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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. 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

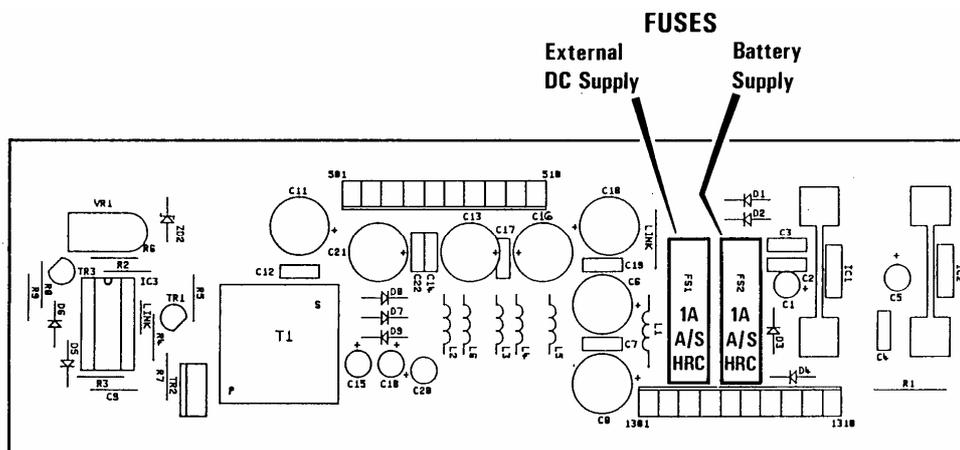
Battery compartment - the mixer has an integral battery compartment to accept 10 size-C cells, and is formed as part of the bottom transversal extrusion. This helps to lower the centre of gravity, as well as adding to the rigidity of the frame.

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.

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

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.



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.

LH-output ident - as an alternative to continuous tone on both left and right outputs (and AUX 1/2 outputs), the LH and AUX 1 outputs can be pulsed with tone for three 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.

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

ALL DIL SWITCHES ARE ILLUSTRATED AND SET AT THE FACTORY AS SHOWN ON BLOCK DIAGRAMS. (Refer to page 74)

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 AD149 mixer and must not be used.

M-S NOTES

GENERAL BACKGROUND

Of the various techniques that have evolved for stereophonic recording, the so-called COINCIDENT, NEAR-COINCIDENT and SPACED-APART techniques have found favour in the recording of classical music - being better suited to preserving natural perspectives and balances; whereas for the recording of popular music, the favoured technique is one which uses many microphones closely positioned to their sources for maximum isolation.

Increasingly, it is being realised that, of the coincident microphone techniques, the rediscovered M-S (middle-side) technique has many advantages over the other coincident techniques, particularly when working in unknown or difficult situations.

Several leading manufacturers are now producing integrated M-S microphones, but many are of the electret variety exhibiting typical frequency-response aberrations. Our own extensive research and experimentation have shown that the highest quality and maximum flexibility are obtained when using the new generation of discrete, true condenser microphones. An extremely high quality M-S system can, for instance, be assembled from the new SENNHEISER range of studio condenser microphones (MKH20,30,40,50,60,70) together with the MZD30 which clips two microphones together in 'piggy-back' style and aligns the capsules in the same plane. (It should be noted that these microphones use RF technology and therefore offer high immunity to moisture in the atmosphere.) With such a system, not only can the most suitable choice be made for the M-signal microphone - omnidirectional, cardioid, hypercardioid, (short-gun, long-gun) - but also, there is no requirement for extra microphone powering units and/or matrix amplifiers as AD149 is able to supply all the control necessary for professional, stereophonic recording and post-production.

For stereo TV and film, M-S is the obvious choice; it permits the easy matching of audio and visual images - focusing performers in the centre of a wide 'stage' or expanding the stereo image and adding ambience on 'long shots', and, if mono compatibility is of great importance, this format is the clear winner. (The majority of the listening audience still hears a mono mix.)

It has been shown that more ambience is preferred when listening in stereo than when listening in mono. Recordings made using the M-S technique satisfy these preferences as the presence of the S-signal increases ambience with the M-signal being true mono. Because of the way the pair of microphones responds to reverberant information, a well done M-S recording preserves a high degree of realism.

When using the M-S technique, the microphone system can be positioned in order that the forward facing capsule produces the optimum mono signal with the best possible frequency response whilst maintaining easily adjustable stereo perspective

and ambience, whereas recordings made with X-Y techniques can suffer from centre-stage high-frequency deficiency caused by the off-axis degradation of HF response of many directional microphones. Critical and discerning listeners are aware of the image shifting with changing frequency caused by less-than-perfect microphone polar patterns or the use of unmatched pairs of microphones. These same listeners will also be disturbed by the phasing effects caused by the movement across the faces of badly positioned 'spot' or highlighting microphones, and by the fact that images at low frequencies are often narrower than those at high frequencies.

In view of the narrowness of the angle subtended by a TV or cinema screen at a typical viewing position, "Why stereo?" is a question frequently asked. "Why not?" is the reply more frequently being given - even stereo dialogue is becoming increasingly common. Advantages of stereo include a more accurate portrayal of the local acoustic, separation of voices with space and air between them, correct placement and perspective of locally occurring sounds and sound effects.

COINCIDENT MICROPHONE TECHNIQUES (Intensity Stereo)

A crossed pair of microphones is used with the capsules USUALLY arranged in a vertical plane and as close to each other as possible; the angle included by the capsules is symmetrical with respect to the central axis of the sound source. The angular accuracy of the stereo image is independent of the distance from sound source to microphone, but the resultant recording can seem rather dry and lacking a sense of space.

There is a choice of capsules available when using this crossed pair or X-Y technique. The earliest of these techniques were developed by Blumlein. ONE of his techniques uses a pair of figure-of-eight microphones set at an included angle of 90° . Best used in large spaces where reflections from boundary surfaces are minimal and where strong signals are not presented to the rear quadrants, this technique produces a very natural sound.

When cardioid or hypercardioid microphones are used, the included angle is usually 90° - 120° . (Some authors reserve the term X-Y to refer only to a crossed pair of cardioid microphones.) The included angle determines the width of the stereo image and is selected with other factors in mind; distance from, and width of the sound source and polar response of the chosen microphones. The polar and frequency responses (together with sensitivities) should be matched as closely as possible, otherwise considerable colouration and image shift can occur with changes in pitch.

When cardioid capsules are used they are usually placed fairly close to the sound source giving a high ratio between direct and reverberant sound, and good rejection of unwanted sound at the rear of the microphone pair. This means that in TV recording the microphone appears on camera. A longer range system is required for picking up dialogue in drama etc. Even listeners in stereo will have difficulty with text if an X-Y pair is set out of sight and probably with neither microphone pointing directly at the sound source.

Even with an optimally positioned X-Y pair of cardioid microphones there is no guarantee that the resultant recording will fill the space between the loudspeakers. Post-production techniques are available to overcome this and other problems, but experience has shown that to take advantage of these techniques the capsules should be **CROSSED OVER** and spaced apart at a distance of approximately 50 mm. (FIG 1)

This spacing, which does not apply to other directional microphones, actually improves stereo imaging.

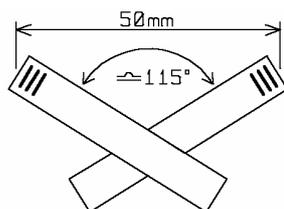


FIG 1

The use of hypercardioid capsules offers a good compromise between crossed pairs of cardioids and crossed pairs of figures-of-eight. The beam of a hypercardioid is narrower and the 'reach' longer than that of a cardioid, therefore these microphones can be used at a greater distance from the sound source for the same direct/reverberant sound ratio but, because of the presence of a rear pickup lobe, there is an increase in the sense of space. The narrower beam is a slight disadvantage in that in order to provide a solid central image, the included angle must be less than that for a pair of cardioid microphones. Hence, this system is not entirely suitable for wide sound sources.

The M-S technique was rediscovered by Lauridsen and uses a microphone (M) aimed along the central axis of the sound source and a figure-of-eight microphone (S) at right angles to this axis (it is assumed that the positive lobe is orientated to the LEFT for the purposes of these notes). The outputs of the microphones are combined in a 'sum and difference' matrix amplifier to produce conventional left and right stereo signals. Later summation of the L-R signals to produce a compatible mono signal is more predictable than when using stereo signals recorded using other formats (=2M); alternatively, the original M-signal can be used. Thus, by careful choice of type and position for the M-microphone, the optimum mono signal can be achieved with a minimum of off-axis colouration.

Of all stereo recording techniques, M-S is the one that offers the engineer the most control and flexibility, not only during recording but also during post-production. By varying the ratio of the signals sent to the matrix amplifier, the width of the stereo image can be adjusted. By changing the pickup pattern of the M-microphone (a remotely controlled polar response being advantageous in this context), the apparent distance from the sound source, and hence the amount of ambient information in the M-signal, can be controlled.

The most popular choice for the M-signal is a cardioid microphone. In this case, after matrixing the two signals, the resultant left-right stereo is equivalent to an X-Y pair of hypercardioid microphones. The cardioid is the only M-microphone which does not encode rear sounds as front sounds - overall pickup from the rear being low. Angular distortion is low over the pickup angle and there is uniform directional encoding of reverberation. If an omnidirectional microphone is used for the M-signal, the X-Y equivalent is a pair of back-to-back cardioid microphones with their common axis parallel to the axis of the figure-of-eight microphone. This arrangement is useful when performers can be arranged on both sides of the microphone pair.

If a figure-of-eight microphone is used for the M-signal, the X-Y equivalent is a crossed pair of figures-of-eight, each being at 45° to the central axis of the sound source. Of all M-S techniques, this form provides the most even distribution of reverberant pickup.

(Note: it is assumed that $M=S$, ie the microphones have the same sensitivity and that there is the same amount of gain in each of the two signal paths.)

When $M>S$

M=omnidirectional: the resultant cardioids become more omnidirectional.

M=cardioid: the rear lobes of the resultant hypercardioids are reduced in size and the included angle decreases.

M=figure-of-eight: the included angle between the resultant crossed pair is less than 90° .

When $M<S$

M=omnidirectional: the resultant cardioids become more hypercardioid.

M=cardioid: the rear lobes of the resultant hypercardioids are increased in size and the included angle increases.

M=figure-of-eight: the included angle between the resultant crossed pair is greater than 90° .

'All things being equal' (which, of course, they never are), once an M-S pair has been installed, varying the input gain of the S-channel will not only alter the width of the resultant L-R presentation, but also effect changes as outlined in the above table. Whereas, if an X-Y pair is being used, changes would involve physical manipulation - alterations to included angle, to position, or replacement with a different type of microphone. (Refer to FIG 2)

M-S / X-Y TRANSFORMATIONS (M=S)

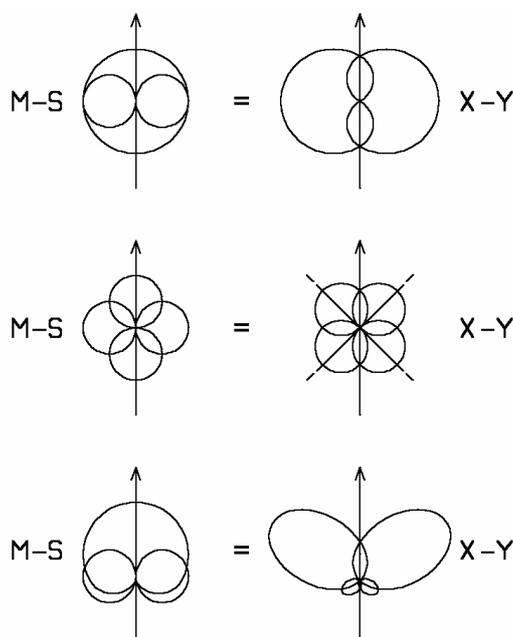


FIG 2

(An X-Y pair of cardioid microphones matrixes to a sub-cardioid and figure-of-eight microphone.)

Sounds arriving where the M- and S-patterns intersect are reproduced fully left or fully right and define the included angle of sound which is spread completely between the loudspeakers.

A very effective M-S combination for use where normally a single gun microphone would be used is a hypercardioid microphone (M) in conjunction with a figure-of-eight microphone (S). However, during our research it has been made very clear to us that the Sennheiser MKH 416 short gun microphone is going to be a very popular choice for the M-signal. When using gun microphones for the M-signal it must be borne in mind that they become more directional with increasing frequency and have LF and MF side lobes. Very odd effects can arise if used against a moving background. A stream of traffic, say, will change its perspective as it moves across the various lobes.

(It should be noted that the sensitivities of the 12V tonader and the 48V phantom powered versions of this microphone are not the same.)

SPOT MICROPHONES

Very great care must be taken when using highlighting microphones within a stereo sound picture. When directional microphones are used, problems include blasting and proximity effects, and image shift as the sound source (soloist) moves.

A 'spot' microphone can adversely affect the balance of the surrounding part of the sound stage and an insensitive engineer can easily make the microphone appear out of context with the overall stereo image. As the signal from the soloist's microphone arrives in the mix ahead of the basic pickup, one solution is to delay this signal 10-15ms with respect to the main signal or even to fall in the region of early reflections; another is for just sufficient of this signal to be added to the mix to increase the presence of the soloist. Precise positioning of the 'spot' microphone within the overall picture is essential in order that the image does not shift with changing frequency and intensity.

When using the X-Y format, a crossed pair of omnidirectional microphones with an included angle of 60° - 90° provides a very stable image of the soloist without blasting, or proximity and directionality effects, but with a well-defined sense of space. When using the M-S technique, further M-S pairs can be used as soloists' microphones; each subsidiary pair will be steered to its correct position within the sound stage, and its included angle will be less than that of the main pair.

So that maximum flexibility is maintained during post-production, these spot microphones will be recorded on separate pairs of tracks on the tape.

For ENG/interview work it is perhaps legitimate to adopt a less purist approach without making too many compromises. A mono microphone may be mixed into the

M-channel ONLY, with the result that its image will take up a central position in the final L-R stereo presentation. During post-production, the width and ambience of the background may be changed, but the balance between the M-S pair and the mono microphone remains fixed.

If a single M-S pair is used, the M-microphone should be pointed between the interviewer and the interviewee. This technique will provide a more stable and coherent image than mixing into the M-channel a microphone moving between

sources - and consequently with a changing background - unless this microphone is so close to the source that its background pickup is negligible compared with the pickup of the M-S pair.

If an M-S microphone is too close to the sound source, small movements of the source lead to large changes in the angle of incidence and thus excessive movement in its position within the sound picture. This is particularly true when the S-signal has a high gain. Conflict will occur when the situation demands a close mic position for the sake of clarity.

In all circumstances, a camera-mounted stereo microphone is quite inappropriate.

When it becomes necessary to employ spot or mono microphones for the majority of the pickup, the addition of a stereo pair will produce the impression of solidity and give a sense of presence, particularly when the main audio is ambience, as in sports events.

NEAR-COINCIDENT MICROPHONE TECHNIQUES

These techniques provide good positional information together with a satisfying sense of space as they rely on both intensity and phase differences. The microphones are set close enough together to be effectively coincident at low frequencies and far enough apart for there to be a time delay between sounds from the left and right extremes of the sound source. Because of the spacing between the capsules, large image shifts can stem from quite small movements within the sound source, hence, close miking is not advisable. (Also, sounds arriving from the extremes of the sound source may cause problems when deriving a compatible mono signal and when disc cutting.)

There are four main near-coincident techniques:

- The ORTF technique was developed by the French broadcasting organisation. Two cardioid microphones are spaced 170 mm apart with an included angle of 110°.
- The NOB system was developed by the Dutch broadcasting organisation. Two cardioid microphones are spaced 300 mm apart with an included angle of 90°.
- A system developed by Faulkner uses two forward-facing figure-of-eight microphones spaced 200 mm apart. The microphones are positioned further back from the sound source than is usual with other near-coincident techniques. The results show many of the qualities of the Blumlein technique and a natural balance between close and distant elements within the sound source.
- The binaural technique was specially developed for listening to playback on headphones but the recordings are considered by some listeners to be unsatisfactory when heard on loudspeakers. This system uses a 'dummy head', usually made from polystyrene, with omnidirectional microphones placed into the 'ears'. This technique provides excellent imaging in both horizontal and vertical planes. A development of the technique uses two omnidirectional or figure-of-eight microphones set approximately 90 mm on either side of a circular, non-reflective baffle. This allows sufficient 'leakage' between the microphones to make reproduction over loudspeakers a satisfying experience.

In recent times attempts have been made to bring to loudspeaker listening the naturalness, and the depth and sense of space present when listening on headphones to dummy-head recordings. These features are usually missing from recordings made with coincident X-Y microphones. To this end, it is necessary that the signals have a natural interaural relationship, but this can only be achieved with some form of dummy-head technique.

The Schoeps KFM6U sphere microphone has been designed to satisfy these requirements. A 180 mm diameter sphere has two omnidirectional microphones set on opposite sides of the sphere and produce a stereo signal with natural interaural

differences. Unlike a dummy-head, this microphone has a flat frontal frequency response and lacks those cues which give front/rear information in headphones listening and which only lead to colouration in loudspeaker listening.

None of these near-coincident microphone techniques is strictly mono compatible.

[Although the use of omnidirectional microphones is rightly preferred, wherever possible, by many engineers, it is worth recalling a few of their frequently overlooked characteristics. They possess an excellent low-frequency response and, although fairly immune to wind and blasting effects, they are particularly adept at responding to traffic, air-conditioning plant and structure-borne vibrations. They provide a good sense of space and 'openness', but, for a given direct/reverberant sound ratio, they require the closest placement to the sound source. They have the maximum potential for pickup of unwanted sounds - although with minimum colouration.]

[When deciding upon an omnidirectional microphone, there is a choice to be made between the type which is designed to have a flat response to an on-axis signal (direct or free field) and the one designed to have a flat response to a reverberant sound field (random incidence or diffuse field), ie with an HF tip-up in the on-axis frequency response. The MKH20 contains treble accentuation circuitry permitting usage in both direct and diffuse sound fields and also gives the microphone a directional effect at high frequencies.]

THE MATRIX PROCESS

Conventional left-right stereo signals may be produced directly by using X-Y techniques, or indirectly by using M-S techniques. In the latter case, the signals have to undergo a further process variously called matrixing, encoding or decoding. The matrix amplifier adds and subtracts the M- and the S-signals - after hard sums, $[M+S=L]$ and $[M-S=R]$. If the positive lobe of the figure-of-eight microphone is to the right, then $[M+S=R]$ and $[M-S=L]$.

If these left and right signals are now added to derive the equivalent mono signal, easy sums show that $[L+R=2M]$. If the matrix amplifier is used for this process, then automatically $[L-R=2S]$. In other words, an existing L-R signal can be matrixed to M-S and then subjected to any treatment which is applicable to M-S signals - manipulation of width of image and ambience etc, for the FINAL L-R presentation. This is a very convenient way of effecting such changes on L-R signals. When using X-Y techniques, such changes have to be made by repositioning the microphone pair.

When the signal is in M-S format, traditionally (if a tradition there be) the stereo spread is controlled by varying the level of the S-component. A narrower spread is obtained by reducing the level of the S-component. Similarly, the width is increased by increasing the S-component. This technique also has the effect of altering perspective and ambient content, without affecting the mono signal.

A refinement of the method is to control the width by a balance control - increasing S at the same time as decreasing M and vice-versa. This has the effect of maintaining

overall level of the stereo signal. To steer the output of an M-S pair to any position within the final stereo picture, the outputs of the matrix amplifier are mixed, differentially, into the main L-R signal. [Refer to FIG 3(a) and FIG 3(b)]

It is essential that the relative phase of all stereo signals is maintained. If X or Y is phase reversed, the stereo image will be imprecise with a lack of low frequencies. A mono reduction will now be the S-signal - which is not a normal audio experience.

If the S-signal is phase reversed, the final presentation will be changed from L-R to R-L. Reversing the phase of S is a useful technique if the orientation of an M-S pair has been changed during a recording. For example, a boom may have been changed from an 'over-the-top' position to an 'up-and-under' position.

In a situation where several M-S pairs cover the sound stage, the S-level control can be used to reduce to narrow stereo (or even mono) in times of trouble or when making difficult cross-fades between pairs of microphones.

MATRIX AMPLIFIER WITH WIDTH CONTROL

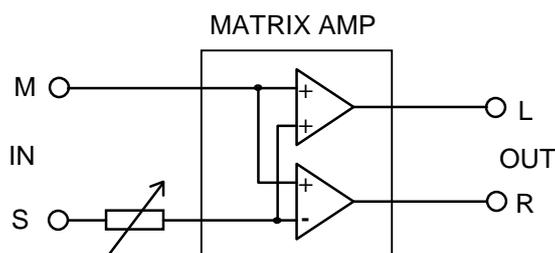


FIG 3 (a)

If a mono signal is to be derived from fully modulated left and right signals, simple addition will result in over-modulation of the mono channel due to the increase in level caused by centrally situated signals. Some manufacturers reduce this resultant level by 3dB and some by 6dB.

Mixing M-S signals is a valid and very powerful technique, particularly when matching sound to visual images. A mono signal is directly available; the stereo image can be made narrower or wider; the position of dialogue can be changed within the stereo image without affecting the stereo ambience and equalisation can be applied without affecting the mono signal.

SHUFFLING

This is a fancy name for frequency-conscious width control and based - naturally - on Blumlein's work. By inserting an equaliser in the S-signal, the gain may be changed in a frequency-conscious manner which can lead to improvements in directionality in certain types of stereo recordings. To increase the width of a stereo image at a particular frequency or over a range of frequencies, the gain of the S-signal is increased at those frequencies. Similarly, to reduce the width of a stereo image at particular frequencies, the gain of the S-signal is reduced at those frequencies.

Later research has suggested that an equaliser should also be inserted in the M-signal so that when the S-signal is increased at specific frequencies, the M-signal may be reduced slightly at the same frequencies in order to maintain overall frequency balance. Similarly, if the S-signal is reduced, then the M-signal is increased slightly at the same frequencies.

Graphic equalisers would appear to be the obvious choice for shuffling, but they are made up of a series of narrow-band (therefore 'ringy') filters and this can lead to a 'ripply' frequency response. Far better results will be obtained by using a constant-Q, parametric equaliser with a very low Q-setting, eg 0.2, so that just one filter section may be used to achieve the required result.

AD066-13 FLEX-EQ is such a filter set. This equaliser also satisfies a further requirement - the transition to the 'equalised' part of the frequency spectrum should be as gentle and smooth as possible. The skirts of the FLEX-EQ low-Q filters are of the order of 6dB per octave.

Shuffling is too powerful and complex a technique to cover adequately in these application notes, but it is worthwhile listing some of the situations in which the technique will be advantageous.

Black records become easier to cut and reproduce when the width is reduced at low frequencies. FM stereo reception noise can be reduced by reducing the width around 7kHz. Reduction of low frequencies in the S-channel helps to eliminate structure borne vibrations. Increasing the width at low frequencies will bring into coincidence high- and low-frequency components of an image which have become separated because of imperfections in microphone polar patterns, eg where cardioid microphones become more omnidirectional at low frequencies.

Dryness and lack of spaciousness are inherent in recordings made with coincident cardioid microphones. The sense of space can be improved by increasing the difference signal below about 700Hz. This can also lead to an improvement in stereo imaging away from the 'stereo seat'.

When shuffling techniques are applied to recordings made with near-coincident microphone techniques or techniques in which out-of-phase images can occur when using non-coincident microphones, shuffling should be restricted to frequencies below about 300Hz, ie to frequencies which arrive in phase at both microphones.

By increasing the S-signal (over the full frequency range), the stereo image of some recordings can be extended beyond the left and right loudspeakers. This is yet another of Blumlein's discoveries and contained in the original patent. It is also a very useful technique for expanding the soundstage of a recording made, for whatever reason, with a too-narrow stereo image. Results may be dramatic but, equally, they may be phasey, confusing and unconvincing.

As may be appreciated, shuffling is an extremely versatile technique, but its effects are largely unpredictable, therefore each recording must be treated on its own merits.

Shuffling, as described, takes place at low frequencies and has no effect at high frequencies; audible noise is not increased, which is not the case with simple overall width increase.

If shuffling techniques are to be used during post-production, the monitoring system must be optimised for precise stereo imaging. This effectively eliminates many of the loudspeakers used as studio monitors.

MATRIX AMPLIFIER WITH WIDTH AND POSITION CONTROLS, AND SHUFFLER

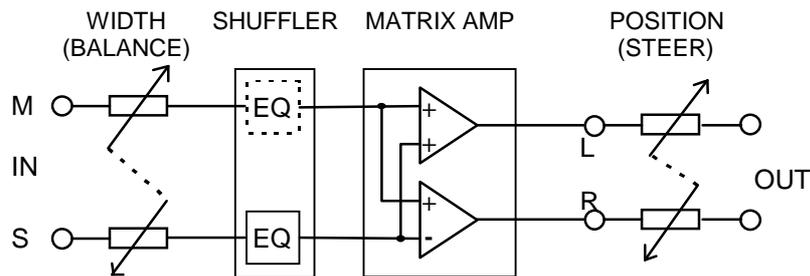


FIG 3 (b)

FURTHER READING

British Patent Specification 394,325. A D Blumlein.
Improvements in and relating to Sound-transmission, Sound-recording
and Sound-reproducing systems.

British Patent Specification 780,337. P B Vanderlyn.
Improvements in or relating to Binaural Sound Transmission Systems.

British Patent Specification 781,187. P B Vanderlyn.
Improvements in or relating to Electrical Sound Transmission Systems.

Stereo Shuffling: New Approach - Old Technique.
Michael Gerzon. Studio Sound: July 1986.

Applications of Blumlein Shuffling to Stereo Microphone Techniques.
Michael Gerzon. J Audio Eng Soc; Vol 42, No 6.

(In later sections we refer to the creation of a 'Blumlein Loop'. This is
a section of signal path in the M-S domain in which Blumlein's
techniques may be applied.)

AD149 AUDIO MIXER

Audio Developments has, for more than 25 years, set the standards for portable audio mixers. With AD149 we have created a new standard for the twenty-first century by offering potential and a degree of sophistication together with a simplicity never previously encountered in a mixer of its type or size: we have created a product worthy to stand alongside, and be used with FLEX-EQ - the world's most powerful equaliser.

AD149 is equally at home with its handle attached, in a studio console or in a post-production/editing suite, and is available powered either by internal batteries or mains only. In both cases, a suitable mains power supply is available for the mixer.

AD146 was designed to be a discreet and supremely quiet mixer. Despite all the extra features that have been incorporated within AD149, by adopting minimum-signal-path topology and introducing circuit-blocks only when required, the mixer's noise performance has been maintained at the level achieved by AD146.

Five modules are available: any combination of input modules - (a) microphone/line, (b) mono line, (c) stereo line - together with an output module and a monitor module (only the latter two having dedicated positions within the frame).

The monitor module is based on the one proved and approved in AD146 - the first four-output mixer in the 140 series. The order of selector switches has been changed to simplify the process of A/B comparisons between sends and returns to and from the recorder. By popular request, each stereo-return has been fitted with a calibration preset. A second EXTERNAL circuit has been included for use by a director/producer and monitoring capabilities available to both external circuits have been increased. A master/safety switch has been added to the communications circuitry, making accidents very unlikely.

Our customers know we are totally committed to providing them with facilities for working in the M-S domain. The new monitor module enables engineers to take advantage of ALL Blumlein's techniques; both during recording and during post-production.

We believe that the monitor path is as important as the main signal-path and one that tells less than the whole truth is valueless. When designing AD146, space was at a premium by the time we arrived at the monitoring section and several (what appeared to be) compromises had to be made. The 'compromises' proved so flexible and versatile, they have been retained in all succeeding models. Because of space constrictions, monitoring had to be simplified - with the following result.

The main monitor path - MONITOR 1 - is addressed via the MON switch on each input module. The secondary monitor path - MONITOR 2 - feeds the stereo-headphones' jack and is addressed via the PFL switch on each input module. BUT - monitor 1 output automatically drives monitor 2 output until any PFL switch is pressed,

when logic changes the source of monitor 2 output to the stereo PFL busses. (When all PFL switches are released, once again monitor 2 equals monitor 1.) This system has now been refined - even further: when using monitor 1 as the main monitor - probably for loudspeaker listening - it is possible to route PFL (which normally only appears on headphones) to monitor 1 by selecting SPLIT (monitor). Pressing a PFL switch will now have the following effect: any stereo signal being auditioned on monitor 1 will be mixed to mono and presented to monitor 1 left-output only; PFL will be mixed to mono and presented to monitor 1 right-output only. As will be seen later (page 54), (mono) PFL can be isolated and presented to both left and right outputs simultaneously.

The monitoring system allows an input channel to be auditioned in three ways:

- (a) PFL; pre-fader listen
- (b) AFL; after-fader listen
- (c) SIP; solo-in-place (non-destructive)

The output module is almost identical with the one designed for AD146 but with simplified meter switching. 'At rest' the meters read L-R output, but this can be overridden in turn by auxiliary 1 & 2 outputs, monitor 1 output and stereo PFL.

Attack and release times of the main-output limiters may be selected on DIL switches on the module's left-hand PCB.

A matrix amplifier - see M-S NOTES - may be inserted in the main L-R output and across auxiliary 1 & 2 outputs.

By request, auxiliary masters are potentiometers rather than faders, but we have included a link switch so that they may be operated as a stereo pair.

An addition is the remote stop/start switch for use with suitably equipped tape recorders, eg Nagra IVS and D. Linked with this switch is the ability to automatically precede and terminate each recording with an ident tone - selected by a DIL switch mounted on the right-hand PCB.

For some time now, line-up tone has proved controversial and contentious - a veritable hornet's nest. When designing AD146 we asked our customers on which side they would prefer interrupted/pulsed line-up tone: logically - and unanimously - the right-hand output was requested. Since that time we have been accused of attempting to create a 'new' standard; but have yet to receive a request for line-up tone to the suggested new international standard (sic). AD149 is supplied with interrupted tone on the left-hand output but can easily be changed by the end-user to the more logical right-hand output.

The input modules have features new to the AD140 series of mixers, several of which are unique to Audio Developments; many being taken from our list of customers' suggestions. In particular, we have been able to accommodate the requirements of film-sound recordists and for non-European methods of working.

Whilst electronic components continue to shrink in size, mechanical components - and human fingers - steadfastly refuse so to do. Throughout the AD140 series, reducing the size of modules has also reduced the panel area available for connectors. We know that connectors mounted on side panels are totally unacceptable, so when insert points were demanded, the only possible position for the jack was at the top of the module fascia: at least this makes it possible to dress insert cables along with their input and output counterparts.

Clean feeds may be switched pre or post fader and we are assured that they will only be used when recording from microphones - enabling usage of the second module connector for channel output.

The highly-acclaimed, discreet and discrete transformer-balanced microphone-amplifier designed for AD146 has been incorporated into AD149's microphone/line module. Under 'normal' circumstances, the channel fader operates around the zero mark, giving a safety-net of 10dB, but film-sound recordists have always had their own particular method of working, and in order to guarantee the capture of everything from "Blow, winds, and crack your cheeks! rage! blow!" to the merest zephyr, they open channel faders to maximum and perform any active mixing on the input-gain controls. This maximises the dynamic range of the input amplifier but throws away the aforementioned safety-net in the process. Unfortunately, it also reduces headroom later in the signal path and ensures a less-than-optimum signal-to-noise ratio. It also seems to negate the *raison d'être* for the linear fader.

To accommodate alternative operating methods, we have replaced our usual, switched input-gain control with a potentiometer and added an additional safety-net in the form of a limiter controlling the gain of the input amplifier (ALC). This limiter has, however, a more important rôle: no longer do we have to tolerate the mixer output being limited because of the output from a particular microphone; the microphone can now be controlled at source. Odd/even pairs of microphone/line input modules may have their limiters linked for stereo operation by a DIL switch on the mother board - and situated between the relevant pair of modules.

Past technical limitations have been responsible for these less-than-perfect ways of operating. Once upon a time, it was possible to achieve a gain swing of only 30dB in the input stage, necessitating the introduction of a 20dB attenuator (pad) - thus degrading the signal-to-noise ratio - to enable the amplifier to accept the full voltage range from a typical microphone.

Now it is possible to produce a virtually overload-proof input stage in which a pad is redundant, and an external attenuator required only on very rare occasions. With such an amplifier, the gain control element can be placed in the ideal position, ie the feedback loop, ensuring the best possible noise performance.

...we have just described AD149's microphone amplifier.

Any audio-mixing system is, inevitably, and at all stages, a compromise between headroom, signal-to-noise ratio, dynamic range, sensitivity. We believe that we have achieved the state of SHIBUI.

Some boom-microphone operators dislike wearing combination headsets, preferring to use the boom microphone itself for talkback purposes. In this case, privacy is essential between boom operator and sound operator. Once a module has been assigned to the boom microphone, an internal DIL switch will guarantee the required privacy - a private line. This is a feature also available to a boom operator wearing the standard headset when a guaranteed private conversation between boom and mixer is required.

By a second internal DIL switch, the boom microphone may be routed to the monitoring section for audition, in isolation, by external 1 and/or external 2, without interfering with the sound-mixer's requirements.

For a mixer of its type and size, AD149 has an ultra-sophisticated auxiliary section. When sourced post fader and with level controls at maximum, auxiliary signals are at the same level as, and track with the main L/R output from the module and are controlled by the channel fader. In this case, auxiliary outputs are available even when the L/R output is switched off. It is normal practice for auxiliary level controls to act only as attenuators, but when sourced pre-fader, an extra 10dB gain is introduced into the auxiliary paths. (We believe this to be unique to AD149.)

An auxiliary stereo link converts auxiliary 1 level-control into a master control for both paths, and auxiliary 2 level-control into a panpot across the auxiliary outputs from the module.

By combining the last two features, AD149 can be configured not only as a four-output mixer but also as two independently-controlled stereo mixers in the same chassis.

Unusually for such a mixer, an 'insert' switch has been included in the microphone/line module for use during production. The availability of portable 8-track recorders enables up to eight microphone signals to be recorded separately, ensuring maximum flexibility during post-production. However, if the mixer and recorder are connected via the insert points (send AND return) rather than the channel outputs (clean feeds), by selecting appropriate insert switches, sample mixes may be auditioned at the time of production.

For stereo operation, linking bars are available for the faders on adjacent modules - rendering a stereo-microphone module totally unnecessary.

All existing evidence indicates that a stereo module is an exercise in compromise - a pair of mono modules with faders ganged for stereo operation will always offer more versatility and control (but not one-handed operation). Personal experience tells us that ganged equalisation is rarely what is actually required in the L-R domain, and is NEVER required in the M-S domain. Ganged equalisation is also, of necessity, less sophisticated than that which it is possible to include in a mono module.

All mixer inputs and outputs are electronically and mechanically suppressed against radio-frequency interference.

NOISE	MIC	<-126dB EIN	20Hz to 20kHz	200R SOURCE
			(with respect to the input)	
	LINE	>70dB SNR	20Hz to 20kHz	
EQUALISATION	HF:	±10dB @ 10kHz	VARIABLE SLOPE	
		±15dB MAX @ 15kHz		
	LF:	±10dB @ 100Hz	VARIABLE TURNOVER	
		±15dB MAX @ 30Hz		
	MF:	±15dB PEAK & DIP RECIPROCAL	Q=1.2	
		CENTRE FREQ = ±18dB @ 1k4Hz		
		MF1 = ±18dB @ 1k8Hz		
		MF2 = ±18dB @ 2k3Hz		
		MF1 + MF2 = ±18dB @ 3k8Hz		
	HPF 1:	PRE-TRANSFORMER	12dB/OCTAVE	
		-3dB @ 150Hz		
	HPF 2:	-3dB @ 90Hz	12dB/OCTAVE	
	HPF 1+ HPF 2 :		REFER TO CURVES	
OUTPUT LIMITER	THRESHOLD	+8dB		
	RATIO	7 : 1		
	ATTACK	2 SETTINGS	INTERNALLY SWITCHABLE	
		1 : 4ms	2 : 0.8ms	
	RELEASE	2 SETTINGS	INTERNALLY	
SWITCHABLE		1 : 250ms	2 : 80ms	
MIC/LINE LIMITER	THRESHOLD	+0dB		
	RATIO	7 : 1		
	ATTACK	COUNTER- CLOCKWISE	0.5ms	
		CLOCKWISE	10ms	
		CENTRE	4ms	
	RELEASE	COUNTER- CLOCKWISE	10ms	
		CLOCKWISE	700ms	
		CENTRE	300ms	

FACTORY SETTINGS FOR ATTACK AND RELEASE TIMES:

OUTPUT LIMITER: SLOW
MIC/LINE LIMITER: CENTRE

CURRENT CONSUMPTION 8 INPUT 370mA

TECHNICAL SPECIFICATION - MECHANICAL

AD149 is supplied with 6, 8, or 10 input modules. A separate meter bridge fitted with 4 VU or PPM meters is supplied to read outputs Left, Right, Aux 1 and Aux 2. The mixer's meters can then be dedicated to monitor outputs.

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

DIMENSIONS IN MILLIMETRES

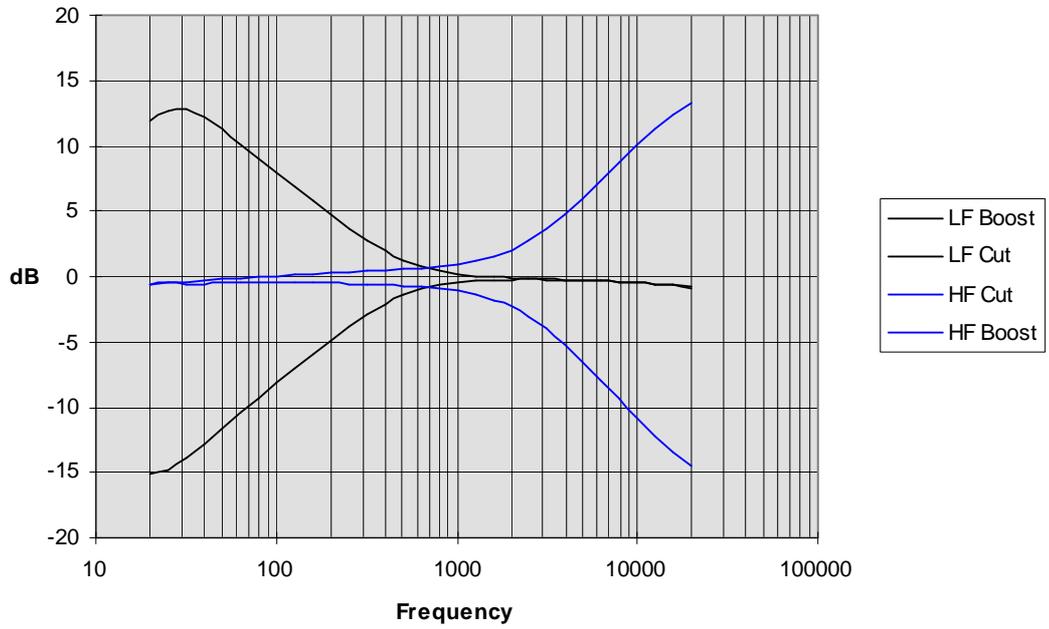
WEIGHT

6 I/P	8 I/P	10 I/P	
8.4	10.0	11.6	

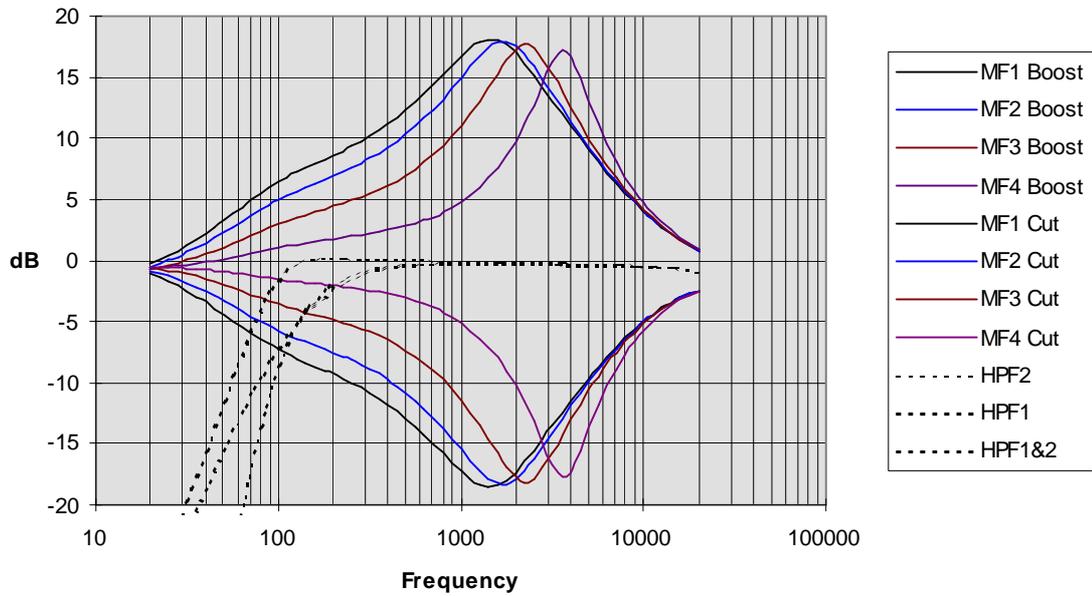
WEIGHTS IN KILOGRAMS

EQUALISATION CURVES

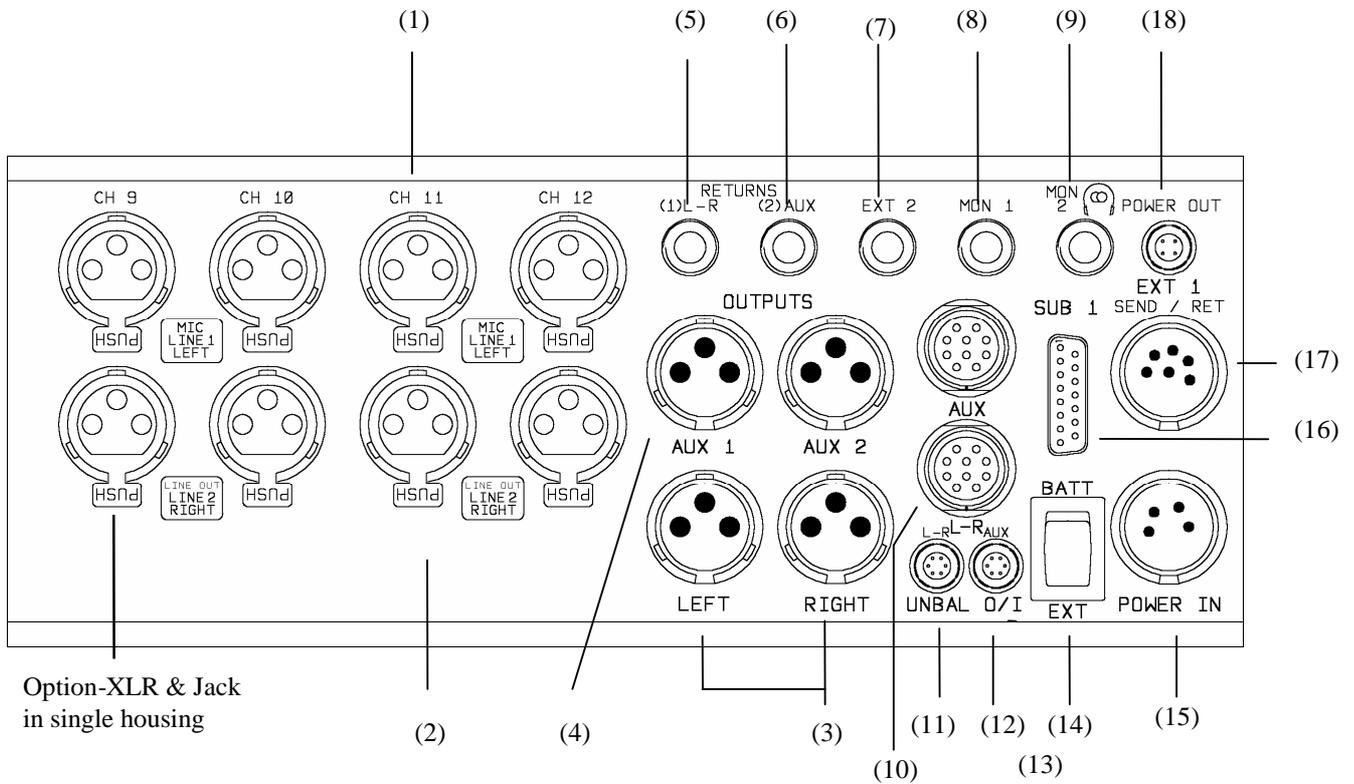
AD149 HF and LF EQ Plot



AD149 MF and HPF EQ Plot



CONNECTOR PANEL



Option-XLR & Jack
in single housing

- | | |
|--------------------------------------|--|
| (1) Microphone input | (10) Left-Right output/return multiway |
| (2) Line input | (11) Left-Right output/return connector (unbalanced) |
| (3) Main Left-Right output | (12) Auxiliary output/return connector (unbalanced) |
| (4) Auxiliary 1-2 output | (13) Auxiliary output/return multiway |
| (5) Left-Right return jack socket | (14) Off/On switch (external & internal) |
| (6) Auxiliary 1-2 return jack socket | (15) External DC power input |
| (7) External 2 output jack socket | (16) Subsidiary connector |
| (8) Monitor 1 output jack socket | (17) Talk send & return |
| (9) Monitor 2 output jack socket | (18) DC power output |

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

All inputs to, and outputs from AD149 are to be found on the connector panel.

Module connector (1) accepts balanced microphones, line 1, stereo left

Module connector (2) accepts balanced, line-level inputs, line 2, stereo right.

XLR (Inputs & Outputs)	Pin 1	Shield
	Pin 2	Signal +
	Pin 3	Signal -

In the case of an unbalanced line-inputs and outputs, pins 1 & 3 should be connected. This will not lead to a loss of level.

(When the option of separate connectors for microphone and line has been selected, the channel output/clean-feed signal appears on a mono, unbalanced jack in the centre of connector (2) - otherwise connector (2) carries channel output.)

Main (3) and auxiliary (4) outputs are transformer balanced, and are at line-level.

The unbalanced stereo-returns enter the mixer on two standard, 'A' type unbalanced stereo jacks; L-R (5) and AUX (6).

Unbalanced monitor 1 output is on a mono, unbalanced ('A' type) stereo jack; MON 1 (8).

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

EXT 2 (7) is a mono, unbalanced jack carrying monitor signals to a second outstation.

MONO UNBALANCED JACK		STEREO UNBALANCED JACK	
Tip	Signal +	Tip	Left Signal
Ring	Shield	Ring	Right Signal
Sleeve	Shield	Sleeve	Shield

Channel insert-jack: Tip = send; Ring = return.

The 6-pin XLR connector, EXT 1 (17), carries all signals to and from an outstation. The balanced return appears only on the headphones output.

XLR	Pin 1	Shield	Pin 4	Ret +
	2	Send Left	5	Ret -
	3	Send Right	6	Control

The unbalanced, send outputs have low-impedance with a capability of driving headphones of 25 ohms impedance or greater. (Refer to page 68)

The balanced (or unbalanced) signal from the outstation feeds on to the PFL mixing buss and is routed by grounding the control line (Pin 6) at the external source.

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

A three-position rocker switch BATT/EXT (14) selects either internal batteries or an external DC source. Power to the mixer is confirmed by the MIX ON LED on the output module. This LED flashes when the internal voltage falls below the safe operating level of 10V.

External powering of the mixer is via a 4-pin XLR (15).

XLR	Pin 1	0V	Pin 3	NC
	2	Charge	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.

SUB 1 (16) is a 15-way 'D' connector which may be used to drive an external meter bridge and supply the necessary DC voltages. The four meters read Left, Right, Auxiliary 1 and Auxiliary 2 outputs. When a meter bridge is being used, the mixer's meters may be routed 'permanently' to monitor 1 or PFL. (All signals are unbalanced.) SUB 1 is also employed when a recorder is to be controlled remotely.

	CONV	IDC		CONV	IDC	
PIN	1	(1)	Chassis	PIN	8	(15) +V voltage rail
	2	(3)	Spare		9	(2) 0V
	3	(5)	N/C contact - Remote		10	(4) PFL left output
	4	(7)	Common - Remote		11	(6) PFL right output
	5	(9)	N/O Contact - Remote		12	(8) Aux 2 output
	6	(11)	PFL control		13	(10) Aux 1 output
	7	(13)	-V voltage rail		14	(12) Right output
					15	(14) Left output

CONV refers to the normal numbering of a 'D' type connector, but note that some IDC types are numbered as indicated in ().

HIROSE connectors (10) → (13) are provided for single-cable connection to the mixer. (10) & (11) and (12) & (13) can be used simultaneously.

(10) carries the balanced L-R output and the unbalanced L-R return (stereo-return 1).
 (13) carries the balanced AUX 1 and AUX 2 outputs and the unbalanced AUX return (stereo-return 2).

	L-R		AUX	
Pin 1	Left	+	Aux 1 +	Signal
2	Left	-	Aux 1 -	Signal
3	Right	+	Aux 2 +	Signal
4	Right	-	Aux 2 -	Signal
5	Ret 1	R	Ret 2 R	Signal
6	Shield		Shield	
7	Ret 1	L	Ret 2 L	Signal
8	Shield		Shield	
9	Shield		Shield	
10	Shield		Shield	

(11) carries an unbalanced L-R output and unbalanced stereo-return 1 - for connection to a camera or DAT recorder.

(12) carries unbalanced auxiliary outputs and unbalanced stereo-return 2 - for connection to a camera or DAT recorder.

	L-R		AUX	
Pin 1	Left		Aux 1	Signal
2	Right		Aux 2	Signal
3	Shield		Shield	
4	Ret 1	L	Ret 2 L	Signal
5	Ret 1	R	Ret 2 R	Signal
6	Shield		Shield	

POWER OUT (18) is a 4-pin HIROSE connector to supply 12V DC @ 100mA to power external equipment, eg two non-diversity radio microphones.

Pin 1	Power ground		
2	Power ground		
3	DC	+12V	Switched
4	DC	+12V	Unswitched

Pin 3 depends on the position of the mixer ON/OFF switch.

Connector (18) may also be used for powering two or three PORT-A-FLEX units. Each unit has an 'in' and an 'out' power socket so that several units can be daisy-chained. Typical usage could be FLEX-EQ, a headphone distribution amplifier

and a four-output, stereo distribution amplifier.
BLOCK DIAGRAM - PART 1

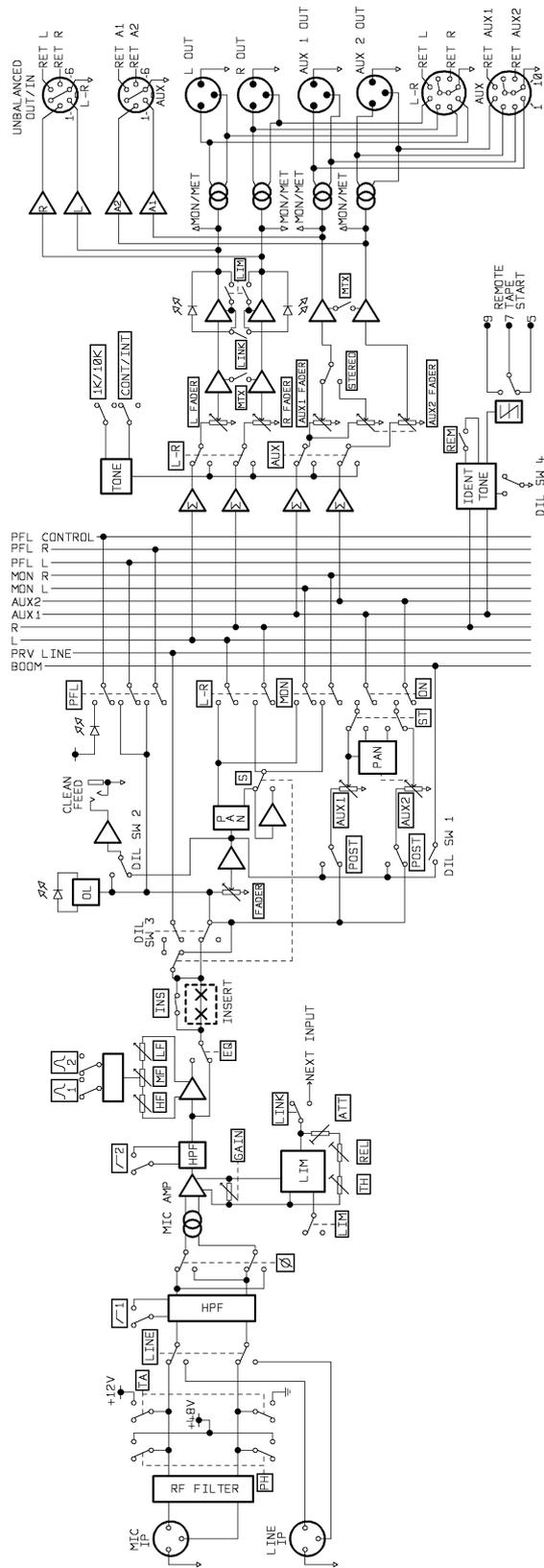
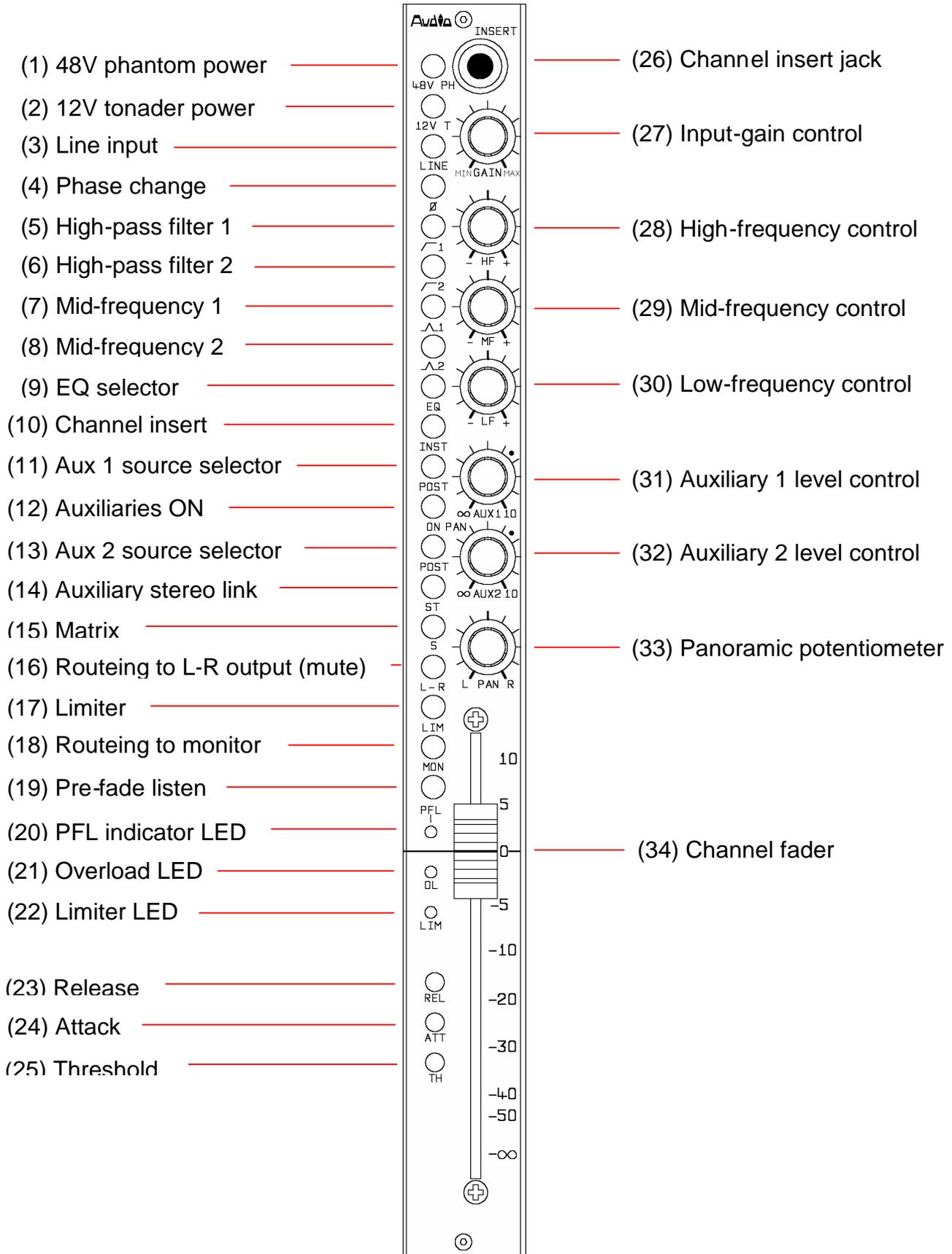


FIG 4

MICROPHONE/LINE MODULE



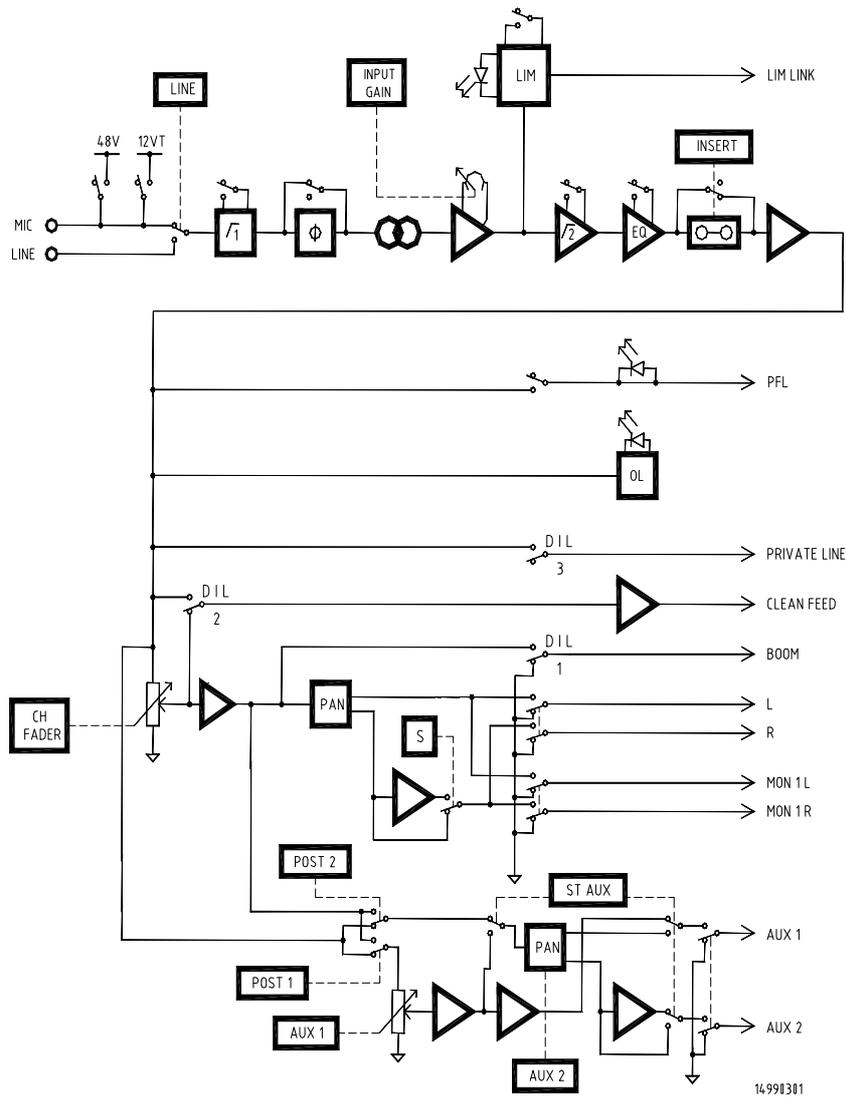


FIG 5

The microphone amplifier has been described in detail in the introduction. Switches (1) and (2) select 48V phantom and 12V tonader power for condenser microphones. The switches are interlocked with tonader taking precedence over phantom. Powering may be selected before or after connecting the microphone, but switch off all powering on unterminated modules to ensure unconditional stability of the mixer. When switched to line input (3), no damage will occur to external equipment should microphone powering remain selected momentarily.

Such is the isolation, a microphone and line-level input may remain connected simultaneously (when separate input connectors are employed).

Phase change (4) is pre transformer and operates on both microphone and line inputs.

The first high-pass filter (5) is pre transformer and, likewise, operates on both microphone and line inputs. In this position, the high-pass filter protects the transformer from low-frequency saturation caused by wind, traffic, air conditioning etc. The second high-pass filter (6) is placed after the microphone amplifier for correct and predictable combining of the two filters when selected simultaneously. Operating frequencies are to be found in the TECHNICAL SPECIFICATION - as are all parameter details. High-pass filter (HPF) 2 operates independently of the equaliser.

To control the gain of the input amplifier, a limiter may be switched into circuit (17) - limiting action being indicated by LED (22). Limiter release time (23), attack time (24) and threshold (25) may be adjusted externally: factory settings and values at the clockwise and counter-clockwise positions of the presets are detailed in the TECHNICAL SPECIFICATION. Despite the limiter on the input amplifier, overload can still occur in later stages. (The limiter also operates on a line input.)

On the mother board, a DIL switch will be found between module positions 1/2, 3/4 etc. When selected - to right - the DIL switch will link the limiters of the pair of modules for stereo operation. (The presets of each module should have the same settings for perfect parallel tracking.) The mixer is supplied with the modules unlinked.

All limiter-link DIL switches are accessed by removal of the back panel covering the mother board. (There is no requirement to remove input modules.)

It is ESSENTIAL that X, M, L signals are inserted in odd-numbered channels and that Y, S, R signals are inserted in the even-numbered channel ABOVE.

Following the microphone amplifier is the equaliser - switch (9). Within the equaliser, it is possible to select - (7) and/or (8) - four frequencies for the peak/dip mid-frequency band.

Because of the inherent inaccuracy of the centre-detent mechanism, it is not Audio Developments' practice to fit such devices to equaliser amplitude controls - HF (28), MF (29), LF (30).

Following the equaliser is an insert point, accessed via the standard 'A' type jack at the top of the module. Because of lack of PCB and panel space, input and output are, of necessity, unbalanced. Tip = send; Ring = return.

The channel insert switch INST (10) enables external equipment to remain permanently connected to the channel. The insert socket is wired in order that a pre-fader channel signal is always present on the jack TIP. Thus at all times, two output signals are available from the module; pre fader from the insert point and post (or pre) fader from the clean feed. A possible and unusual usage during production has already been detailed on page 25.

The two auxiliary paths have a common ON switch (12), but each path has its own pre- or post-fader source selector (11) and (13). The level controls (31) and (32) are scaled from 00 to 10 with a calibration mark at zero. If sourced post fader the level control acts as an attenuator and when at maximum, auxiliary signals track with, and are at the same level as the module's main output (L/R). Under no circumstances does auxiliary operation depend upon channel routing, ie it is independent of the channel mute (16).

When sourced pre fader, an additional 10dB gain is available in the auxiliary signal paths when the level controls are set to their calibration dots. When the channel fader is on its calibration mark, auxiliaries may be switched between pre and post fader without any change in signal level.

The auxiliary stereo link - ST (14) - configures the two auxiliary paths as a 'stereo' pair: the auxiliary 1 level control becomes the master gain potentiometer and the auxiliary 2 level control pans the module signal between auxiliary 1 and 2 outputs. When in 'stereo' mode, switch (11) is the source selector, switch (13) being redundant.

Thus, post equaliser, and with auxiliary 1 sourced pre fader, AD149 is two independent stereo mixers in one chassis where auxiliary 1 level control effectively becomes a second channel 'fader' with a gain reservoir of 10dB.

The unbalanced, post-fader channel output or clean feed is available from CONN (2). By special request, the mixer may be fitted with separate connectors for microphone and line inputs. In this case, the channel output will appear on the jack in the centre of the LINE input connector. The channel output can be selected to pre fader via DIL switch 2 on the main PCB.

'S' (15) enables a matrix amplifier to be created across adjacent pairs of input modules. This feature receives in-depth treatment elsewhere in the manual. 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 phase of the S-microphone channel will rectify the situation - without having to disturb the microphone.
But see, also, M-S NOTES.

The routing switch (16) to the left and right mix busses is, in effect, the channel mute.

Panoramic potentiometer - panpot - (33) routes the module's signal proportionately between the left and right mix busses.

MON (18) routes the true (L/R) output from the module to the monitor module where it can be treated as an AFL (after-fader listen) signal or as an SIP (solo-in-place) signal. MON is independent of L-R routing (16) and is derived from the output of the panpot and 'S' switch (15) if selected.

Pre-fader listen (19) with its LED indicator (20) is a third way of aurally monitoring the module.

The overload LED (21) illuminates 3dB before clipping at the input to the fader.

The Penny & Giles fader (34) 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.

BOOM

Once a module has been assigned to a boom microphone, DIL switch 1 mounted on the module's sub-board will route the signal to the monitor module where it can be independently selected to either or both of the external circuits, eg to boom operator and director/producer. This feature is operative when the DIL switch slider is moved to the right. Once this assignment has been made, we suspect it will remain in place during the lifetime of the mixer.

PRIVATE LINE

It has come to our attention that many boom operators dislike having a personal microphone attached to their headset - instead, they prefer to use the boom microphone itself for communication with the sound mixer. (This, of course, precludes conversation during programme.) Under this system, complete privacy must be guaranteed ...

To create a private line between boom operator and mixer, DIL switch 3 on the main PCB of the (boom microphone) module should have its slider moved downwards. (This does not interfere with normal operation of the module.) At the end of a take, pressing 'S' (15) - S=Safety - isolates the boom microphone from all sections of the mixer except the operator's headphones jack and **GUARANTEES COMPLETE PRIVACY** of communication between EXTERNAL 1 and sound operator.

MIC/LINE MODULE PCBs

Note: By only one from the position 1 ON. The any other

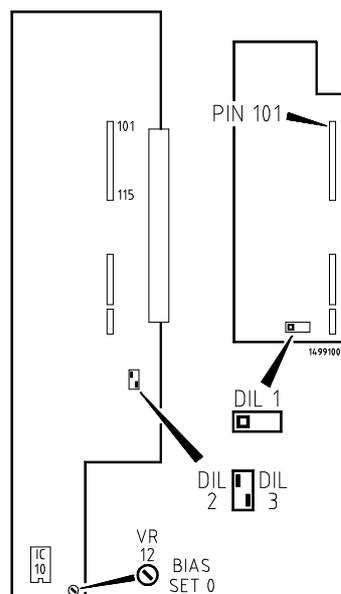


FIG 6

request, DIL switch 3 is now fitted to module. The mixer is despatched factory with this module installed in in the chassis with the facility set module may, of course, be placed in position in the chassis.

LIMITER CALIBRATION

Refer to page 49, #1 for use of external test equipment.

Switch off the limiter and set the presets for Release, Attack and Threshold to the centre. Set the bias preset on the main PCB - labelled SET 0 on FIG 6 - to its centre.

Apply a 1kHz signal to the channel for a reading of 0dBu at pin 101 indicated on FIG 6. Turn SET 0 preset counter-clockwise until the reading at pin 101 reduces by not more than 0.2dB.

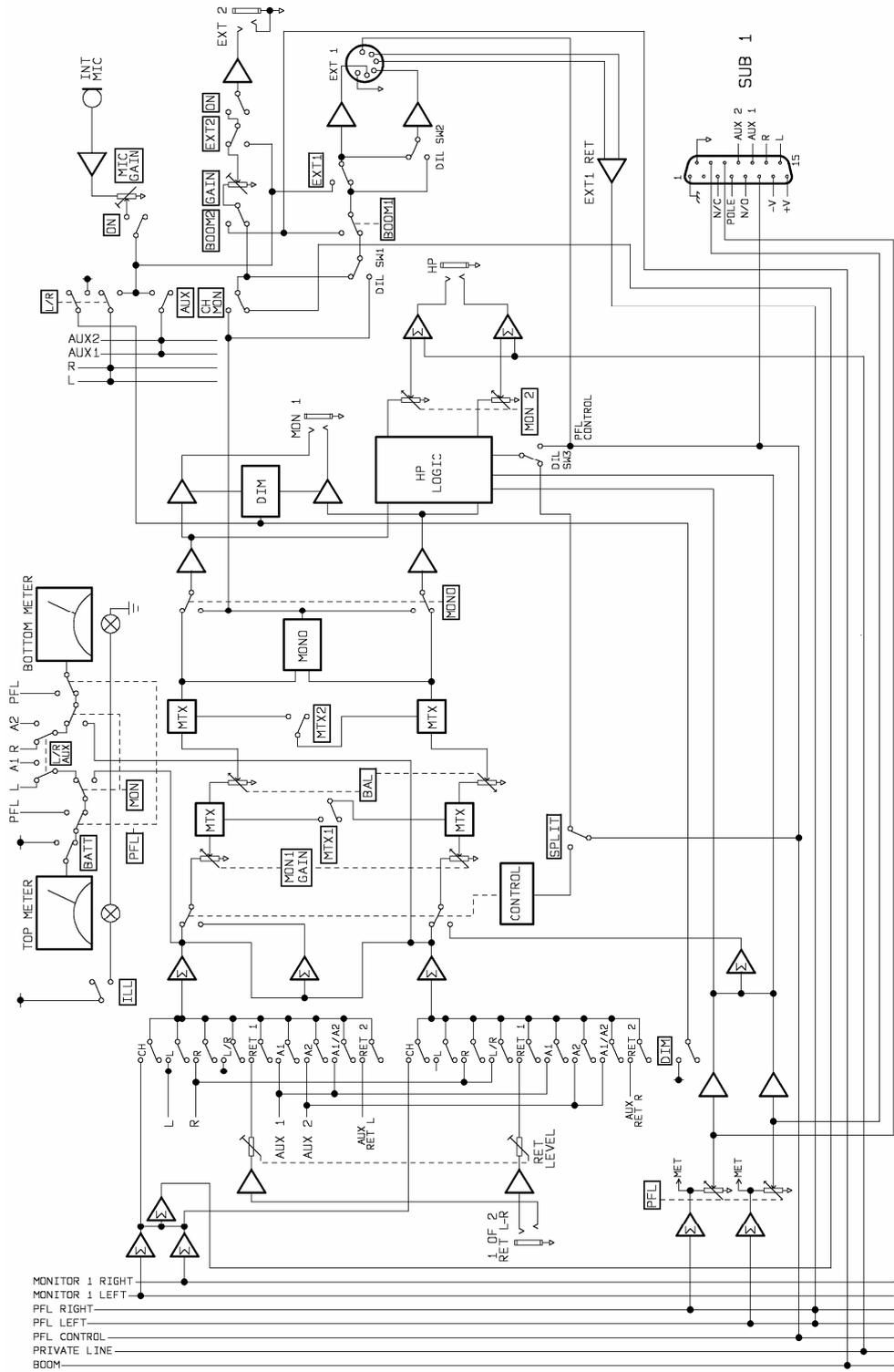
Increase input-gain control (27) for a reading of +1dBu at pin 101 and switch in the limiter (17). Turn the threshold control until the reading at pin 101 is reduced to 0dBu - at which point the limiter LED (22) should illuminate.

This is the setting requested by customers, leaving 18dB before clipping within the channel. We believe that this setting is over-cautious and unfortunately destroys input headroom which the limiter is actually intended to protect.

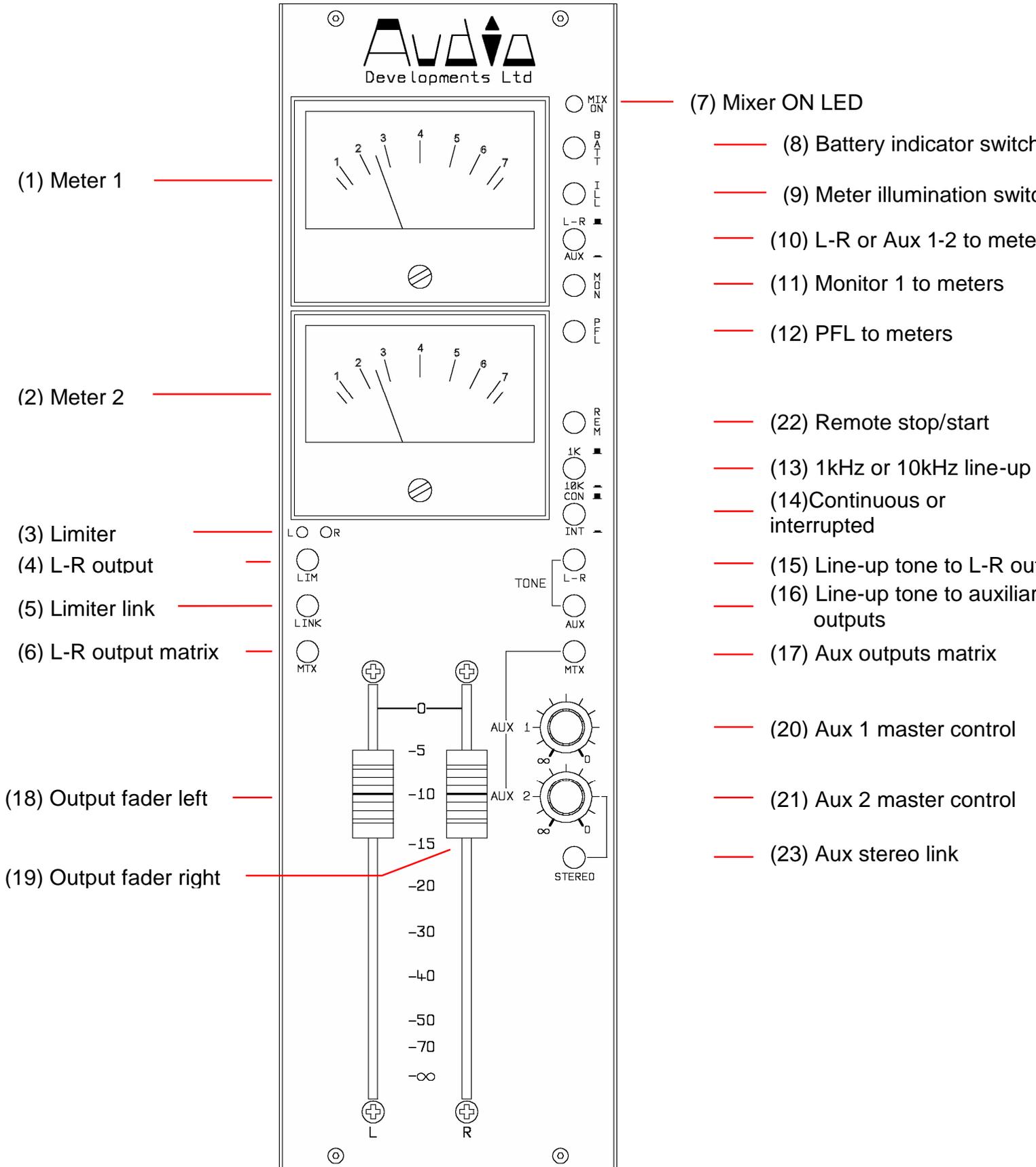
The range on the threshold preset is -5dBu (ccw) to +15dBu (cw). If set at +15dBu, the limiter and overload LEDs would operate together. For optimum use of the limiter, we suggest it be set at +12dBu, ie 6dB before clipping.

(The DC voltages on IC10 should be approximately -2.6V at pin 10 and -2.0V at pin 12, ie a difference of 0.6V)

BLOCK DIAGRAM - PART 2



OUTPUT MODULE



This module controls all metering of PFL and output signals, and also the level of main and auxiliary output signals. Calibration is achieved when the faders and auxiliary potentiometers are at maximum.

The upper meter (1) reads left output, auxiliary 1, monitor 1 left, PFL left and the lower meter (2) reads right output, auxiliary 2, monitor 1 right, PFL right - by use of switches (10), (11), (12). When no switch is selected, meters read main L-R output; thereafter, switch (11) overrides switch (10) and switch (12) overrides switches (10) and (11). Switch (11) reads the signal entering monitor 1 path; a signal that can be either in the L-R domain or in the M-S domain. Battery status is read on the upper meter by BATT (8) and the meters may be illuminated by ILL (9) - overuse of this function will rapidly drain the batteries. When the voltage drops below a safe operating level, the MIX ON indicator LED (7) will flash.

Output signals to be metered follow output faders, matrix amplifiers and limiters.

When the auxiliary paths are being used as a stereo pair, their output potentiometers (20) and (21) may be linked (23) for one-handed operation - AUX 2 becoming the stereo master. A matrix amplifier may be inserted across auxiliary 1 and 2 outputs, MTX (17).

Separate Penny & Giles output faders (18) and (19) are employed for maximum flexibility, but may be linked by a ganging clip.

A pair of limiters may be switched into the main L-R output - LIM (4) - and linked for stereo operation - LINK (5). A choice of fast or slow attack and release times may be made via the two DIL switches on the left-hand PCB. When the limiters are being used as a stereo pair, they should have their attack times and release times set identically. LEDs (3) indicate when limiting is taking place. A matrix (6) may be inserted in the main L-R output path.

LEFT-HAND PRINTED CIRCUIT BOARD

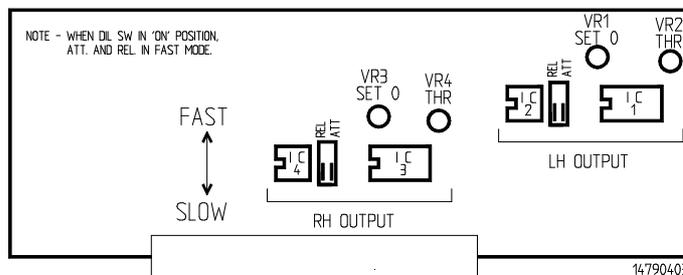


FIG 12

Line-up tone, when selected, replaces the normal signals on the main, L-R output (15) and on the auxiliary outputs (16). The frequency of line-up tone may be selected between 1kHz and 10kHz (13). Tone is normally continuous on L & R and on AUX 1 & 2 or interrupted tone, in 3-second bursts, may be selected to L and AUX 1 by switch (14). By the simple addition of two links, cutting a track and removing a through-pin on the right-hand PCB, the user may (more logically) place the interrupted tone on the R and AUX 2 outputs. Line-up tone is calibrated on the (BBC-scaled) meters to PPM4, giving a line-output level of 0dBu with L & R and AUX 1 & 2 controls set to maximum.

Alternative calibration levels are possible via the internal preset.

RIGHT-HAND PRINTED CIRCUIT BOARD

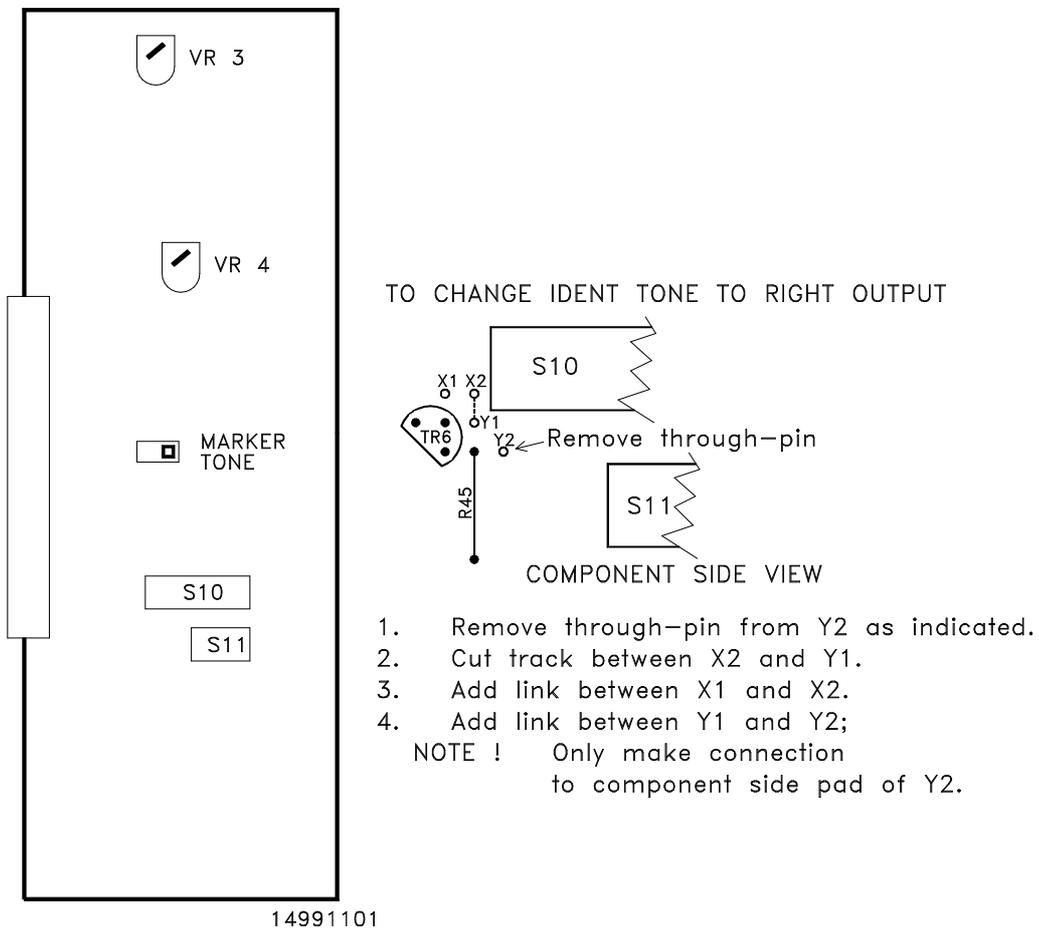


FIG 13

REMOTE

REM (22) is a latching switch to start and stop a tape recorder fitted with a remote stop/start function. Via a DIL switch on the right-hand PCB, it can be arranged for each recording to be 'topped and tailed' with a marker, ie a few seconds of low-frequency tone. To INHIBIT the marker tone, set DIL switch to the left. (Refer to FIG 13)

The remote stop/start facility is accessed via the 'D' connector - SUB 1:

Pin 3	normally closed
Pin 4	common
Pin 5	normally open

Some recorders require a pair of contacts to be opened - in which case use pins 4 & 3, and some require a pair of contacts to be closed - in which case use pins 4 & 5.

NOTE

1 A common timing circuit is used for interrupted line-up tone and remote marker-tone. It is recommended, therefore, that the interrupt button, INT (14), is not operated during recording.

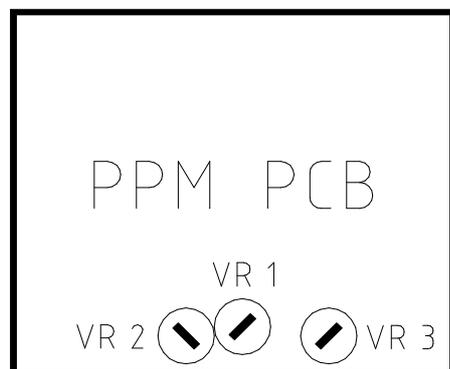
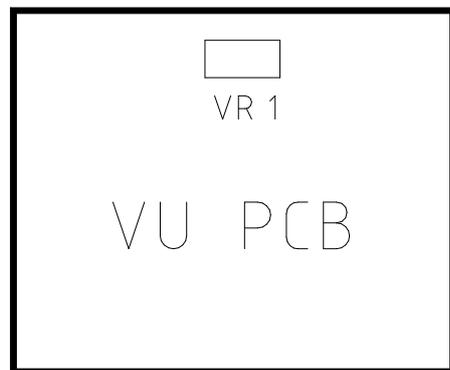
2 Care should be taken when wiring the REMOTE at the 'D' connector. There are two conventions in use - one for the common wire type, the other for IDC. Refer to connection details on page 33.

ADJUSTMENTS AND CALIBRATIONS

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

Line-up-tone oscillator - the preset (VR4) 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. (Refer to FIG 13)

Low-battery indicator - the preset (VR3) associated with this facility is mounted on the right-hand PCB of the output module. The low reading is set at the factory and corresponds to a set of batteries having discharged to 10V. On a VU meter, low volts is at 0VU. On a PPM meter it is indicated by a separate marking: yellow on a BBC scale and red on an N10 and SDR scale. (Refer to FIG 13)



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FIG 14

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 meter - the 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.

Main-output limiters - calibration involves two presets per output. Using the 'L' output as the example and referring to FIG 12 ...

Biasing and threshold are adjusted as follows: VR1 biases the limiter circuit to the point of correct operation and VR2 sets the threshold. (Labelled SET 0 and THR respectively on the left-hand PCB.)

Determine whether the left-hand PCB is issue 3 and below or issue 4 and above. If issue 4+, read 'PIN 5 of IC2' instead of 'PIN 12 of IC1' in the following paragraphs.

(Issue 4+ does not indicate an early improvement in circuitry; simply a change in layout to make testing simpler.)

With no signal present, VR1 should be adjusted to give a reading in the range -1.5V to -2.5V at PIN 12 of IC 1. VR2 should be adjusted to give a reading in the range -2.5V to -3.5V at PIN 10 of IC 1.

Apply a signal at 1kHz to the mixer to give a level of 0dBu at 'L' output. Then select the limiter function and adjust VR1 until the output signal starts to fall (typically 0.2dB). The DC voltage at PIN 12 of IC 1 should be approximately -2.0V.

After setting VR1, adjust the output signal to just greater than +8dBu with the limiter deselected. Introduce the limiter and adjust VR2 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 IC 1 should be approximately 0.6V different from that at PIN 12.

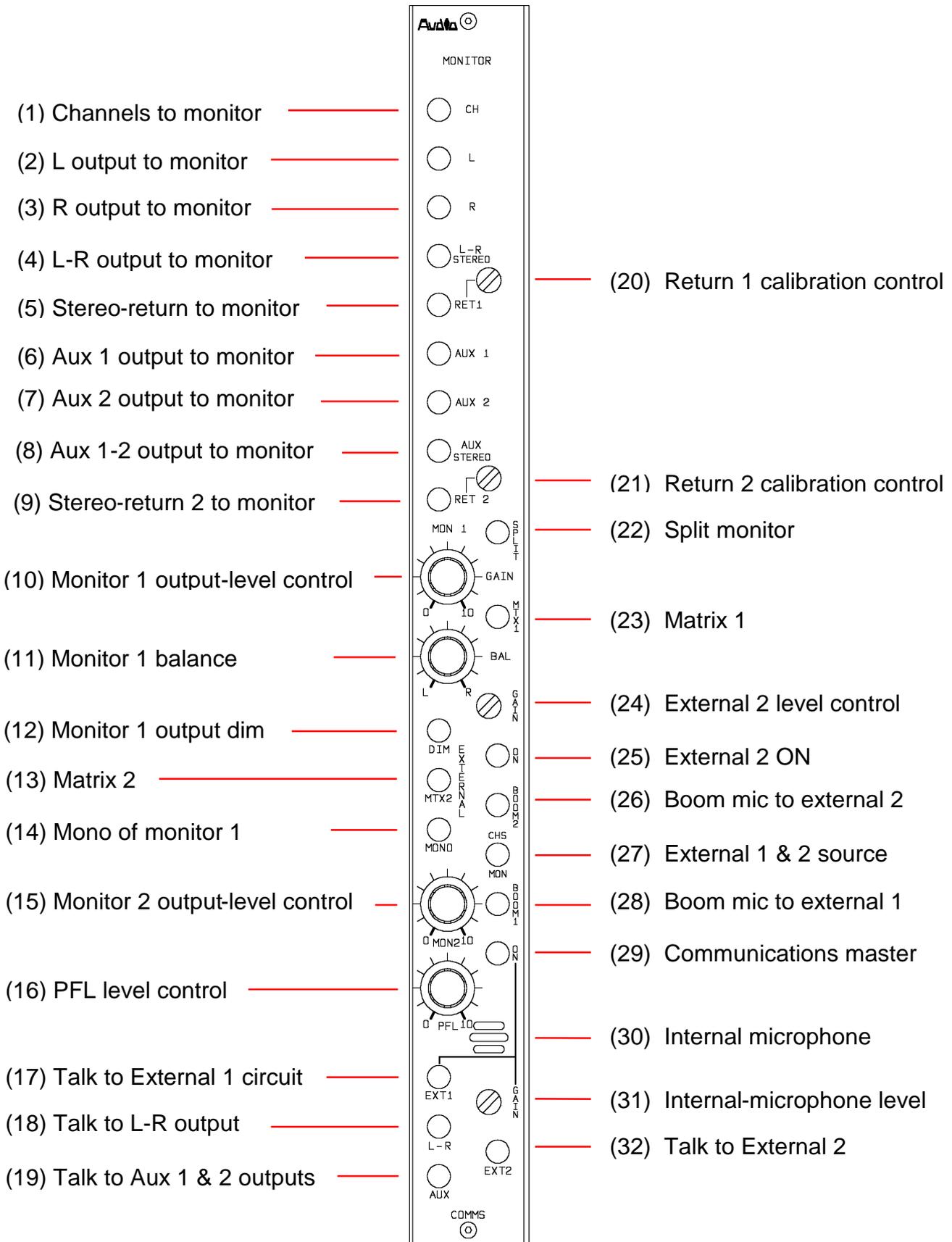
The right output can be set using the above procedure, but the preset and IC numbers change to correspond with the output being calibrated.

The link function has no individual setting of its own. To ensure this works within specification it is important that L and R outputs are set up as a pair. After following the above procedure, the DC voltages at PINS 10 and PINS 12 of the ICs should be identical. (Tolerance $\pm 0.05V$ with typical figures at PIN 10 of -2.6V and PIN 12 of -2.0V)

There are two DIL switches associated with the attack and release times: one for the left output, the other for the right output.

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MONITOR MODULE



The monitor module is associated with all aural monitoring of input and output signals together with communications between the sound operator and two outstations - EXTERNAL 1 & 2.

(Metering of monitor 1 is performed pre fader and matrix amplifiers. Metering of PFL is performed pre fader.)

It is necessary to be able to audition (in left-right stereo, ie a form the human brain understands) an X-Y or 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 and auxiliary outputs, 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 (unbalanced) outputs of monitor 1 and monitor 2 will be 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 the main left-output from the mixer to monitor 1 (& 2) left and right outputs.

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

Switch (4) selects the main stereo-output from the mixer to monitor 1(& 2).

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 main output from the mixer. L+R [(2)+(3)] produces the same result.

Switch (5) routes the calibrated (20), stereo-return 1 to monitor 1 (& 2). The return will probably be the signal from the playback head of the tape recorder. By alternating adjacent switches (4) and (5), A/B comparisons to and from tape may be made. [Stereo-return 2 (9) - with its calibration control (21) - is identical with stereo-return 1.]

(Stereo-return 1 is also called L-R return and stereo-return 2 is also called auxiliary return.)

Switches (6), (7), (8) provide the same functions for the auxiliaries as switches (2), (3), (4) do for the main mixer output.

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

CH (1) operates in conjunction with the MON switch on the channels. This MON signal is the same as that which drives the left and right 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 L-R STEREO (4) and routing the appropriate microphone channel to MON, the question is easily answered. This is non-destructive SIP (solo-in-place) usage of the monitor 1 path, rather than 'destructive' in the recording-studio sense.

A mix, different from the main mix, may be set up on the monitor 1 path using CH (1) only - and independent of the L-R channel routing. (This could be a mix-minus.) The (identical) contribution of any channel to both the mix-minus (or monitor mix) and main output will be controlled by the channel fader.

The output from monitor 1 path (and to monitor 2 output) is controlled by ganged potentiometer (10). It is now possible to insert a 'Blumlein Loop' into monitor 1 path - MTX 1 (23), MTX 2 (13) with a zero-to-zero balance (width) control (11) between the two matrix amplifiers. When the balance control is fully counter-clockwise (L), only the LH signal appears on monitor 1 (& 2) outputs. Similarly, when fully clockwise (R), only the RH signal appears on monitor 1 (& 2) outputs. Selecting MONO (14) will place these LH & RH signals on both sides of monitor 1 (& 2) outputs - thus can PFL be isolated on monitor 1 output when SPLIT (22) is selected. (See last #, this page). The balance control may also be used to compensate for an off-centre position when monitoring on loudspeakers.

At ANY TIME during recording or post-production, a (monitor) signal in the L-R domain may have its potential evaluated by converting to M-S with MTX 1 (23) and width changed before re-converting to L-R, MTX 2 (13). Also when in the L-R domain, the monitor signal may be steered, BAL (11), to match a visual image. MTX 2 (13) will also reveal mono compatibility (M) of a L-R stereo signal on monitor 1 left, and out-of-phase components (S) on monitor 1 right. Both of these signals may be isolated using the BAL control, perhaps in conjunction with the MONO (14) switch.

Similarly, any M-S signal appearing on monitor 1 may have its M/S relationship changed using the monitor BAL control (11) - without affecting any previous settings - before being converted to L-R by MTX 2 (13).

Operating on monitor 1 is a dim control (12) which lowers the output by 20dB. This dim function is automatic when the internal microphone (30) is routed to L-R (18) and/or AUX (19) outputs. Dim does not affect a monitor 1 signal that is routed to the headphones.

The relationship between monitor 1, monitor 2 (headphones) and PFL has been fully described on pages 22 & 23. Split monitoring is a facility unique to AD149 and operates on both monitor 1 and monitor 2 outputs - the function being selected by SPLIT (22). Split monitoring is set at the factory to operate simultaneously on monitor 1 and monitor 2 when SPLIT (22) is selected, but if not required on

monitor 2, this may be disabled by DIL switch (3) on the left-hand PCB. (Refer to FIG 15) Use of split monitoring makes no change to the meters when reading monitor 1, but note that a signal entering monitor 1 path in the M-S domain will mix to monitor 1 left as a stereo-left signal - as it will when MONO (14) is selected.

Once the correct level has been set for monitor 1 output (10), the level of this signal to the headphones can be further adjusted by the monitor 2 level control (15). The return signal from EXTERNAL 1 is fed to the PFL mix busses and PFL level to monitor 2 is set by potentiometer (16) - to be balanced against monitor 1 level on monitor 2 path.

MONITOR MODULE PCBs

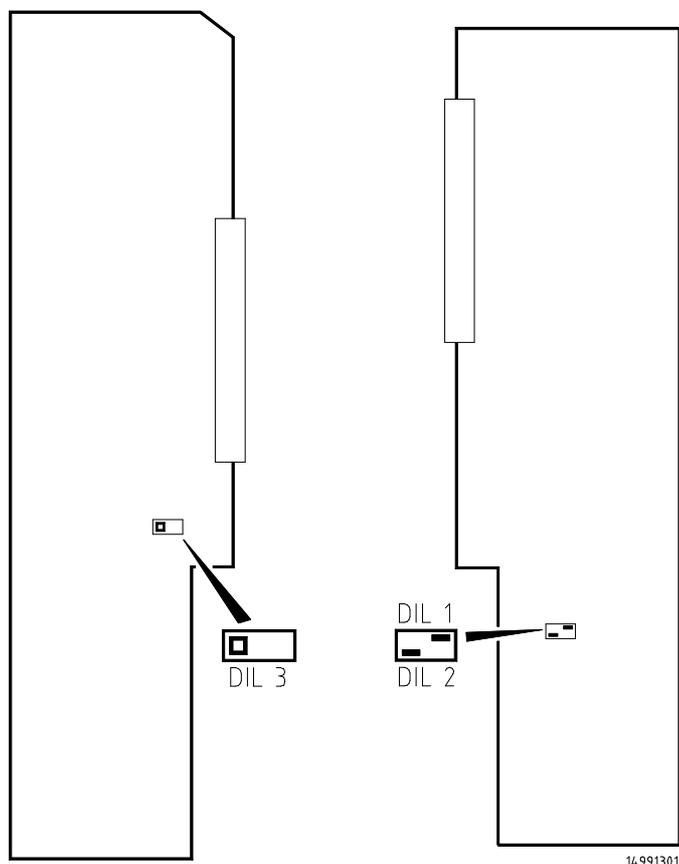


FIG 15

DIL 1 - selects MON 1 only facility to EXT 1 - bypassing CH/MON (27) switch

DIL 2 - selects talk + monitor facility on EXT 1

DIL 3 - selects facility of SPLIT MONITOR (22) on monitor 2 output

EXTERNAL 1 & 2

AD149 has the capability of sending signals to two outstations - external 1 and external 2. It is suggested that external 1 be used by the boom operator as it has the facility to talk back to the mixer - it also lacks a level control and on/off switch as these will be available on the operator's belt pack. External 2 has been designed to drive headphones for use by a director/producer etc. External 2 has its own on/off switch (25) and screwdriver-operated level control (24) but has no provision for talking back to the mixer.

At all times a monitor signal will be present on external 1 output. Selected by DIL switch 1 on the right-hand PCB of the monitor module, there is a choice of two signals:

- (a) a mono mixdown of the signal appearing on monitor 1 output. In view of the fact that, during production, the sound operator is more likely to be interested in PFL, monitor 1 output may be customised to the boom operator's requirements - probably the main output, so that he may audition his microphone within the overall context. (This will, probably, also satisfy the sound operator's requirements - monitoring the final mix, together with PFL override.)
- (b) the same signals that are available to external 2, but independently and externally selected. There is a choice CHS/MON (27) between MON as (a) above and CHS, which is a mono mix of all channels selected to MON, but WITHOUT CH (1) necessarily being selected. (Refer to FIG 16)

In the case of external 1, BOOM 1 (28) will override the selection made on CHS/MON (27) - and route the boom-microphone to the operator's headset. In the case of external 2, BOOM 2 (26) will override the selection made on CHS/MON (27) - and route the boom microphone to the director's/producer's headphones (This facility must already have been selected on the boom-microphone module. Refer to microphone/line module section.)

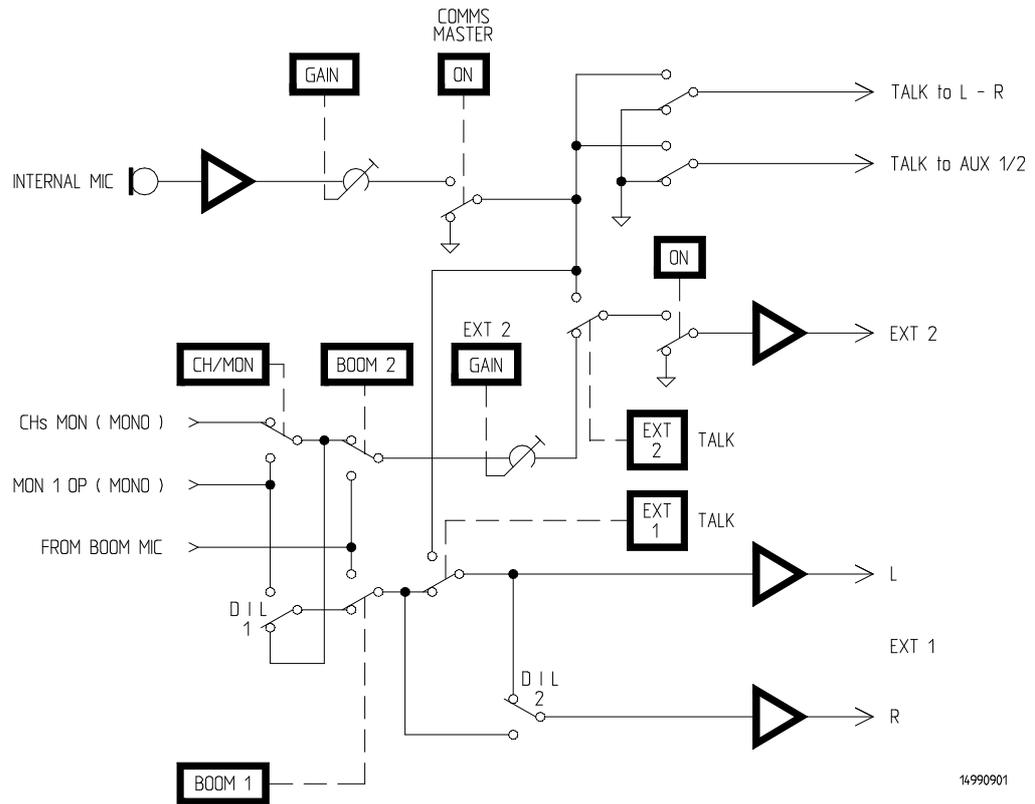
External 1 has the additional capability of its own split monitor, selected by DIL switch 2 on the right-hand PCB of the monitor module. Whenever communication is taking place from mixer to external 1, the monitor signal remains on the right earpiece and communication appears on the left earpiece. (Refer to FIG 16)

COMMUNICATIONS

In order that the sound operator can communicate with any output, the safety switch (Comms Master) ON (29) must be selected: L-R (18) routes the internal microphone to the main output and AUX (19) routes the internal microphone to both auxiliary outputs. [Switches (18) and (19) have a momentary action, ie they are non-latching.] EXT 1 (17) and EXT 2 (32) route the signal from the internal microphone to replace any signal appearing on EXT 1 and EXT 2 outputs. The level of the internal microphone is controlled by preset (31).

Communication from EXT 1 to the sound operator is initiated at the outstation and feeds the PFL mix busses. Like other PFL signals, it is under logic control and replaces any other signal on monitor 2 output, ie the operator's headphones. The level of this return signal is controlled by potentiometer PFL (16).

FIG 16



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MIXING

During recording/production there are eight possible signal paths through the mixer: inputs to the module in M-S or X-Y; mixing in the M-S or L-R domain; mixer output in M-S or L-R. (Refer to FIG 17) Whichever path is selected, no constraints will be placed on the auxiliaries as mixing in the M-S domain is a legitimate technique. When auxiliaries are not being used for usual applications, they may be used to create a mix different from the main mix; or identical with it - but in the opposite domain. A recording made in the M-S domain will be ready for the application of all possible post-production techniques.

Of the eight paths through the mixer, we believe that numbers 1, 2 and 3 will be the ones most frequently used, with 4 & 4a having no likely usage. (When using stereo microphones, it must again be stressed that odd-numbered modules are occupied by the X/A/M microphone and that even-numbered modules are occupied by the Y/B/S microphone.) A combination of M-S, X-Y and spot microphones may be mixed either in the L-R domain or the M-S domain. (In each case the stereo microphone must be matrixed to the appropriate mixing domain.) When all inputs are in X-Y form, mixing will normally be done in the L-R 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.

Any M-S microphone (or input) will be in the M-S domain at the pre- and post-fader auxiliary points. M-signals may be mixed on AUX 1 and S-signals on AUX 2, then matrixed to L-R on the auxiliary outputs. Alternatively, a L-R signal may be recorded from the main output, and an M-S signal from the auxiliary outputs. Spot microphones would be added only to the AUX 1 mix.

When mixing in the L-R domain and using one or more M-S stereo microphones, the panpots on M- and S-channels will be centred and the S-switch (15) selected on the S-channels to create a matrix across the M- and S-channels. With CH (1) selected on the monitor module, MON on the M-channels will audition an M-microphone on monitor 1 L & R (and headphones L & R) outputs. When 'setting up' M-S microphones, the S-channels may be monitored in the same manner - BEFORE the S-switch is selected. After selecting the S-switch, the complete M-S microphone may be monitored - automatically - in L-R stereo. At this stage, the potential of these M-S microphones may be checked by changing the M/S relationship, ie by varying the input gain of the S-channels.

X-microphones will be panned hard left, and Y-microphones will be panned hard right.

Spot microphones will be panned to their correct position in the final L-R image.

Earlier we mentioned conflicting monitor requirements; a problem arises when monitoring an S-channel after its S-switch has been selected. In this case +S would appear on MON left and -S on MON right - not very useful! To overcome this difficulty, swing the monitor BAL control (11) hard left and select MONO (14), thus isolating the in-phase signal from the S-channel. Alternatively, and if the engineer has requested a restricted swing on the BAL control (11), selecting either of the MTX switches and MONO (14) will achieve the same result.

When mixing in the M-S domain, M-channels will be panned hard left and S-channels will be panned hard right. Spot microphones will be panned left to mix with M-channels only. Any X-microphone will be panned centre and any Y-microphone will be panned centre and have its S-switch (15) selected - thus converting the X-Y microphone to the M-S domain.

MON on an M-channel will audition the M-signal on monitor 1 left output, and MON on an S-channel will audition the S-signal on monitor 1 right output. By selecting MONO (14) on the monitor module, each of these signals will appear on monitor 1 left and right outputs. If the M- and S-channels are selected simultaneously to MON, the resultant (M-S) signal is incomprehensible, but can be converted to the L-R domain by selecting MTX 2 (13) on the monitor module. By using the monitor BAL control (11), this L-R signal can now have its width changed and potential realised without changing channel settings.

An M-S main mix may be matrixed to the L-R domain for the final output from the mixer by MTX (6) on the output module (before any final limiting).

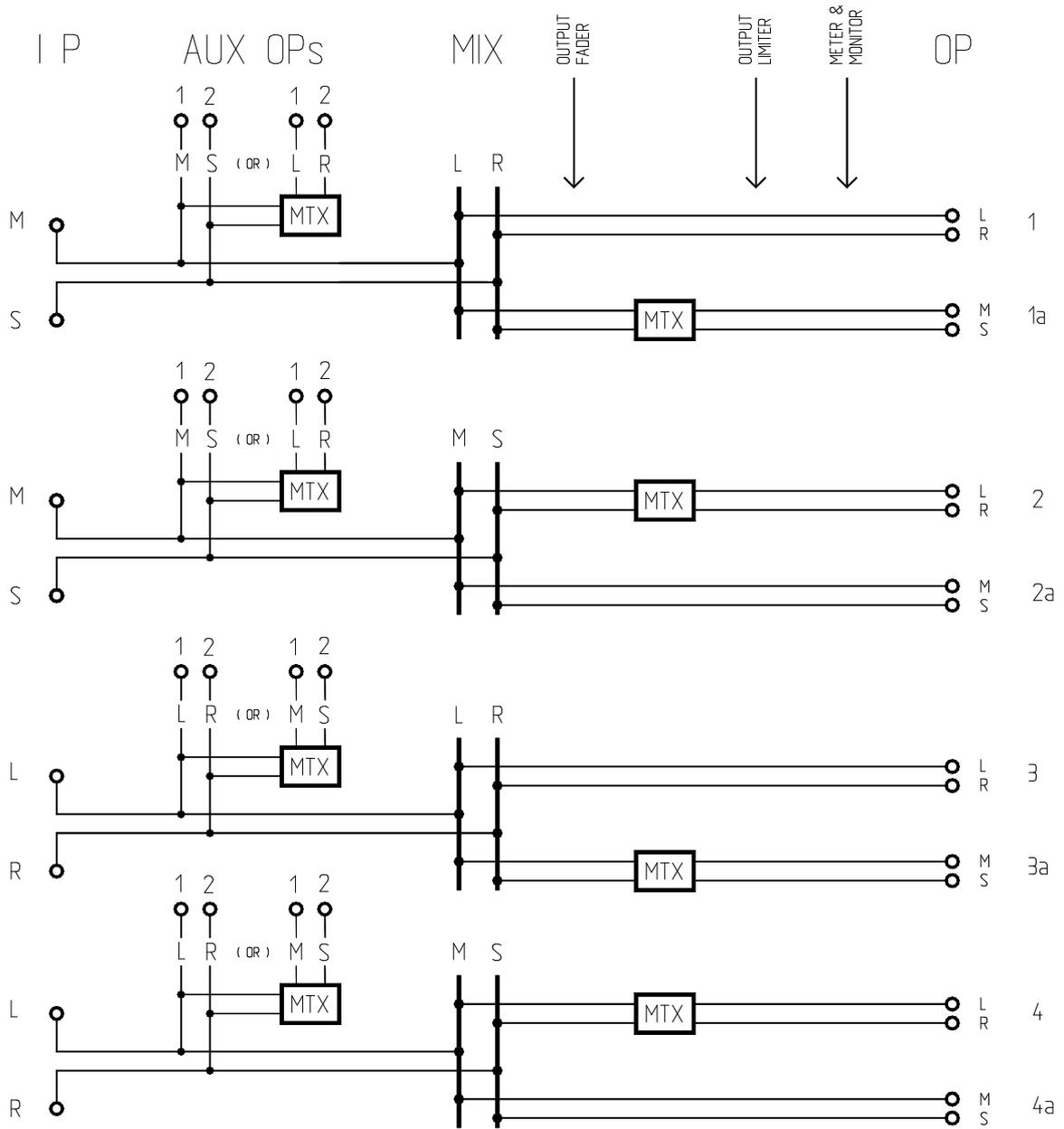
When mixing in the L-R domain without M-S microphones, X-Y (A-B) microphones will be panned hard left (X) and hard right (Y), and any spot microphone will be panned to its correct position in the final L-R image.

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.

When monitoring an X-Y microphone, MTX 1/BAL/MTX 2 will reveal the potential of angular changes to the microphone - or possibly a change of position - without having to gain access to the microphone.

(All auxiliary sends will be in the L-R domain.)

MIXER PATHS



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FIG 17

PLAYTIME

The following experiments may be performed on the monitor module. Select a couple of L-R recordings; one in natural stereo and with a centralised soloist, and one in unnatural, panpotted stereo.

The object of the exercise is to compare and contrast the two possible mono reductions. The simplest method is to add the left and right signals - but remember those nasty phase distortions. The second method is to matrix the L-R input signal to M-S and extract the M-signal only. At this stage, take the opportunity of also auditioning the out-of-phase components - S-signal. (Throughout the experiment, note the effect of introducing the phase inversion switch on the right input signal.)

Compare the vitality of the M (mono) mix from the M-S domain with the dull and rather lifeless mix from $L+R=MONO$.

If a stereo line module is available, creation of a Blumlein Loop will allow both the effect of phase inversion on the S-signal and the differential application of equalisation within the M-S domain to be appreciated.

Refer to page 11 - #2 and #3.

DIL SWITCHES

MOTHER BOARD

Input limiters	Link	→
----------------	------	---

MIC/LINE MODULE

Main PCB

Clean feed	DIL 2	Pre fader	↓
Private line	DIL 3	Select	↑

Sub-board

Boom	DIL 1	Select	→
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MONO LINE MODULE

Main PCB

Clean feed	Pre fader	→
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OUTPUT MODULE

Left-hand PCB

Limiter attack	Fast	←	on
	Slow	→	off
Limiter release	Fast	←	on
	Slow	→	off

Right-hand PCB

Ident marker tone	Disable	←
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MONITOR MODULE

Left-hand PCB

Split - monitor 2	DIL 3	Disable	→
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Right-hand PCB

EXT 1 - CH/MON bypass	DIL 1	MON 1	←
EXT 1/R - talk + monitor	DIL 2	Enable	←

**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 110V/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

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	OV	PIN 3	NOT CONNECTED
	PIN 2	CHARGE	PIN 4	+14V DC

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