, regardless of the unit's apparent physical condition. This is in case a claim has to be made at a later date because of previously undetected transit damage. The packaging material should not be discarded until the mixer has been acceptance tested and a suitable transit/storage case is available for secure, safe storage.

### Visual Inspection

Identification - please make a separate note of the serial number for your own capital equipment records. Ensure that it agrees with the number on the invoice/packing note. The serial number label is on the back cover, adjacent to the battery compartment.

Configuration - check that the correct number and combination of input modules have been installed. If, for any reason, you wish to change the positions of the modules, do not do so until after completion of any acceptance tests. Any tests then made will be to a known configuration and can be compared with our factory records.

Temperature - check the meter glasses for condensation. If the package has been in transit during cold weather, leave the mixer for at least 12 hours to allow it to return to normal room temperature. Any measurements or subjective tests then made, will be to a known temperature reference.

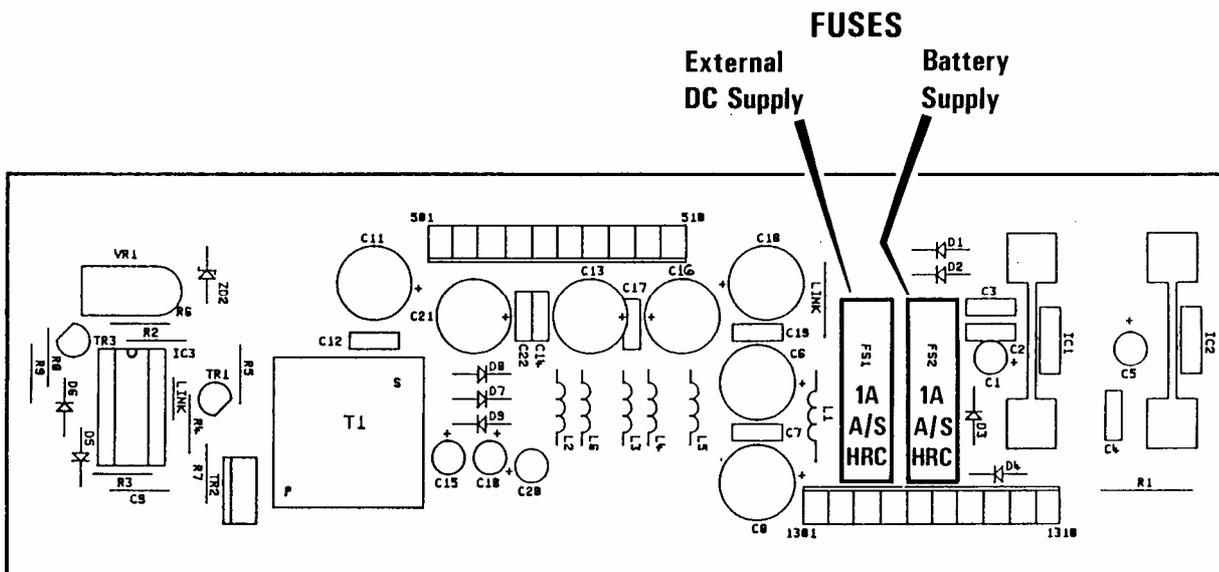
### Specific Points

Battery compartment - the mixer has an integral battery compartment to accept 10 size-C cells, and is formed as part of the bottom transversal extrusion. This helps to lower the centre of gravity, as well as adding to the rigidity of the frame.

DC-DC converter - is mounted on the top side of the battery housing and is underneath the fader section of the output and last four input modules.

In order to facilitate the testing and calibration of all modules, a set of two extender modules is available from the factory.

Fuses - to protect the mixer and internal power supply are mounted on the power supply/converter board. Access is gained by removing the output module. 20mm, 1A ANTI-SURGE HRC fuses are used - one for the internal battery power supply and one for the external DC supply.



Until the mixer has been acceptance tested we strongly recommend that modules are not removed for adjustment. If adjustments are necessary, refer to the section on adjustments and calibration. All customer functions are external, except the following.

Limiter threshold - is set at the factory at +8dBu (PPM 6), but an internal preset potentiometer allows adjustment to any other level above 0dBu. The limiters are to be found on the left-hand PCB of the output module. Presets are also available for adjusting limiter attack and release times.

Reverse-talkback output - is set at the factory to provide a mono mix of monitor 1 left and right outputs, rather than mute, when the main talkback switch is released. This function can be disabled by the DIL switch to be found on the left-hand PCB of the monitor module.

Line-up tone - at 1kHz and 10kHz is calibrated on the meters to PPM4, giving a line output level of 0dBu with L/R and AUX 1/2 controls set to maximum. NOTE - PPM4 refers to the mark on a BBC PPM Scale. Other meter scales have a different mark at 0dBu level. Alternative calibration levels are possible via the internal preset.

RH-output ident - as an alternative to continuous tone on all main outputs; B & D outputs can be pulsed with tone for 3 seconds.

Microphone powering - the mixer will remain unconditionally stable if the powering on unterminated input channels is switched off - this also improves the noise performance and crosstalk. Powering - 48v phantom or 12v tonader - may be selected before or after the microphone is connected to the module.

Module fix screws - Hexagon head screws are used to fix the modules and back cover. The size being 1.5mm HEX A/F with a 2.5M thread.

If it becomes necessary to remove modules or back cover it is strongly advised using a good quality hexagonal head driver.

Use of a screwdriver, however desperate, is not recommended.

Surfaces - every working surface is covered by a hard plastic membrane which is printed on the reverse side. These surfaces may be cleaned by most solvents, and without fear of the legends being erased.

Connectors - for convenience, the connector panel is labelled for reading from above. All connectors are in line with their corresponding module.

Every input module is mechanically shielded against radio-frequency interference.

#### Abbreviations

PPM	peak programme meter
VU	volume unit
VR	variable resistor
X-Y	coincident stereo-microphone
A-B	spaced-apart stereo-microphone
M-S	middle-side; techniques and microphone described by A D Blumlein
PCB	printed circuit board
HPF	high-pass filter

Finally, may we draw your attention to our range of portable, battery-operated audio-toys called PORT-A-FLEX (AD066) which will complement your new mixer.

No 1 - broadcast quality compressor/limiter. Nos 3, 7, 8, 10 - distribution amplifiers for microphone- and line-level signals. No 13 - the aforementioned FLEX-EQ in its portable guise.

We wish you many trouble-free hours of use from your mixer. As a company, we are fully committed to BS EN ISO 9001. Should you have any problems or require any further information on FLEX-EQ or the M-S microphone technique, please do not hesitate to contact us on 01543 375351 or by fax on 01543 361051.

# **WARNING**

## **IMPORTANT SAFETY INSTRUCTIONS**

The user of electrical products must be familiar with their potential dangers, and fundamental precautions must always be taken. Please read the following text carefully.

Power supply units manufactured by Audio Developments Ltd are not user serviceable. There are no user-serviceable parts associated with any such power supply unit.

### **THE OUTER COVERS MUST NOT BE REMOVED**

Such a power supply unit is solely for use with audio mixers and sound processors - hereafter called the equipment - manufactured by Audio Developments Ltd. Always use a cord set accepted by a National Approved Body.

**EARTHING/GROUNDING:** When using an external power supply unit that is connected to the mains supply to drive the mixer it must be **CONNECTED TO EARTH**.

In certain types of malfunction or breakdown, earthing provides a path of least resistance for electric current and considerably reduces the risk of electric shock.

**DANGER:** Incorrect connection of the equipment grounding/earthing conductor can result in the risk of electric shock. Where possible obtain a pre-wired mains lead from a reputable supplier with the correctly fitted mains connector for the type of mains outlet in use; otherwise, one correctly wired and checked by a qualified electrical engineer. If your mains lead is not suitable for the mains outlet, have the correct plug fitted by qualified personnel.

The **MAINS PLUG** of this equipment is the primary disconnect device. Therefore, in the final application, ensure it remains close to the equipment and easily accessible.

## POWERING

The mixer may be powered from either internal cells or an external DC power source.

The integral battery compartment requires a total of 10 size-C cells. Access is gained via a captive lid which is retained by two, 90-degree-turn buckles. The lid hinges outwards 45 degrees from the back panel. When installing new cells, the row nearest the hinge should be fitted first.

Either conventional dry, or rechargeable nickel-cadmium cells may be used. NICADs may be recharged in situ through the 4-pin POWER IN connector. (The circuit for recharging is already incorporated within the mixer.) A voltage in the range +15V to +24V DC @ 250mA is required on PIN 2 of the POWER IN XLR.

When driving the mixer from an external power source, PIN 1 is the 0v connection and a voltage in the range +12V to +15V DC should be supplied to PIN 4. The power source should be capable of delivering approximately 400mA - allowing some capacity for phantom powering.

If an external power supply unit (PSU) is to drive the mixer and simultaneously charge a set of NICADs, a current capability of at least 650mA is required. It is poor practice to run a PSU at its limit, therefore we recommend a minimum of 750mA. Audio Developments AD100-09 PSU is a suitable unit.

**WARNING:** When NOT using the PSU (AD100-09) supplied for the mixer, ensure your 4-pin XLR is correctly wired to match the POWER IN connector. Failure to do so may result in the breakdown of the internal DC-DC converter. Make this check even if using a PSU which may have been supplied to you in the past, eg AD100-06.

AD100-05 PSU is NOT suitable for use with an AD146 mixer and must not be used.

## TECHNICAL SPECIFICATION - ELECTRICAL

REFERENCE	0dB = 775mV at 1kHz unless otherwise stated							
MAX GAIN	MIC	85	75	65	55	45	35	dB
	LINE	50	40	30	20	10	0	dB
MAX INPUT LEVEL	MIC	-42	-32	-22	-12	-2	+8	dB
	LINE	-7	+3	+13	+23	+33	+43	dB
	RETURNS						+24	dB
INPUT IMPEDANCE	MIC	>1k5R						
	LINE	>10kR						
	RETURNS	>100kR						
MIC POWERING	+48V DC and +12V DC PHANTOM POWER +12V TONADER (to order)							
MAX OUTPUT	+23.5dBm (600R LOAD) - A,B,C,D; TRANSFORMER BALANCED CLEAN FEED/CHANNEL OUT; ELECTRONICALLY BALANCED +18dBm (600R LOAD) - MON 1 L&R; XLR MON 2; UNBALANCED STEREO JACK							
OUTPUT IMPEDANCE	<60R A,B,C,D; CLEAN FEED/CH OUT							
	<20R MON 1; MON 2 (HEADPHONES JACK)							
FREQUENCY RESPONSE	+0; -1dB 20Hz to 20kHz MAIN SIGNAL PATHS							
	+0; -1.5dB 20Hz to 20kHz MONITOR SIGNAL PATHS							
EQUALISATION	LF; SHELIVING ±10dB @ 100Hz VARIABLE TURNOVER							
	MAX ±15dB @ 30 Hz							
	MF; PEAK/DIP ±15dB @ 2k8Hz Q = 1.2 RECIPROCAL							
	HF; SHELIVING ±10dB @ 10kHz VARIABLE SLOPE MAX ±15dB @ 30kHz							
HIGH-PASS FILTER	-3dB @ 90Hz and 150Hz, 12dB/OCTAVE SLOPE							
HARMONIC DISTORTION	<0.03% @ 1kHz 0dBm OUTPUT							
	<0.15% @ 40Hz to 20kHz +15dBm OUTPUT							
CROSSTALK	<-70dB 40Hz to 15kHz INTERCHANNEL & INTERGROUP							
	<-55dB 40Hz to 15kHz BETWEEN L&R ON STEREO CHANNEL							

NOISE MIC <-126dB 20Hz to 20kHz 200R SOURCE  
(WITH RESPECT TO THE INPUT)  
LINE >70dB SIGNAL-TO-NOISE RATIO 20Hz to 20kHz

LIMITER THRESHOLD +8dB  
RATIO 7 : 1  
ATTACK 2ms (SET AT FACTORY)  
(VARIABLE VIA INTERNAL PRESET)  
RELEASE 250ms (SET AT FACTORY)  
(VARIABLE VIA INTERNAL PRESET)

POWERING EXTERNAL AD100-09 POWER SUPPLY UNIT  
+14V DC, 4-PIN XLR  
3W (WITHOUT MICROPHONE POWERING)

## TECHNICAL SPECIFICATION - MECHANICAL

AD146 is supplied with 6, 8, 10 or 12 input modules. A separate meter bridge fitted with 4 VU or PPM meters is supplied to read outputs A,B,C and D. The mixer's meters can then be dedicated to monitor outputs.

	6 I/P	8 I/P	10 I/P	12 I/P
A	350	411	472	533
B	325	386	447	508

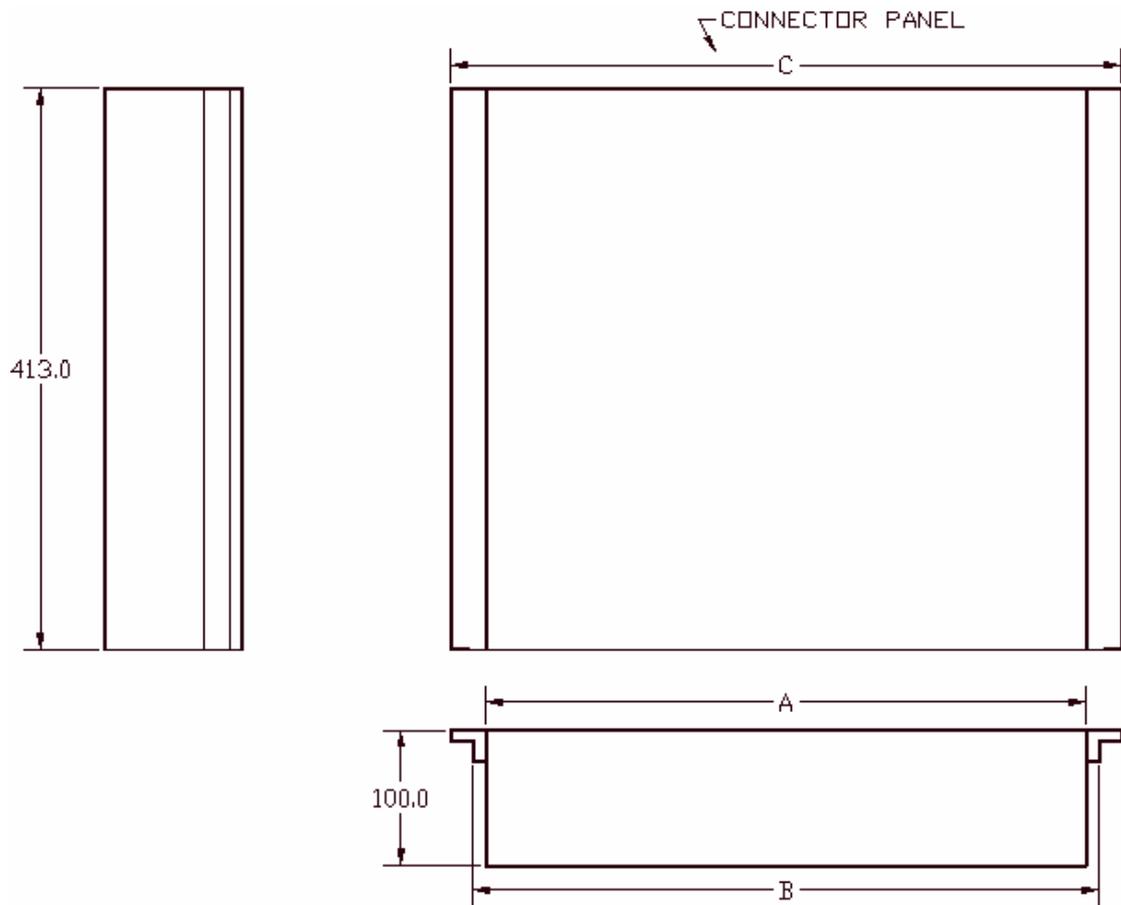
### DIMENSIONS IN MILLIMETRES

### WEIGHT

6 I/P	8 I/P	10 I/P	12 I/P
8.2	9.7	11.2	12.7

### WEIGHTS IN KILOGRAMS

SIZE



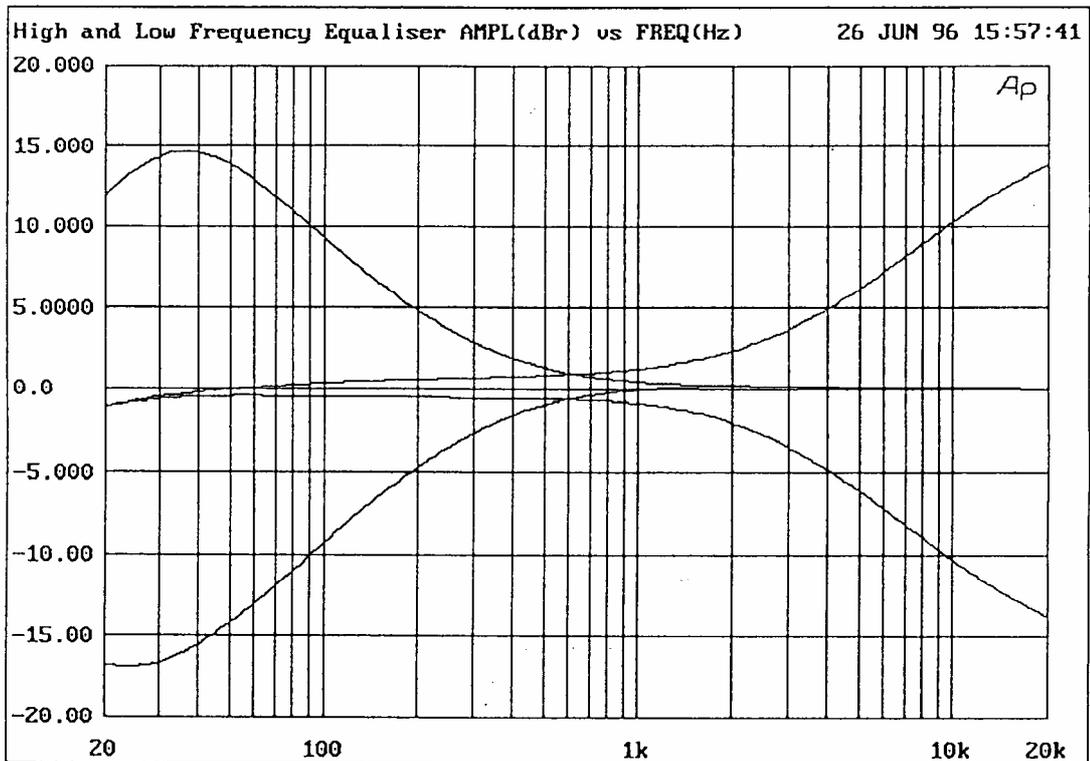
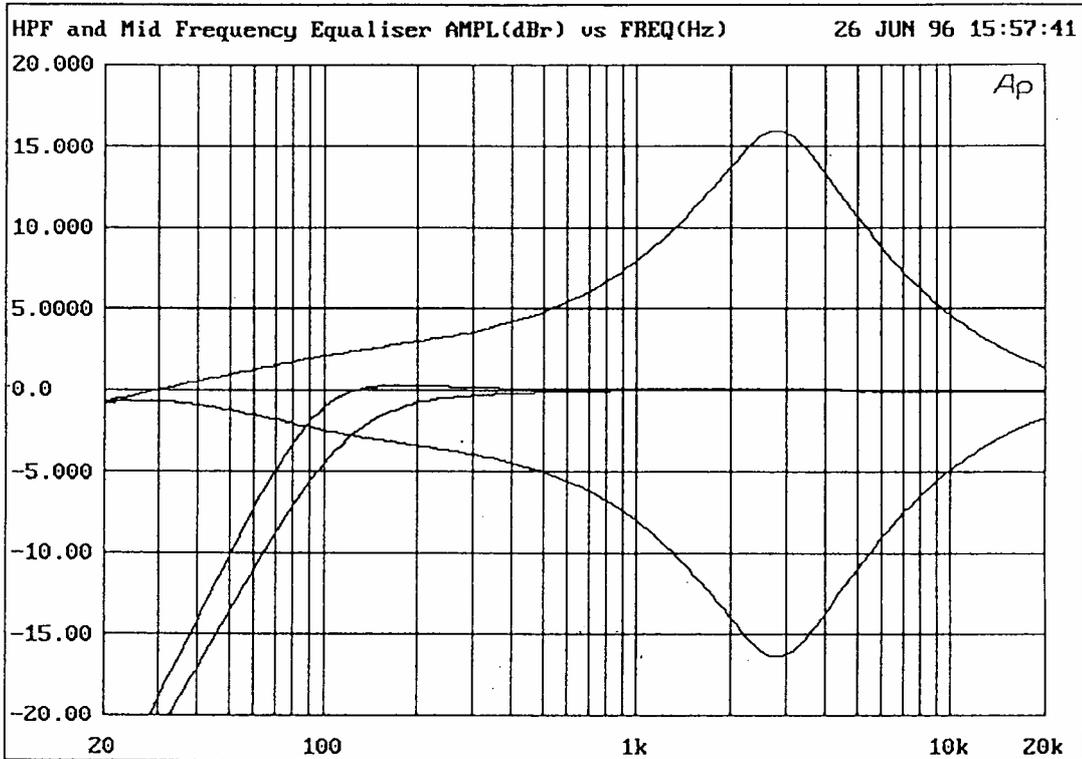
DROP THROUGH VERSION OF MOUNTING

	6 I/P	8 I/P	10 I/P	12 I/P
A	318	379	440	501
B	324	385	446	507
C	369	430	491	552

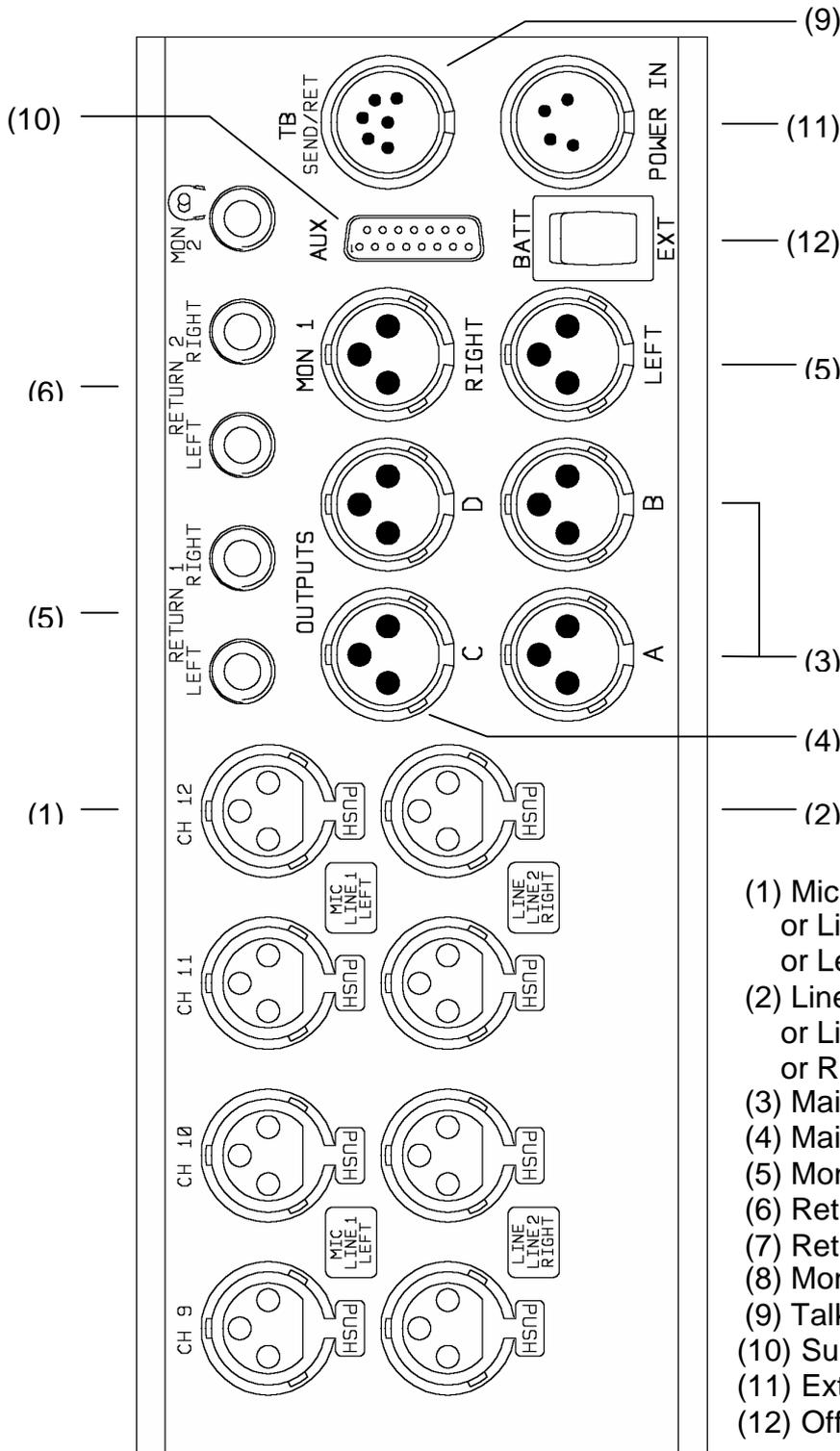
MEASUREMENTS IN MILLIMETRES

WEIGHT AS PREVIOUS

# EQUALISATION CURVES



# CONNECTOR PANEL



- (1) Microphone input or Line 1 of mono input or Left of stereo input
- (2) Line input or Line 2 of mono input or Right of stereo input
- (3) Main A-B outputs
- (4) Main C-D outputs
- (5) Monitor 1 Left-Right outputs
- (6) Return 1 Left-Right jack sockets
- (7) Return 2 Left-Right jack sockets
- (8) Monitor 2 output jack socket
- (9) Talk send & return
- (10) Subsidiary connector
- (11) External DC power input
- (12) Off/On switch (external & internal)

All input and output impedances are to be found in the TECHNICAL SPECIFICATION.

All inputs to, and outputs from AD146 are to be found on the connector panel.

Module connector (1) accepts	balanced microphones	-	mic/line
	balanced line 1	-	mono line
	balanced left	-	stereo line

Module connector (2) accepts	balanced line	-	mic/line
	balanced line 2	-	mono line
	balanced right	-	stereo line

XLR	Pin 1	Shield
	Pin 2	Signal +
	Pin 3	Signal -

In the case of an unbalanced line-input, pins 1 & 3 should be connected.

If a post-fader, balanced channel output (clean feed) has been included, the signal appears on a jack in the centre of XLR (2).

Main outputs (3) → (6) are transformer-balanced and are at line-level. When driving unbalanced loads, the mixer outputs may be unbalanced by connecting pins 1 & 3. This will not lead to a loss of output level.

XLR	Pin 1	Shield
	Pin 2	Signal +
	Pin 3	Signal -

The two electronically-balanced stereo-returns enter the mixer on pairs of standard, 'A' type stereo jacks - left & right (7) and (8).

BALANCED JACK		STEREO UNBALANCED JACK	
Tip	Signal +	Tip	Left Signal
Ring	Signal -	Ring	Right Signal
Sleeve	Shield	Sleeve	Shield

The monitor 2 (headphones) output appears on an 'A' type stereo jack (9) and is capable of driving 25R at 0dB.

A three-position rocker switch BATT/EXT (12) selects either internal batteries or an external DC power source. Power to the mixer is confirmed by the MIX ON LED on the output module. The LED flashes when the supply voltage falls below the safe operating level of 10V.

External powering of the mixer is via a 4-pin XLR (13).

XLR	Pin 1	0V	Pin 3	15V - 24V DC
	Pin 2	Charge	Pin 4	12V - 15V DC

A suitable external power supply is Audio Developments' AD100-09, but any external DC source must be capable of delivering 750mA at 12V.

Note - When the chassis size exceeds 12 inputs, or when specified by the customer, a

5 pin XLR (13) is fitted to accept external powering. The internal DC-DC Converter is omitted. A suitable external power supply is Audio Developments AD 100-10.

XLR	Pin 1	0V	Pin 4	+ 48V DC
	Pin 2	+ 12V DC	Pin 5	NC
	Pin 3	+ 12V DC		

'D' connector (14) provides connection to an optional meter bridge - in which case, the mixer's meters should be routed to MON.

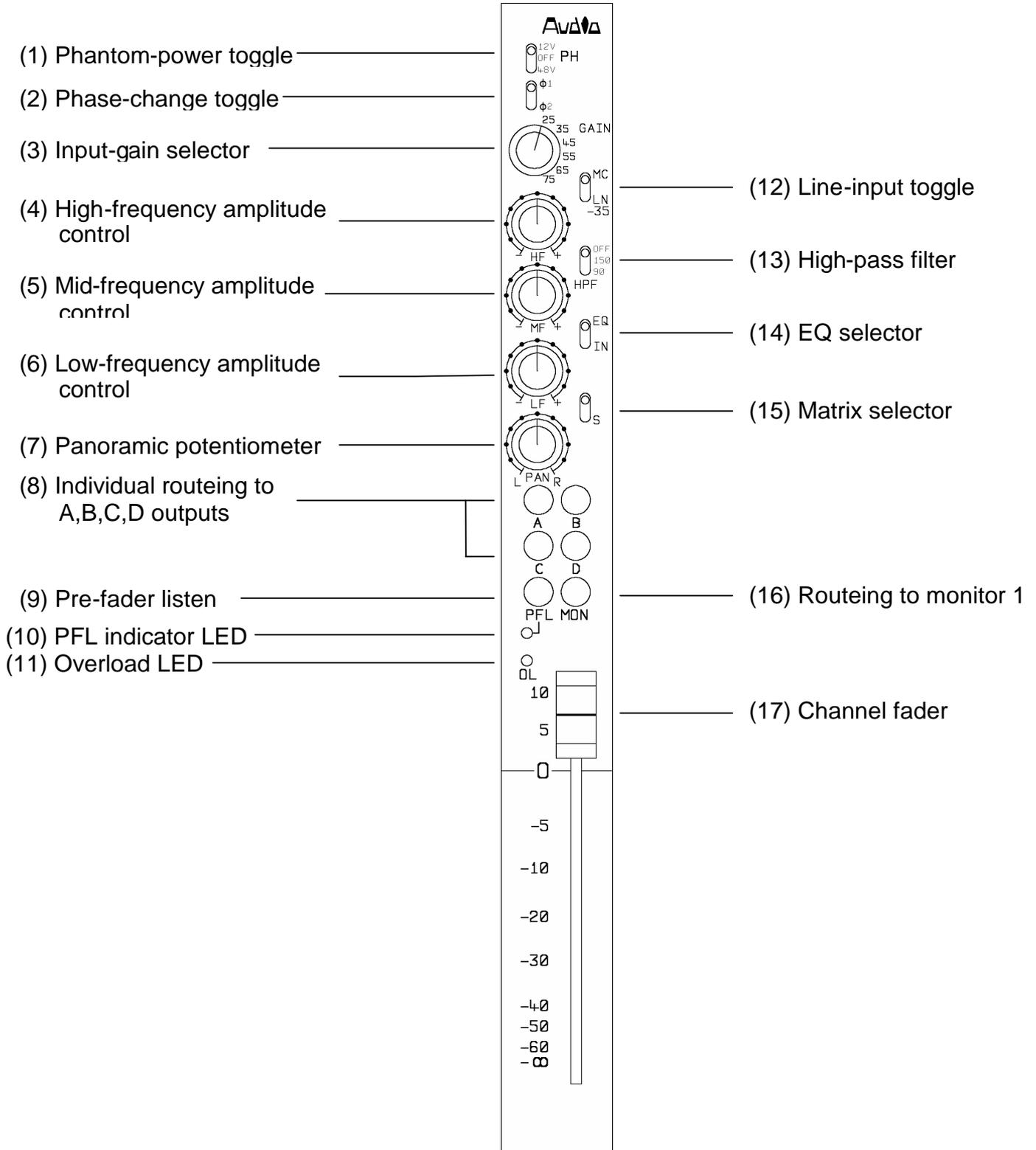
CONV			IDC	CONV			IDC
Pin	1	(1)	Chassis	Pin	8	(15)	+V voltage rail
	2	(3)	Spare		9	(2)	0V
	3	(5)	Spare		10	(4)	PFL - left output
	4	(7)	Spare		11	(6)	PFL - right output
	5	(9)	Spare		12	(8)	Output D
	6	(11)	PFL control		13	(10)	Output C
	7	(13)	-V voltage rail		14	(12)	Output B
					15	(14)	Output A

The 6-pin XLR connector (15) carries all signals to and from an outstation (boom operator, director etc). The return appears only on the headphones output.

XLR	Pin 1	Audio Ground	Pin 4	Ret + ] Balanced return
	Pin 2	Send +	Pin 5	Ret - ] at line-level
	Pin 3	Send -	Pin 6	Control

The talkback-send output has a low output-impedance with a capability of driving headphones of 25 ohms impedance or greater. Refer to MONITOR MODULE section for a full operation description of this output/return.

MICROPHONE/LINE MODULE



The transformer-balanced, ultra-low-noise microphone amplifier does not require an input attenuator as the available 50dB gain change is achieved with a 6-position rotary switch (3) acting on the feedback alone - in 10dB steps. This has the advantage of not degrading signal-to-noise performance as the gain is reduced. (The rotary switch may be replaced with a potentiometer for continuously-variable gain control.)

A DC voltage is available for phantom powering, either at 48v or 12v, condenser microphones (1). If the T (tonader) powering option has been included in the mixer, it will be selected by switch (1) to 12v.

The line input is isolated from the microphone power and no damage can be done to external equipment if microphone power remains selected when the channel is switched to accept a line input (12). Such is the isolation between signal paths, a microphone input (Conn 1) and a line input (Conn 2) may simultaneously be connected to a channel.

With line selected (12), input gain switch (3) at 35 and channel fader (17) at 0dB, the mixer has unity gain, with 10dB gain in reserve. (All output faders must be set at 0dB.)

Phase change (2) follows the mic/line selector and is therefore active on both inputs.  $\phi 1$  is the normal position.

The high-pass filter (13) is also active on both inputs and precedes the input transformer. In this position, the filter protects the transformer against saturation and overload caused by excessive low-frequency content in the input signal. The high-pass filter has a slope of 12dB/octave and is -3dB at 150Hz in position 2 and -3dB at 90Hz in position 3. The filter is independent of the equaliser.

The EQ selector switch (14) routes the signal through the equaliser section. The high-frequency control (4) has an amplitude of  $\pm 10$ dB at 10kHz, with variable slope. The mid-frequency control (5) has an amplitude of  $\pm 15$ dB at 2.8kHz, with a proportional-Q of 1.2. The low-frequency control (6) has an amplitude of  $\pm 10$ dB at 100Hz, with variable turnover. The EQ switch auditions the signal pre and post equalisation without disturbing chosen settings.

If the balanced, post-fader channel output (clean feed) option has been included, the signal will appear on a stereo jack in the centre of (XLR) CONN 2. Alternatively, it may be chosen to replace LINE input on CONN 2. The components associated with the clean feed circuitry appear on the sub-board. The DIL switch allows the module to drive unbalanced loads - without a reduction in signal level. Refer to TECHNICAL LIBRARY.

The overload LED (11) illuminates 3dB before clipping at the input to the channel fader (equaliser output).

S-switch (15) - creates a matrix amplifier across an adjacent pair of input modules. Refer to M-S NOTES.

The panoramic potentiometer (7) routes the signal proportionately between A and/or C outputs (L) and B and/or D outputs (R). Routing switches (8) are independent of each other and also act as channel mutes.

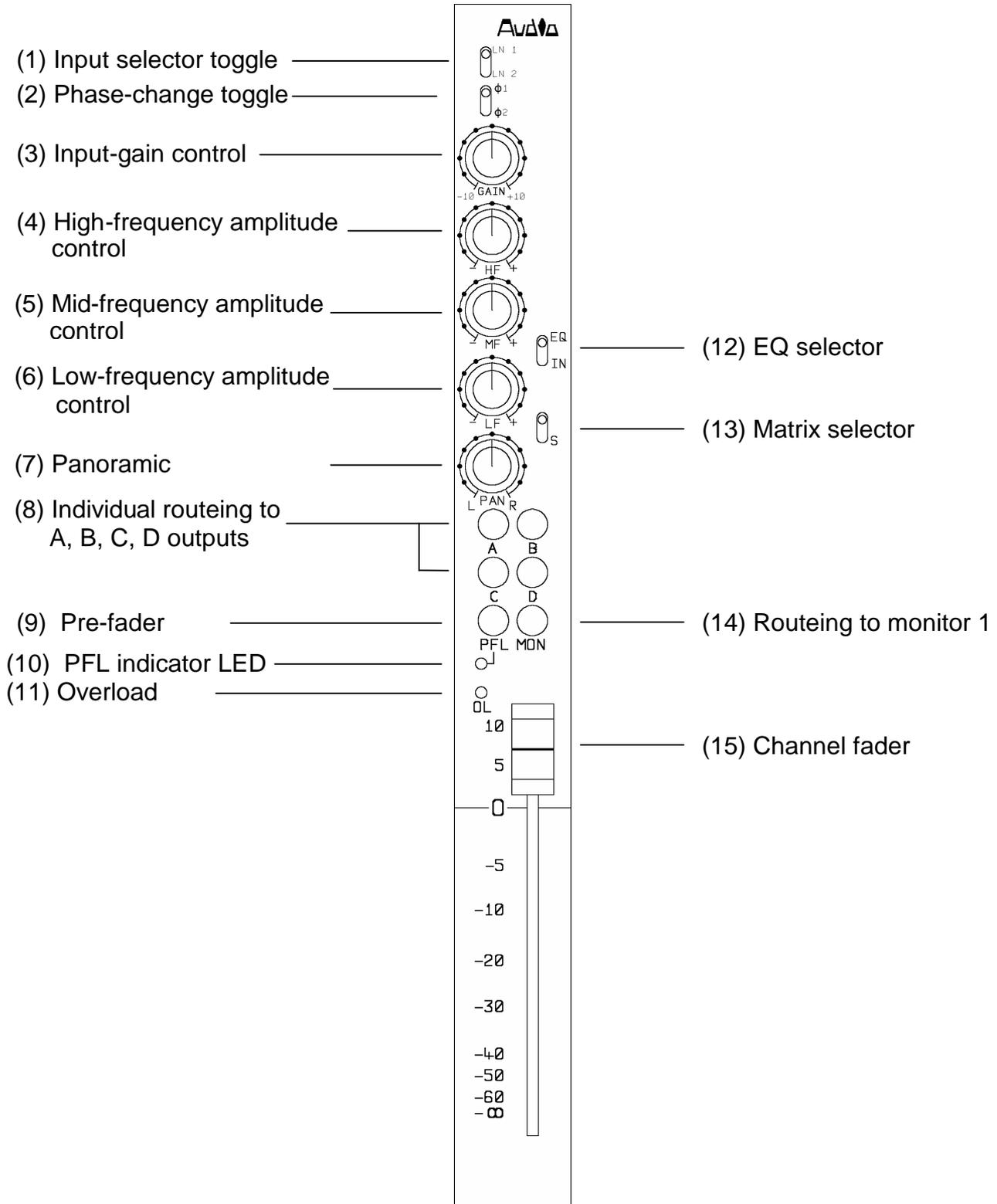
The monitor 1 path is the true output from the module (independent of any other routing) and is derived from the output of the panpot - and S-switch (15) if selected. MON (16) routes the channel signal to monitor 1 module and is auditioned when CH (1) is selected. This monitor 1 output also automatically appears on monitor 2 (headphones) output. MON can also be used as an AFL (after-fader listen) function, ie to monitor the mono output of the module. In this case, select MON (16), and CH (1) and MONO (14) on the monitor module. SIP (solo-in-place) usage of MON is outlined in the MONITOR MODULE section.

PFL (9) routes the signal at the fader input to monitor 2, ie the headphones-output jack. PFL overrides any other signal appearing on monitor 2.

The Penny & Giles fader (17) is calibrated 10dB down from its fully open position, allowing the operator to work with 10dB of gain in reserve. Faders on adjacent modules can be coupled for stereo operation by the use of standard ganging clips. (The fader may be calibrated at its fully open position if desired - and the scale changed accordingly.)

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MONO LINE MODULE



For optimum performance at line level, this module employs a double-balanced, instrumentation-grade input amplifier (3 x ICs) rather than the inferior differentially-balanced amplifier (1 x IC).

Isolation between inputs (Line 1 and Line 2) is so high, they may remain connected to the mixer simultaneously. Line 2 input is selected by switch (1).

Phase change (2) precedes the radio-frequency, interference-suppression circuitry. Input gain is switched in 2dB steps allowing up to 10dB of gain or attenuation (3). (The switch may be replaced with a potentiometer for continuously-variable control of input gain.)

The equaliser is identical with that in the microphone/line module and is engaged by EQ (12).

The clean feed option is also identical with that in the microphone/line module.

The panoramic potentiometer (7) routes the signal proportionately between A and/or C outputs (L) and B and/or D outputs (R). Routing switches (8) are independent of each other and also act as channel mutes.

The overload LED (11) illuminates 3dB before clipping at the input to the channel fader (equaliser output). Should the overload LED operate, check for excessive equalisation before reducing the input gain.

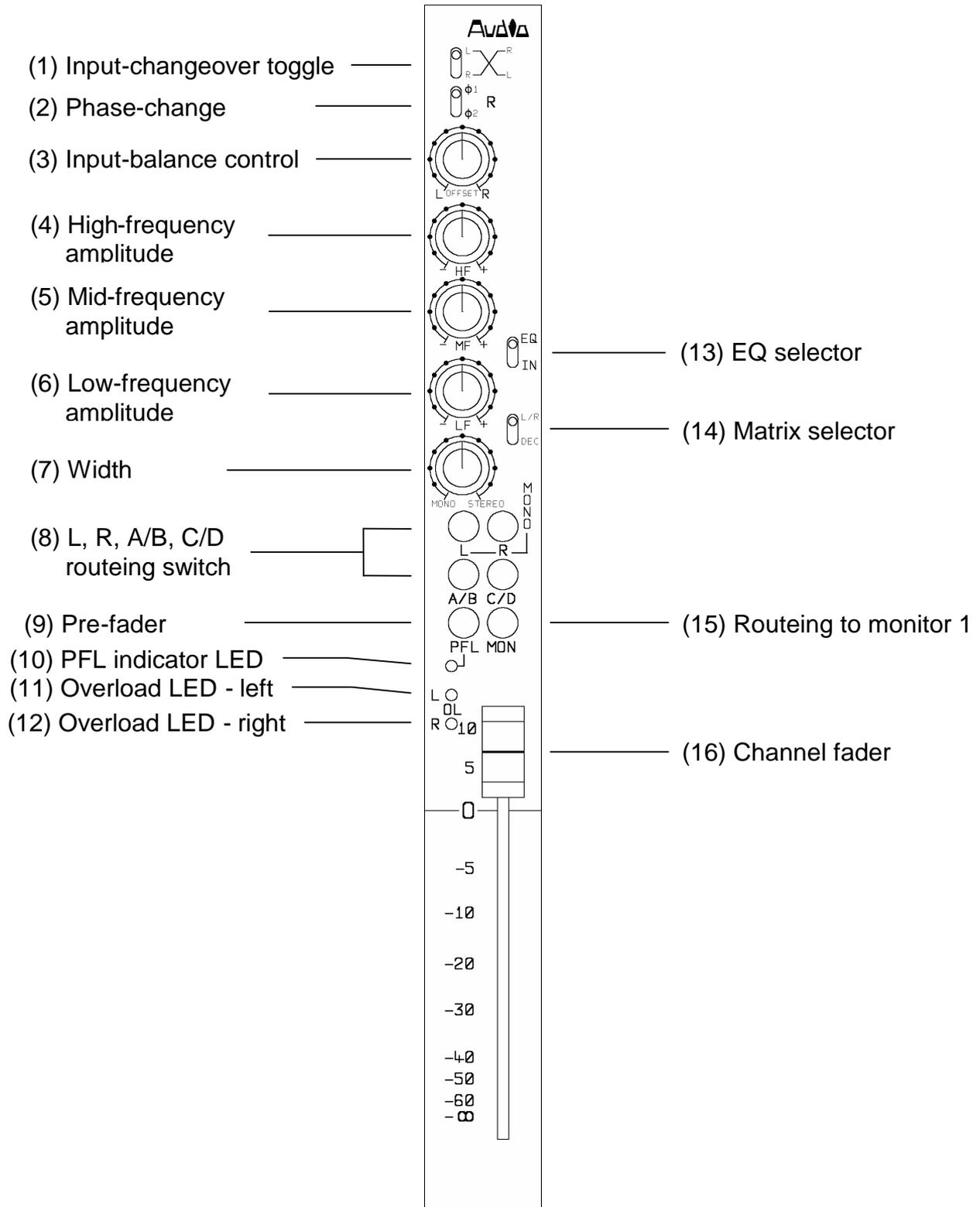
S-switch (13) creates a matrix amplifier across an adjacent pair of input modules. Refer to M-S NOTES.

PFL (9) - with indicator (10) - routes the signal at the fader input to monitor 2, ie the headphones-output jack. PFL overrides any other signal appearing on monitor 2.

MON (14) routes the true output from the module to the monitor module. This signal is independent of any other routing and is derived from the output of the panpot - and S-switch (13) if selected - and therefore has L and R components. This monitor signal is auditioned by selecting CH (1) on the monitor module.

The Penny & Giles fader (15) is calibrated 10dB down from its fully open position, allowing the operator to work with 10dB of gain in reserve. Faders on adjacent modules can be coupled for stereo operation by the use of standard ganging clips. (The fader may be calibrated at its fully open position if desired - and the scale changed accordingly.)

# STEREO LINE MODULE



For optimum performance at line level, the stereo module employs RFI suppressed, double-balanced, instrumentation-grade input amplifiers.

The input-changeover toggle (1) interchanges the inputs to connectors 1 & 2 without having to replug. The phase-change toggle (2) changes the phase of the signal on connector 2 and is unaffected by the position of toggle (1).

The module's input stages have fixed, unity gain; nevertheless it is possible to rebalance the stereo input-signal with the OFFSET control (3);  $\pm 10\text{dB}$  variation is available.

Equalisers identical with those of the microphone/line module may be inserted in both left and right signal paths via EQ selector (13). The high-, mid- and low-frequency amplitude controls are ganged, operating simultaneously on both equalisers.

When an input is in the M-S domain, the width may be altered by the OFFSET control (3) and the signal then decoded to the L-R domain by L/R - DEC (14). For a stereo picture the MON/ST potentiometer (7) will be fully clockwise; counter-clockwise rotation will gradually reduce the width of the picture to mono, presenting the M-input signal (only) to the L&R outputs from the module by reducing the ambient content.

When an input is in the L-R domain, the following paragraph describes how to perform a mono reduction. BUT, compare such an operation with the M-signal after converting to M-S by L/R - DEC (14)... Because of phase cancellations, a L+R=MONO has less 'vitality' than the equivalent M extraction.

Routing buttons (8) A/B and C/D route the channel stereo signal to outputs A & B and/or outputs C & D. L routes the LH input signal to both A and B outputs; similarly R routes the RH input signal to both A and B outputs; L AND R routes a mix of the stereo signal to both A and B outputs - for these mono capabilities, A/B must also be selected, (C/D remains an independent routing to C and D outputs).

Stereo PFL (9) overrides any other signal on monitor 2 output.

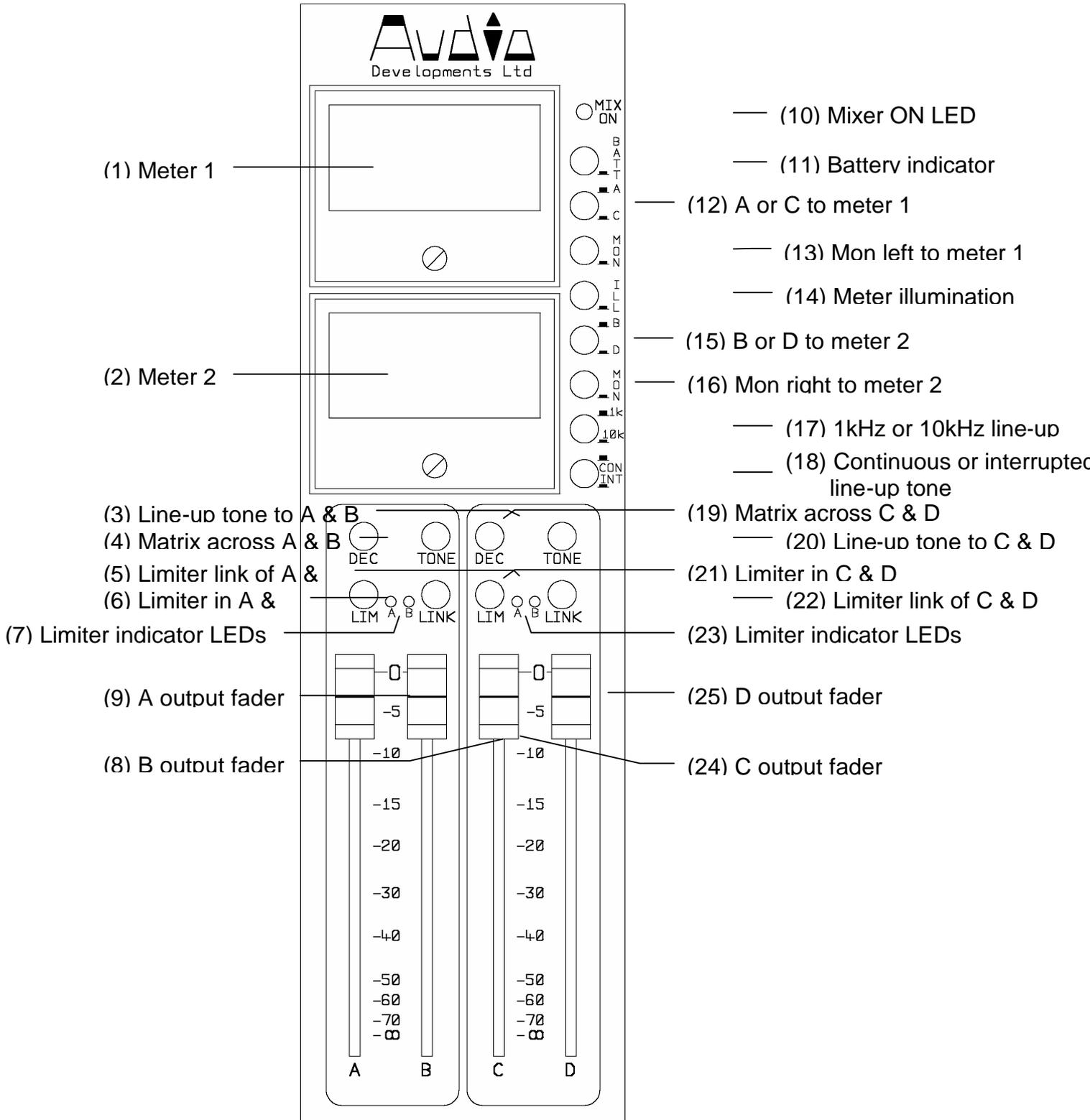
MON (15) is identical with MON on the mic/line module.

Should LEDs (11) and (12) - red=left, green=right - indicate overload, the channel should be checked for excessive equalisation. If this is not the cause of overload, the signal TO the channel should be reduced in level. The LEDs illuminate at 3dB before clipping at the input to the stereo fader.

The stereo Penny & Giles fader (16) is calibrated 10dB down from its fully open position, allowing the operator to work with 10dB of gain in reserve. (The fader may be calibrated at its fully open position if desired - and the scale changed accordingly.)

(NOTE: It should be realised that a pair of mono input modules will always be more flexible and versatile than a stereo module.)

# OUTPUT MODULE



The meters are to broadcast specification with switchable functions; PPM and VU movements are available to choice.

The upper meter (1) reads battery status BATT (11), A and C output levels and monitor 1 left-output level. The lower meter (2) reads B and D output levels and monitor 1 right-output level. The monitor 1 signal will be the one (or mix) selected on the monitor module.

LED (10) illuminates when the mixer is switched ON. Should the internal DC voltage fall below the safe operating level, this LED will flash.

Meter illumination is available via switch (14).

Line-up tone may be selected between 1kHz and 10kHz via switch (17). Tone will be continuous on A, B, C, D unless interrupted tone is selected (18); in which case tone will be continuous on A and/or C but will appear in 3 second bursts on B and/or D for identification of the right-hand signal path. When tone is selected to A & B (3) and/or C & D (20), it will replace any other signal on those outputs.

A matrix amplifier may be switched into the A/B pair of outputs (4) and the C/D pair of outputs (19). See M-S NOTES for further information.

Limiters may be switched into A and B outputs (6) with their actions being monitored by LEDs (7). LINK (5) links the pair of limiters for parallel tracking when A and B are used as a stereo-output pair. Similar limiters are provided for C and D outputs. Limiter threshold is set at the factory to PPM6 (+8dBm) as standard.

A, B, C, D output faders by Penny & Giles (8), (9), (24), (25) are calibrated at the top of their travel to PPM4 with a line-output level of 0dBm. (Fader ganging clips are available for stereo use.)

(NOTE: Metering of main outputs is performed at the input of the output transformers. Metering of monitor 1 is pre monitor-output potentiometer.)

## ADJUSTMENTS AND CALIBRATIONS

Refer to component location diagrams in TECHNICAL LIBRARY.

Line-up-tone oscillator - the preset (VR1) that adjusts its level at the outputs of the mixer is mounted on the right-hand PCB of the output module and can be adjusted from the copper foil side of the board.

Meters - either a VU or a PPM may be selected. A choice of 3 scales is available for the PPM: BBC, N10, SDR.

VU meter - there is one preset, on the VU meter PCB, for meter adjustment. Set an input module for a line input and introduce a 1kHz tone from an audio-signal generator. Adjust levels to achieve a reading of +4dBu, at a main output, on an AC millivoltmeter - measured across pins 2 & 3 on the output XLR. Switch the appropriate meter to the selected output and adjust the preset to give a reading of 0VU.

PPM - driver PCB has three calibration controls (presets). VR1 adjusts the reference level, VR2 and VR3 adjust the 'law' of the meter. VR2 adjusts the upper sector of the scale and VR3 adjusts the lower sector. To initiate calibration, set each preset to its mid position.

BBC Scale - using the tone generator and millivoltmeter as described above, adjust mixer levels for a reading of 0dBu, on the millivoltmeter, from a main output. Switch the appropriate meter to the selected output and adjust VR1 for a reading of 4. Increase the output signal to +8dBu and adjust VR2 for a reading of 6. Decrease the output signal to -8dBu and adjust VR3 for a reading of 2.

Repeat this procedure until an accurate set of readings is obtained without further adjustment to any of the presets being required. Now check all points 1 to 7 on the scale to determine whether they are within specification.

N10 Scale - for a meter calibration of TEST = 0dBu. With a main-output signal of -6dBu, adjust VR1 to give a meter reading at -6. Increase the signal to +6dBu and adjust VR2 to obtain a meter reading at +6. Lower the output signal to -18dBu and adjust VR3 to obtain a meter reading at -18. Repeat this procedure to obtain an accurate reading at each of these three points. Now check the calibration of all meter points.

SDR Scale - with a main-output signal level of -6dBu, adjust VR1 to obtain a reading of -12. Increase the output signal to +6dBu and use VR2 to obtain a reading of 0 on the meter. Lower the output signal to -18dBu and adjust VR3 to give a reading of -24. Repeat the procedure to obtain an accurate reading at each of these three points. Now check the calibration of all meter points.

Output limiters - calibration involves four presets per output; however, only two are involved with a particular parameter. Using 'A' output as the example...

Biasing and threshold are adjusted as follows: VR104 biases the limiter circuit to the point of correct operation and VR103 sets the threshold. (Labelled SET 0 and TH respectively on the left-hand PCB.)

With no signal present, VR104 should be adjusted to give a reading in the range -1.5v to -2.5v at PIN 12 of IC7. VR103 should be adjusted to give a reading in the range -2.5v to -3.5v at PIN 10 of IC7.

Apply a signal at 1kHz to the mixer to give a level of 0dBu at 'A' output. Select the limiter function and adjust VR104 until the output signal starts to fall, (typically 0.2dB). The DC voltage at PIN 12 of IC7 should be approximately -2.0v.

After setting VR104, adjust the output signal to just greater than +8dBu with the limiter deselected. Introduce the limiter and adjust VR103 until the output signal falls to +8dBu. This is the THRESHOLD setting. (If a different threshold setting is required, alter the signal levels accordingly.) The DC voltage at PIN 10 of IC7 should be approximately 0.6v different from that at PIN 12.

The other outputs can be set using this procedure, but the preset and IC numbers change to correspond with the output being calibrated.

The LINK function has no individual setting of its own. To ensure this works within specification it is important that A & B and C & D outputs are set up in pairs. After following the above procedure, the DC voltages at PINS 10 and PINS 12 of the ICs are identical for each pair of outputs. (Tolerance  $\pm 0.05v$  with typical figures at PIN 10 of -2.6v and PIN 12 of -2.0v.)

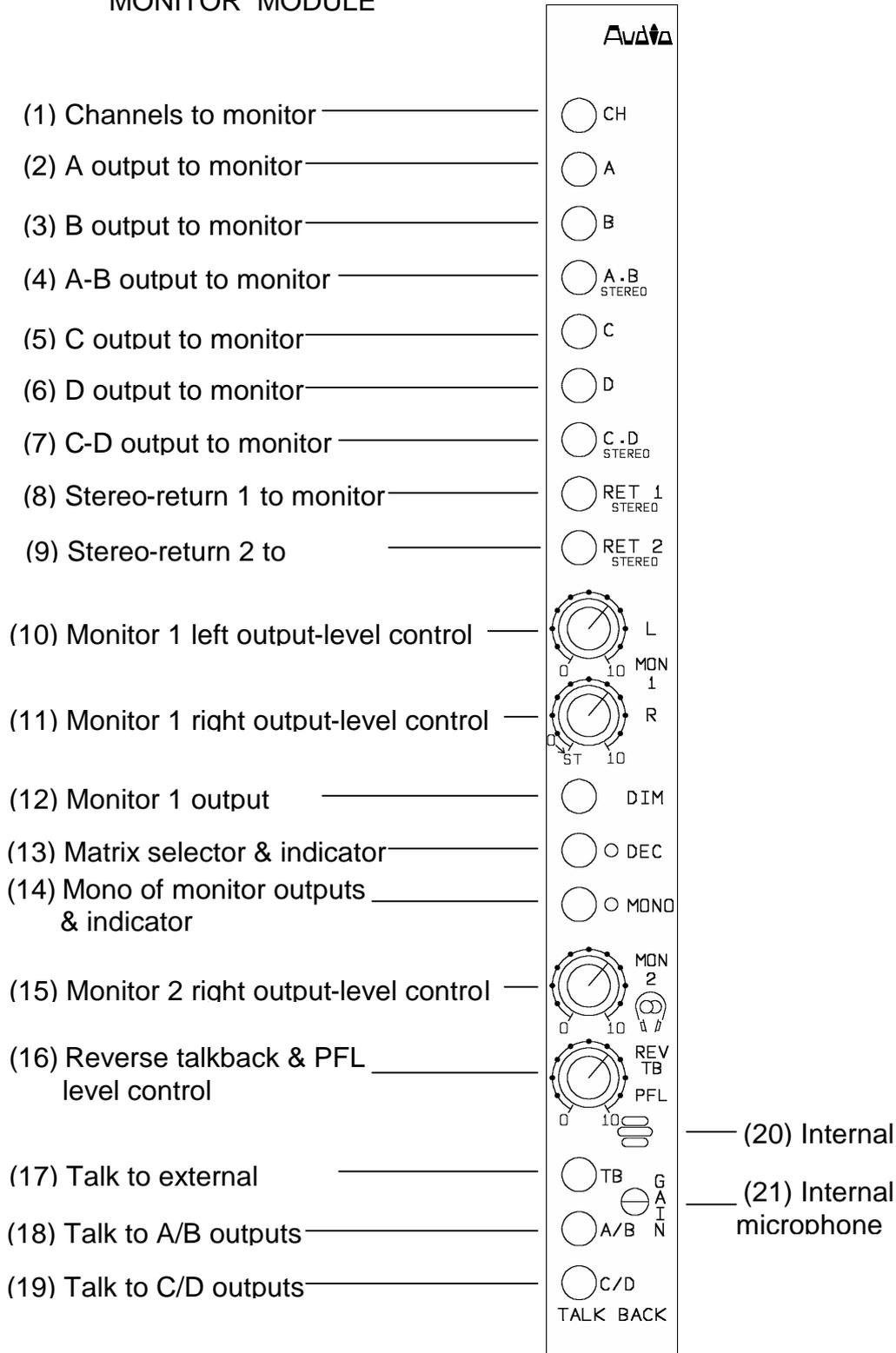
The ATTACK time is varied by VR102. The lower the resistance across VR102 (counter-clockwise), the faster the attack.

The RELEASE time is varied by VR101. The lower the resistance across VR101 (counter-clockwise), the faster the release

(If appropriate timing or measuring equipment is unavailable, VR101s and VR102s should be set in the same position, as accurately as possible, by eye.)

For correct operation, it is essential that the attack times for each pair of outputs and the release times for each pair of outputs be the same, ie A & B and C & D. (The two PAIRS may have different settings.)

# MONITOR MODULE



The monitor module is associated with all aural monitoring of input and output signals together with communications between the sound operator and an outstation.

It is necessary to be able to audition (in left-right stereo, ie a form the human brain understands), an X-Y (A-B) stereo microphone and an M-S stereo microphone whether the channel panpots are set centrally or hard left and hard right (thus creating conflicting monitor requirements). It is also necessary to be able to audition X, Y, M, S in isolation. Similarly, facilities must be provided to monitor main outputs individually and as stereo pairs, whether they be in the M-S or L-R domain. It is a bonus to be able to determine the potential of these various signals in their after-life, ie in post-production.

The signal/s to be auditioned on the left and right outputs of monitor 1 and monitor 2 is/are selected from the bank of switches (1) to (9). These switches are not interlocked in order that any desired mix of signals may be achieved.

Switch (2) selects A output from the mixer to monitor 1 (&2) left and right outputs.

Switch (3) selects B output from the mixer to monitor 1 (&2) left and right outputs.

Switch (4) selects outputs A & B as a stereo pair to monitor 1.

MONO (14) mixes any two-channel signal appearing on monitor 1 path and presents it to both left and right outputs of monitor 1 - in this case a mono mixdown of the A.B stereo signal. A & B [(2) & (3)] produces the same result.

Switches (6), (7), (8) provide the same functions for C and D outputs from the mixer.

Signals for output monitoring are taken from the final mixer outputs, ie after the faders, output matrix-amplifiers and limiters.

Switches (8) and (9) route the two stereo-returns to monitor 1

CH (1) operates in conjunction with the MON switch on the channels. The MON signal is the same as that which drives A,B,C,D mix busses. If MONO (14) is also selected, the monitored signal becomes a true (mono) AFL (after-fader listen) signal from the channel.

'How does a microphone signal sit in the mix?' By alternating CH (1) and STEREO (4) or (7) and routing the appropriate microphone channel to MON, the question is easily answered. This is SOLO-IN-PLACE (SIP) use of the monitor 1 path.

The output from the monitor 1 path (and to monitor 2 output) is controlled by a pair of potentiometers - left (10), right (11). If (11) is turned fully counter-clockwise, (10) controls monitor 1 output as a stereo pair. Should the monitor 1 signal be in the M-S domain at this point, varying R (11) will change the width of the stereo signal - after it

has been matrixed to the L-R domain by DEC (13). Any signal in the L-R domain may be matrixed (13) in order to reveal mono capability on monitor 1 left and out-of-phase components on monitor 1 right. These signals may be isolated by selective use of level controls (10) & (11) and MONO (14). Refer to M-S NOTES.

Operating on monitor 1 output is the dim control (12) which dims the output by 20dB. This dim function is automatic when the internal microphone (20) is routed to A/B (18) and/or C/D (19). Dim does not affect a monitor 1 signal that is routed to the headphones.

Monitor 1 output (whether stereo or mono) is fed to the headphones, ie monitor 2 amplifier - which also has its own level control (15) - via the headphones logic. Once the correct level has been set for monitor 1 output, the level of this signal to the headphones can be further adjusted by the monitor 2 level-control (15).

The talkback-return signal is fed to the PFL mix busses, which also have their own ganged potentiometer (16) to control the level to the headphones.

When PFL is engaged on a channel, or if the talkback-return is in use, logic changes the source of monitor 2 from monitor 1 to the PFL mix busses.

The internal microphone (20) can be routed, TB (17), to the 6-pin XLR connector, mixed into A&B outputs (18) and C&D outputs (19). Microphone level is set by potentiometer (21). When TB (17) is selected, the microphone signal replaces any other signal present on TB SEND. The normal signal at TB SEND is a mono mix of Monitor 1. This function is set at the factory but may be disabled by the DIL switch on the left-hand PCB of the monitor module.

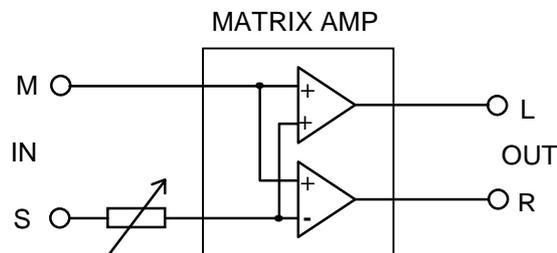
The return signal from an external source feeds onto the PFL line. This is initiated by grounding the control line (PIN 6) at the external source.

Thus, for example, a two-way conversation can take place between mixer and boom operator with the boom operator being able to listen to programme when no communication is taking place.

## M-S NOTES

The M-S techniques proposed by Alan Blumlein in the early nineteen-thirties have recently been (re)discovered. These techniques fall into two parts: those used during original recordings, and those used in post-production. The way to change the width of the image from an X-Y stereo microphone (without physically moving it), or from an existing L-R recording, is to convert to the M-S domain in order that the M/S relationship can be altered. Because the human brain is incapable of unscrambling signals in the M-S domain, all M-S signals must be returned to the L-R domain. These conversion procedures can be achieved electronically or by the use of a specially designed transformer, and are called encoding - or is it decoding? In order to overcome any possible ambiguity, we refer to the encoding and decoding processes as matrixing - since they involve a matrix amplifier. This matrix amplifier will accept a L-R input signal and convert it to an M-S output; or an M-S input signal and convert it to a L-R output - hence the confusion.

The matrix is a sum and difference amplifier which adds, in phase, the two input signals to produce a left signal [ $M+S=L$ ]; it also adds the two input signals in anti-phase to produce a right signal [ $M+(-S)=R$ ]. Similarly [ $L+R=M$ ] and [ $L-R=S$ ].



Increasing the S-signal widens the L-R stereo image; decreasing the S-signal makes the stereo image narrower.

Changing the phase of an S-signal swaps over the left and right outputs of the matrix. Thus if an M-S microphone is set up with incorrect orientation of the S-microphone, changing the phase of the S-microphone channel will rectify the situation - without having to disturb the microphone.

Mixing in the M-S domain is a legitimate technique but X-Y microphones will first have to be matrixed to the M-S domain.

Blumlein's microphone techniques have advantages over all others (X-Y and A-B) such that it is worthwhile using M-S microphones even when mixing is to be done in the L-R domain. For example, the M-microphone will be placed in the optimum position for a mono image, whereas mixing to mono the signals from an X-Y stereo pair introduces phase anomalies.

When using M-S microphones we suggest, for convenience, that M-microphones are inserted in odd-numbered channels, and the corresponding S-microphones are inserted in the next channels above.

Many are the tricks one can perform in the M-S domain, but the only one considered here is the possibility of changing the width of a L-R stereo signal. Should readers wish to investigate further M-S techniques, they are invited to request a copy of the M-S Handbook written for our PORT-A-FLEX units AD066-11, M-S stereo microphone amplifier and AD066-12, M-S post-production matrix amplifier.

AD146 will accept any combination of M-S and X-Y microphones and mix them in either the M-S domain or the L-R domain. Therefore the outputs from the mixer may be in either the M-S or L-R domain. The two domains can be combined to produce eight possible mixing paths; of these, the following three are the most commonly used:

- (a) M-S microphones (with or without X-Y microphones) mixed in the M-S domain: output M-S or L-R.
- (b) M-S microphones (with or without X-Y microphones) mixed in the L-R domain: output L-R.
- (c) X-Y microphones mixed in the L-R domain: output L-R.

When mixing in the L-R domain and using one or more M-S microphones, the panpots on M- and S-channels will be centred and the S-switch selected on the S-channel to create a matrix across the M and S channels. With CH (1) selected on the monitor module, MON on an M-channel will audition the M-microphone on monitor 1 L & R (and headphones L & R) outputs. When 'setting up' M-S microphones, an S-microphone may be auditioned in a similar manner - BEFORE the S-switch (15) is selected. After selecting the S-switch (for S-microphones only), the complete M-S microphone may be monitored - automatically in L-R stereo. The potential of such a microphone may be checked by changing the M/S relationship, ie by varying the input gain of the S-channel. To monitor an S-channel after the S switch has been selected, CH (1) will reveal +S on monitor 1 left and -S on monitor 1 right - not very useful. To overcome this difficulty, reduce the right output to zero and select MONO (14) in order to isolate the in-phase component from the S-channel. Spot microphones will be panned to their correct position in the final L-R image, and may be monitored as non-destructive solo-in-place or as true AFL.

When mixing in the M-S domain (and) using M-S microphones, M-channels will be panned left, S-channels will be panned right and spot microphones will be panned left to mix with the M-channels only. With CH (1) selected on the monitor module, MON on an M-channel will audition the microphone on monitor 1 left but use of MONO (14) will place the signal on both outputs of monitor 1. Similarly, MON on an S-channel will audition that microphone on monitor 1 right. If MON is selected on BOTH channels of an M-S microphone, the resultant signal (M-S) on monitor 1 will be incomprehensible and must be matrixed to the L-R domain by DEC (13).

An M-S main output A/B (4) or C/D (7) may be monitored in the L-R domain by selecting DEC (13).

An M-S main mix may be matrixed to the L-R domain for the final output from the mixer by DEC (4) for A/B and DEC (19) for C/D on the output module.

Signals for output monitoring are taken from the final outputs of the mixer, ie after the faders and output-matrix amplifiers.

Monitor 1 left and right outputs may be mixed to mono by selecting MONO (14) - an LED indicates when this function is in use. The mono signal is fed to both sides of monitor 1 output.

When mixing in the L-R domain without M-S microphones, X-Y (A-B) microphones will be panned hard left (X) and hard right (Y), and any spot microphone will be panned to its correct position in the final L-R image.

CH (1) on the monitor module and MON selected on both channels of an X-Y pair auditions each stereo microphone. MON on a spot microphone is solo-in-place usage of the monitor system.

POWER SUPPLY UNIT  
TYPE AD100-09

The AD100-09 mains POWER SUPPLY UNIT is suitable for driving most of AUDIO DEVELOPMENTS' range of portable audio mixers. This PSU is a single-rail device providing 500mA of current at +14v DC potential and is used as a substitute for battery power with mixers containing an internal DC-DC converter.

The AD100-09 may be powered from either a 110/120v AC source or a 220v/240v AC source. Ensure that the AC Voltage Selector Switch on the front panel is in the correct position for the source in use. Operating the equipment at the wrong voltage could be hazardous. Care must be taken to connect the LIVE, NEUTRAL and EARTH pins of the PSU's IEC mains connector to the corresponding terminals associated with the AC source. The ON/OFF switch contains an indicator that illuminates when the PSU is operational.

FOR SAFETY REASONS, AD100-09 POWER SUPPLY UNIT MUST BE CONNECTED TO MAINS EARTH. Any maintenance to the PSU or its mains cable assembly should be performed by a qualified engineer.

**CHARGING:** If nickel-cadmium cells are fitted in an AD140 series mixer, they may be recharged in situ from AD100-09 power supply - whether the mixer is in use or not. (Maximum current is set at 250mA - in addition to the 500mA of current supplying the audio electronics.) The charging circuit has its own ON/OFF slide switch and LED indicator. **DO NOT ACTIVATE THE CHARGE CIRCUIT UNLESS THE MIXER IS FITTED WITH NICKEL-CADMIUM CELLS.**

**FUSES:** Two 20mm ANTI-SURGE fuses protect AD100-09 against fault conditions. Should either fail, it is strongly recommended that the cause be traced. Refer to the TECHNICAL LIBRARY. Only suitably qualified personnel should service the power supply unit. The fuse holder on the front panel contains the mains fuse.

250mA HRC TYPE T 240v AC
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For continued safety the specified fuse link must be fitted in the mains fuse holder when a replacement is required. Ensure it is of a type approved by a National Approved Body.

DC-OUTPUT XLR	PIN 1	0v	PIN 3	NOT CONNECTED
	PIN 2	CHARGE	PIN 4	+14vDC

**DO NOT REMOVE THE OUTER COVERS**

NOTE: The power supply unit should be serviced by a suitably qualified engineer. Only genuine spare parts with identical specifications must be used.

It is DANGEROUS to change the specification or modify the product in any way.

## CUSTOMER NOTES AND FACTORY MODIFICATIONS

TECHNICAL LIBRARY