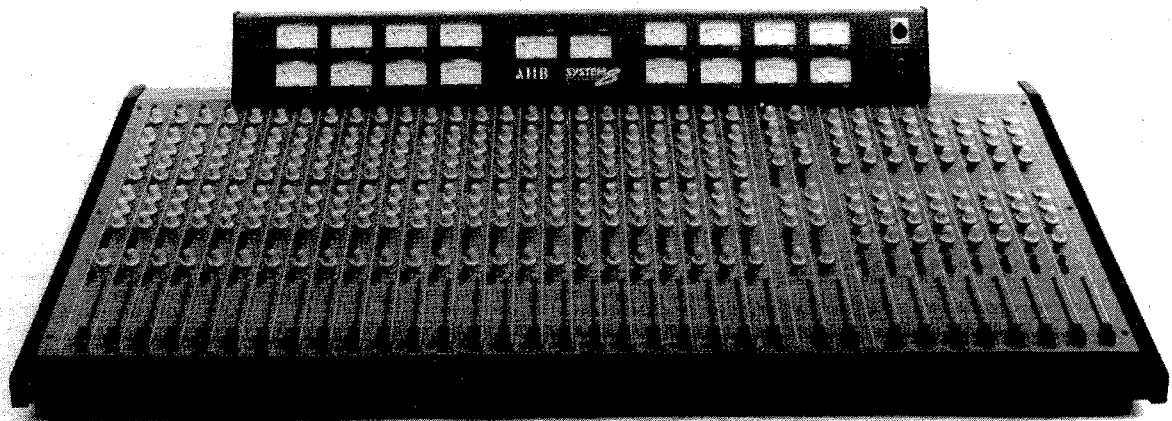


AHB

SYSTEM 8 OWNER HANDBOOK

Mk 3



SYSTEM 8 OWNER HANDBOOK Issue 3 June 1986

Allen & Heath Brenell Ltd 69 Ship Street, Brighton BN1 1AE, England

Telephone (0273) 24928 Telex 878235 MBIAHB G

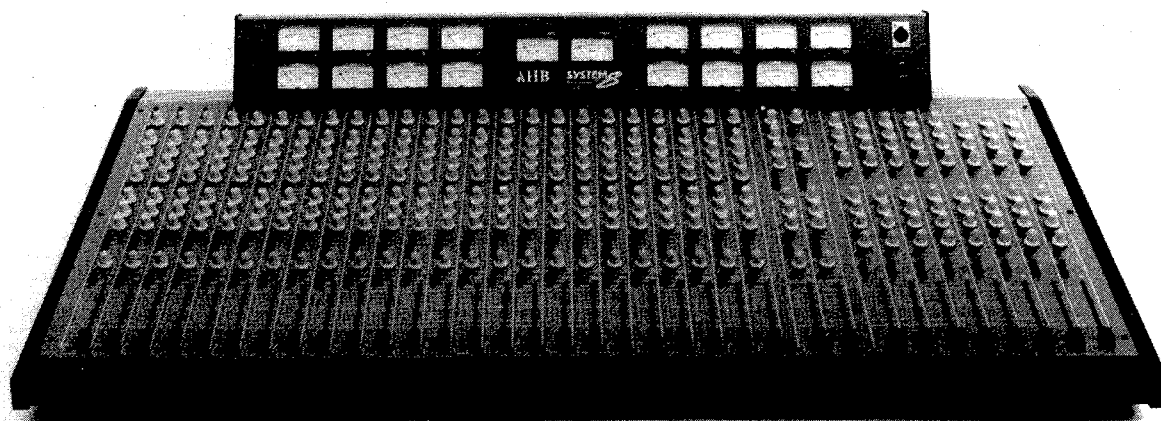
Allen & Heath Brenell (USA) Ltd Five Connair Road, Orange, CT 06477, USA

Telephone (203) 795 3594 Telex 643307 AHBUSA ONG

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SERVICE AND GUARANTEE INFORMATION

SERVICE

There are no adjustments or alignment procedures required to maintain the performance standard of SYSTEM 8 products.

To preserve the working life of the unit and its presentation, avoid the use of chemicals, abrasives and solvents. The control panel is best cleaned with a soft brush and a damp cloth. Faders, switches and potentiometers are lubricated for life; the application of electrical lubricants to these parts is not recommended.

In the event of a failure, refer the work to your AHB Sales and Service Agent. He has the information and staff to make an effective repair, and is authorised to make repairs under Guarantee. If the equipment has to be returned to the Service Agent, always include the Power Supply and as much information as possible in writing on the nature of the fault. Always include the model number and serial number with service queries to ensure that accurate information is obtained.

A Service book on SYSTEM 8 products is available on request at a small charge, and gives circuit details, spare parts descriptions and recommended procedures

GUARANTEE

SYSTEM 8 products are made in the U.K. by ALLEN & HEATH BRENELL LTD, and are guaranteed against defective parts and workmanship for a period of ONE YEAR from the date of purchase by the original owner. Other than the work specified in the OPTIONS Section of this handbook, no alterations to the original construction of the product are authorised by AHB or its agents and any such work invalidates the Guarantee.

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Section 1: SPECIFICATION

SYSTEM 8 MK 3 - High quality compact semimodular sound mixers for all applications.

MODEL 1616 - 16 input channels, 8 group outputs plus stereo mix, 16 track monitoring.

MODEL 2416 - 24 input channels, 8 group outputs plus stereo mix, 16 track monitoring.

MODEL EX8 - 8 input channels for expansion of above models.

POWER SUPPLY MODEL MPS8P applies to all models.

Audio circuits: $\pm 15\text{v}$ DC at 1 amp per rail

Phantom power: $+48\text{v}$ DC at 50 milliamps

May be internally wired to suit mains voltage between 100v and 240v AC.

Mains setting marked on MPS8P label.

GAIN (Input to Output)	<u>MIC</u>	<u>LINE</u>
minimum (includes mic PAD)	+4dB	-11dB
normal range	+24dB to +64dB	-11dB to +30dB
maximum (includes fader boost)	+74dB	+40dB

Normal operating level (outputs, line and tape inputs):

Selectable either $+4\text{dBv}$ (1.23v rms) or -8dBv (300mV rms = -10dBV)

Internal headroom: +18dB

Maximum output level: $+21\text{dBv}$ into 5kOhms or more. $+18\text{dBv}$ into 600 ohms.

Peak indicator warning at 3dB before clipping overload.

FREQUENCY RESPONSE Line input to output unity gain: 20Hz to 30kHz $\pm 1\text{dB}$ ref. 1kHz

Mic input to output +45dB gain: 30Hz to 30kHz $\pm 1\text{dB}$

EQUALISER CHARACTERISTIC: HF Shelving equaliser $\pm 16\text{dB}$ at 12kHz or 8kHz

LF Shelving equaliser $\pm 12\text{dB}$ at 120Hz or 80Hz

MID peak/dip Q=1.5 $\pm 12\text{dB}$ at 400Hz to 6kHz sweep

DISTORTION (Mic input to output +45dB gain): better than 0.05% THD 30Hz to 20kHz

NOISE 20kHz bandwidth rms noise ref. 0dBv (0.775v rms):	<u>1616</u>	<u>2416</u>
Stereo output (one line input routed unity gain):	-79dB	-79dB
" (all inputs routed and faders closed):	-81dB	-79dB
Group output (one line input routed unity gain):	-81dB	-81dB
" (all inputs routed and faders closed):	-78dB	-74dB
Auxilliary output (unity gain and inputs closed):	-77dB	-74dB

All figures better by 4dB ref. OVU = +4dBv

Microphone equivalent input noise: -125dB all models 200 ohm source.

CONSTRUCTION: Steel main panel and covers, hard stove-enamel paint finish with epoxy ink screenprinted legend. Covers removeable for service access. Semimodular electronic printed circuit assemblies. IC op-amp and discrete transistor audio system of transformerless design.

DIMENSIONS:	<u>1616</u>	<u>2416</u>	<u>EX8</u>
Width	1065mm (42")	1350mm (53")	381mm (15")
Height	245mm (9 $\frac{3}{4}$ ")	245mm (9 $\frac{3}{4}$ ")	125mm (5")
Front/back	672mm (26 $\frac{1}{2}$ ")	672mm (26 $\frac{1}{2}$ ")	672mm (26 $\frac{1}{2}$ ")
Weight	47kg (103 lb)	64kg (141 lb)	17kg (37 lb)

Section 2: CONNEXION DETAILS

GENERAL

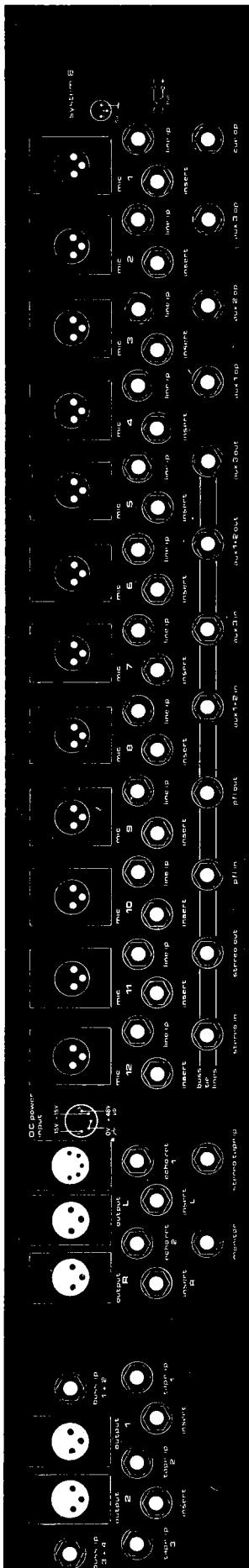
SYSTEM 8 is a thoroughly modern mixer that has been designed with the user in mind and presents no difficult connexion challenges. In general you can connect any signal source to any input and the high input impedance (typically 50 k ohms) will not load down the other equipment. The outputs are low impedance (typically 22 ohms) and can drive all commonly encountered external equipment, including 600 ohm inputs, without loading problems. Full specification is achieved with loads of a few thousand ohms (k ohms) or more. The low output impedance means that levels are maintained whether a load is connected or not, and also that long cable runs introduce the minimum HF roll off.

Particularly pay attention to matching the tape recorder input and output levels. Several popular tape systems operate 12dB below the studio standard +4dBm level, that is at -8dBm (0.30v) for 0 VU. SYSTEM 8 is engineered to work perfectly with these low level tape systems by reversal of internal selectors. You won't even need a soldering iron!

The last area worthy of attention is the microphone input which can be phantom-powered for Condenser microphones (Power supply MPS8P). You should take care to connect only 3-wire balanced cables, connectors and external equipment when the Phantom Power is on. If in doubt, look for links on XLR pins 1 and 2, and remove them. Or check with an ohm meter before plugging in. This way you will avoid the loud noises and possible damage caused by putting +48 volts where it wasn't meant to go.

For those who want the full details, they now follow.

Happy recording.



CONNEXIONS

Connector details, all versions

Power is provided by AHB power supply: type MPS8P.

Prior to use, check the marked AC voltage rating against the local AC supply. MPS8 units are connected internally for either 100v, 110v or 220v nominal AC voltage. Connexion with the mixer is with the cable and locking 5 way XLR connector provided.

AUDIO CONNEXIONS. General conventions.

CONNECTOR TYPE	BALANCED	UNBALANCED
XLR:pin 1	earth	earth
pin 2	signal -	earth
pin 3	signal +	signal +
$\frac{1}{4}$ " jack:tip	-	signal +
(mono) case	-	earth
$\frac{1}{4}$ " jack:tip	signal +	L/odd numbers
(stereo) ring	signal -	R/even numbers
case	earth	earth
$\frac{1}{4}$ " jack:tip	-	insert return (input)
(stereo) ring	-	insert send (output)
(insert) case	-	earth

0 dBv = 0.775v RMS

0 VU = +4dBv or -8dBv (0.30v)selectable

Max level = +21dBv

Fig. 1 - Back panel details

Name	Function	Connector type	Impedance in out		Nominal level for 0VU
Mic	Balanced microphone input	XLR 3 pin Female	2k Ω		-60dBv min (0.8mv) 0dBv max (0.77v)
Line	Balanced line input	$\frac{1}{4}$ " stereo jack	47k Ω		-26dBv min (39mv) +15dBv max (4.4v)
Insert	unbalanced breakpoint pre-equaliser	$\frac{1}{4}$ " stereo jack	5k Ω	22 Ω	+4dBv (1.23v)
Outputs 1-8 and L-R	unbalanced group output	XLR 3 pin male	-	22 Ω	+4dBv (1.23v) or -8dBv (0.31v)
Insert 1-8	unbalanced breakpoint pre fader	$\frac{1}{4}$ " stereo jack	5k Ω	22 Ω	+4dBv (1.23v)
Tape ip 1-8 (Model 1616 1-16)	unbalanced line input	$\frac{1}{4}$ " mono jack	50k Ω	-	+4dBv (1.23v) or -8dBv (0.31v)
Stereo Tape ip	unbalanced line input	$\frac{1}{4}$ " stereo jack	50k Ω	-	+4dBv (1.23v) -8dBv (0.31v)
Echo Return 1 & 2	unbalanced line input	$\frac{1}{4}$ " mono jack	10k Ω	-	+4dBv (1.23v)
Monitor	unbalanced stereo output	$\frac{1}{4}$ " stereo jack	-	100 Ω	+4dBv (1.23v)
Headphone	unbalanced stereo output	$\frac{1}{4}$ " stereo jack	-	100 Ω	For 8-600 Ω headphones
Aux 1 op	unbalanced line output	$\frac{1}{4}$ " mono jack	-	22 Ω	+4dBv (1.23v)
Aux 2 op	(as Aux 1 output)				
Aux 3 op	(as Aux 1 output)				
Cue op	unbalanced stereo line output jack	$\frac{1}{4}$ " stereo jack	-	22 Ω	+4dBv (1.23v)
Buss ip 1 & 2 3 & 4, etc.	unbalanced line inputs	$\frac{1}{4}$ " stereo jack	40k Ω	-	+4dBv (1.23v)

Name	Function	Connector type	Impedance in	Impedance out	Nominal level for 0VU
Stereo in	unbalanced line inputs	$\frac{1}{4}$ " stereo jack	40k Ω	-	+4dBv (1.23v)
Stereo out	unbalanced line outputs	$\frac{1}{4}$ " stereo jack	-	22 Ω	+4dBv (1.23v)
Aux 1 & 2 in	unbalanced line inputs	$\frac{1}{4}$ " stereo jack	50k Ω	-	+4dBv (1.23v)
Aux 3 in	unbalanced line input	$\frac{1}{4}$ " mono jack	50k Ω	-	+4dBv (1.23v)
Aux 1 & 2 op	unbalanced line outputs	$\frac{1}{4}$ " stereo jack	-	22 Ω	+4dBv (1.23v)
Aux 3 op	unbalanced line output	$\frac{1}{4}$ " mono jack	-	22 Ω	+4dBv (1.23v)
PFL in	unbalanced line input and control	$\frac{1}{4}$ " stereo jack	12k Ω (tip) 100k Ω (ring)	+ 4dBv (1.23v) earth to activate relay	
PFL out	unbalanced line output and control	$\frac{1}{4}$ " stereo jack	12k Ω (tip) 100k Ω (ring)	+4dBv (1.23v) earth to activate relay	

IMPORTANT - PLEASE READ CAREFULLY!**EARTHING**

There are two reasons why mixer and equipment earthing is important:

1. HEALTH - Unearthed equipment is a hazard in the event of a breakdown.
2. SOUND QUALITY - Incorrectly-earthed equipment is liable to pick up local transmitters, mains interference and power line hum, with loss of audio output quality.

In order that earthing can be made correct, many pieces of equipment are supplied not earthed. You then make the right connexions in your set up.

Make these simple checks:

- (1) Is the power cord 2-wire or 3-wire? If it is 2-wire, then an earth has to be provided via the chassis or the audio connexions to the equipment.
- (2) If the power cord is 3-wire, make this test:

Disconnect from power and all other equipment.

Measure resistance between the earth wire in the power cord and audio connector cases and pins.

Look for zero ohm resistance. Make a note that this piece of equipment can be used to pass on the power earth to other equipment via the audio connectors. The zero ohm reading means that there is a direct circuit between the power earth and the audio connector case contacts. This is also known as audio ground or audio common.

You now have two groups of equipment

- (a) Those that have 3-wire power cords and direct earth circuit to audio common, and
- (b) Those that have 2-wire power cords, or have 3-wire power cords but no direct circuit to audio common from

the earth wire.

SYSTEM 8 units are in the latter group. The power supply case is earthed for safety purposes. The power cable to the mixer does NOT include this earth. One has to be provided by the installation wiring.

Next, check on the electricity supply available to you. It must include an earth wire. DO NOT ASSUME THAT THE PERSON WHO MADE UP THE MULTIPLE POWER OUTLET BOARD ALSO PUT ALL THE EARTHS ON. Disconnect it from the mains and examine it to ensure that the earths are connected all the way back to the wall plug. In larger installations go further and have the building supply earth checked by the supply company. The more equipment you are going to power up, the greater the potential problems.

Now you can make connexions for audio and power in a methodical fashion.

EXAMPLE

- (i) Your power amplifier has a 3-wire power cord AND audio common is connected to the incoming earth. Check for zero ohms between power cord earth wire and audio connector case (jacks and phonos) or XLR pins 1.
- (ii) The power amplifier is chosen as your main earth. Its power cord earth MUST be connected all the way through to the wall plug. Check it.
- (iii) Connect the SYSTEM 8 monitor output to the power amplifier input connectors. Ensure that the audio screen is on at both ends (jack case at the mixer: phono case, XLR pin 1 at the amplifier).
- (iv) Connect the rest of your equipment (tape machines, echo units, etc) to the mixer using audio cables where the audio screen is on at both ends.
- (v) On all other equipment EXCEPT THE POWER AMP, disconnect and secure the power cord earth wire in the power plug. With 2-wire powered equipment there is nothing to be altered.

With this system the mixer is getting its earth from the power amp, and all other equipment is earthed via cables to and from the mixer. Note that when you disconnect the audio cables between a unit and the mixer, you are also disconnecting the safety earth.

You could have chosen another piece of equipment as the main earthed item. Then you make sure its earth is on, connect all other equipment with audio cables with screens connected to common at both ends. Then lift the power plug earths on the other units, but not on the main unit.

For example, you could earth SYSTEM 8 via an audio connector. Choose one that remains in place because if you should later remove it you would also remove the system earth. Preferably choose a locking connector such as the stereo output XLR connector. When making up the output connector, add an earth wire to pin 1. This should be insulated wire, medium-to-heavy duty, and the other end must be taken to the AC power plug earth terminal.

SYSTEM 8 is now your main earthed unit from which other units get their earth via the audio cable screens. Make sure that it is connected to the connector cases at both ends. On units with 3-wire power cords and audio common connected to earth, lift the earth wire in the AC power plug. 2-wire powered equipment needs no attention.

If in any doubt about the earthing of your equipment, have it checked by a competent technician. The earlier in your planning that you take studio earthing into account, the better your chances are of building up a quiet, trouble-free system.

INSTALLATION

OPERATING SYSTEMS AND AUDIO CONNEXIONS. Typical uses of SYSTEM 8 mixers are shown with connexion schematics. Details of individual connector configurations follow in list form.

SYSTEM 8 AND STUDIO MULTI-TRACK RECORDING. All recorders, 4 track, 8 track and 16 track, require connexion to the mixer in the same manner. Details for 8 track are given and where differences arise from 4 track and 16 track use, references are made.

EQUIPMENT LIST:

- SYSTEM 8 mixer (Models 128 or 168)
- Microphones and instruments
- Multi-track recorder
- Stereo recorder
- Monitor power amplifier and loudspeakers
- Echo effects unit
- Cue headphone amplifier and outlets

CONNEXION OF MICROPHONES AND INSTRUMENTS

CHANNEL MIC (microphone) inputs are low input impedance, high gain inputs for the connexion of all types of microphone. Best performance will be obtained using balanced microphones of 200 ohm to 600 ohm output impedance connected via screened 2-core cable. In this configuration, the mixer phantom power distribution may be used to power condenser microphones. Unbalanced microphones should be connected with the signal on connector pin 3 and screen on pins 1 and 2 joined together. In this configuration, phantom power must NOT be applied. The mixer input withstands the +48 voltage put out by certain types of condenser microphone external powering unit.

Mic inputs can also be used with instruments such as guitars, and a Direct Injection box is required to impedance match the instrument to the preamp and balance the circuit to reduce interference pick-up.

Instruments with medium output levels (50-500 mV), such as keyboards, may also be connected to the mic inputs if the

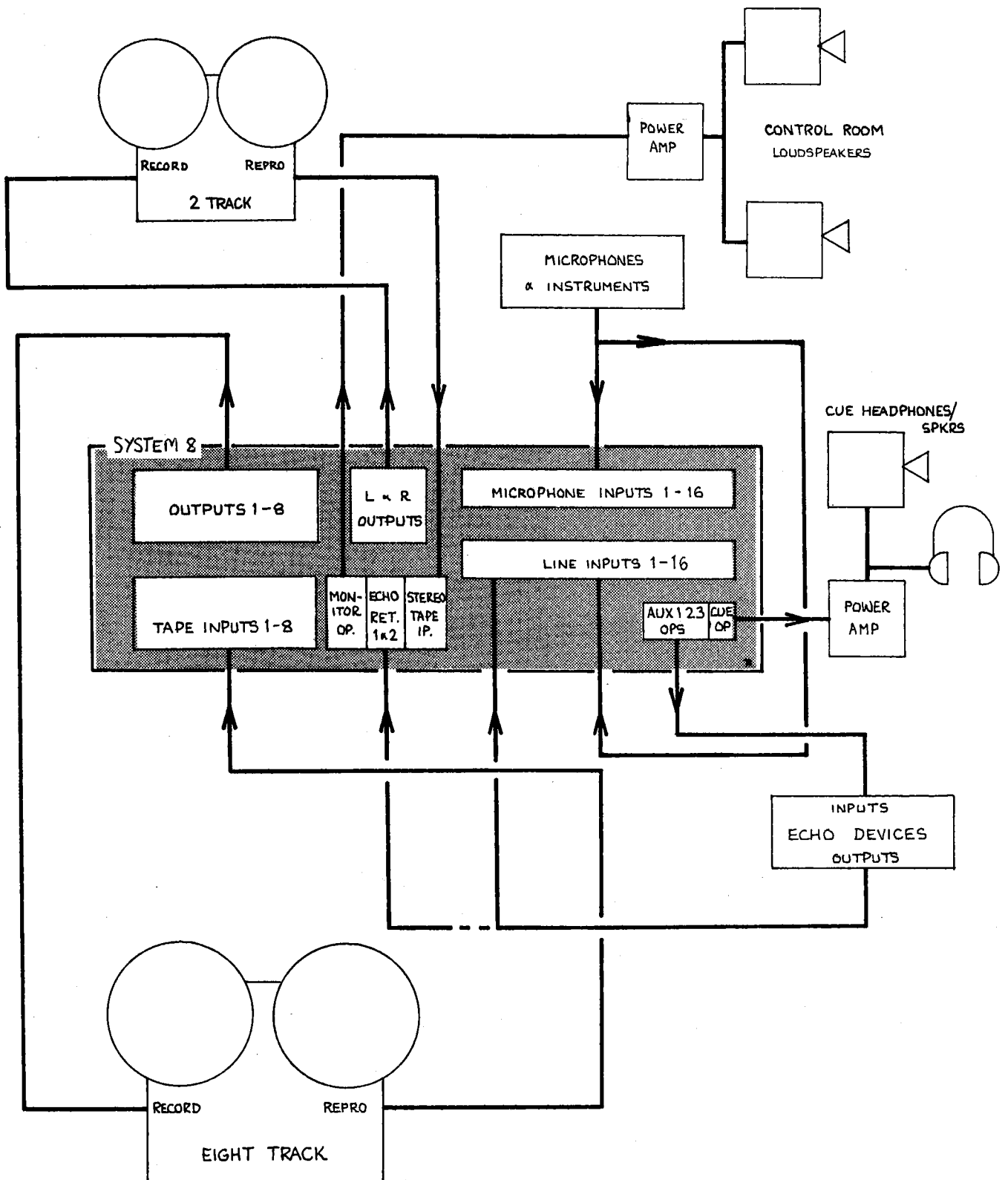


Fig.2 SYSTEM 8: EIGHT TRACK RECORDING

instrument output is low impedance. Connect the instrument signal wire to the mic connector pin 3. Connect screen to pins 1 and 2, joined together. DO NOT SELECT PHANTOM POWER when this arrangement is in use. Should the instrument output be high impedance, then loss of signal will occur and the signal-to-noise ratio will suffer. Selecting mic PAD may produce an acceptable input.

When the instrument has a choice of High or Low (level) outputs, use the high level. Operating SYSTEM 8 with many keyboards and effects, you can use mic inputs in this way to get into the mixer. You will have the instrument available on the channel in the mic mode and a tape track available on the same channel when selected to line mode.

CHANNEL LINE INPUTS are high input impedance, medium gain inputs for the connexion of high level signals. Eight line inputs will be needed to replay the tape tracks for remix. The rest are available for connexion of instruments with high output levels, such as echo returns, special effects, additional tape machines, gram and cart machines, etc.

The input is balanced. Connect balanced equipmeny to tip and ring of the jack; screen to case. Connect unbalanced signals to the tip and join ring and screen to the case.

CONNEXION OF THE MULTI-TRACK RECORDER

The AHB 8 track harness includes all the necessary connectors and links for the system described.

For eight-track work, sixteen connections are made at the tape machine - eight record inputs and eight repro outputs. The eight mixer outputs are unbalanced on XLR, pins 1 & 2 (earth) and 3 (Signal). Connect these to the record inputs of your multi-track. There are several standard signal levels for tape recorder operation. SYSTEM 8 has low impedance outputs that work with all tape machine levels. Refer to the OPTIONS LEVEL MATCHING Section for information on the correct level for your machine.

The eight multi-track outputs require connexion to the mixer for sync monitoring during overdubs and for remixing through

channels.

Connect each multi-track output to its correspondingly-numbered mixer TAPE input. The input is unbalanced.

Connect signal to tip, screen to ring and case.

Each multi-track output is internally linked to its correspondingly-numbered channel line input through the switching contacts of the LINE input socket. That is, TAPE input 1 automatically connects to LINE input 1 when it is unplugged, TAPE input 2 connects to LINE input 2, and so on up to TAPE input 16. When remixing through the channels ensure that the corresponding LINE input sockets are unplugged.

LEVEL MATCHING for the TAPE input is automatically correct and gives correct loudness and meter calibration when the mixer output circuit has been set. Refer to OPTIONS, LEVEL MATCHING Section.

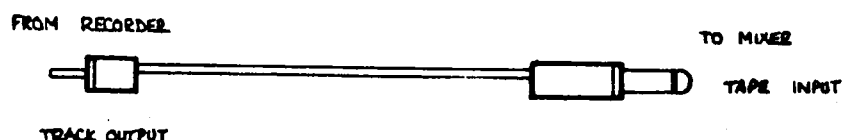


fig. 3

A versatile addition to the recording set-up is a patch point jack on every Tape input circuit:

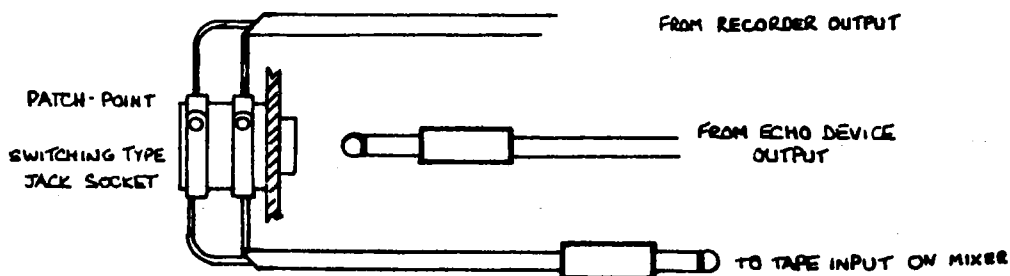


fig. 4

SIXTEEN TRACK RECORDER CONNEXIONS:

SYSTEM 8 mixer outputs 1-8 are used to provide the mix for eight tracks that can be recorded simultaneously.

Connect each mixer output to a pair of recorder inputs as follows:

MIXER	RECORDER
Output 1	Input 1 and Input 9
Output 2	Input 2 and Input 10
Output 8	Input 8 and Input 16

This requires an output cable that splits the signal two ways. The split can be made in the mixer XLR output connector body or in the tape machine input connector body.

Recorder outputs are connected to TAPE inputs 1 to 16 which automatically connects to channel LINE inputs 1 to 16 for remixing.

A second method of 16 track recording uses channel direct outputs to place single instruments on individual tracks.

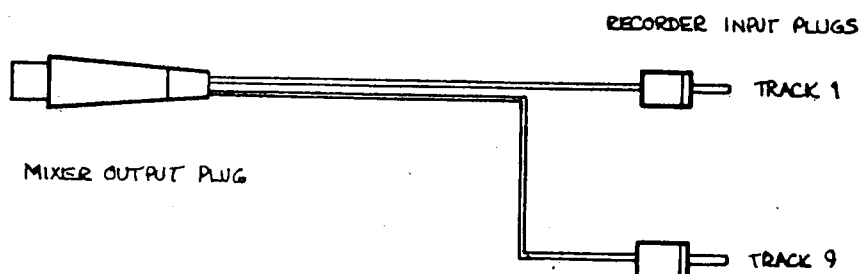


fig. 6

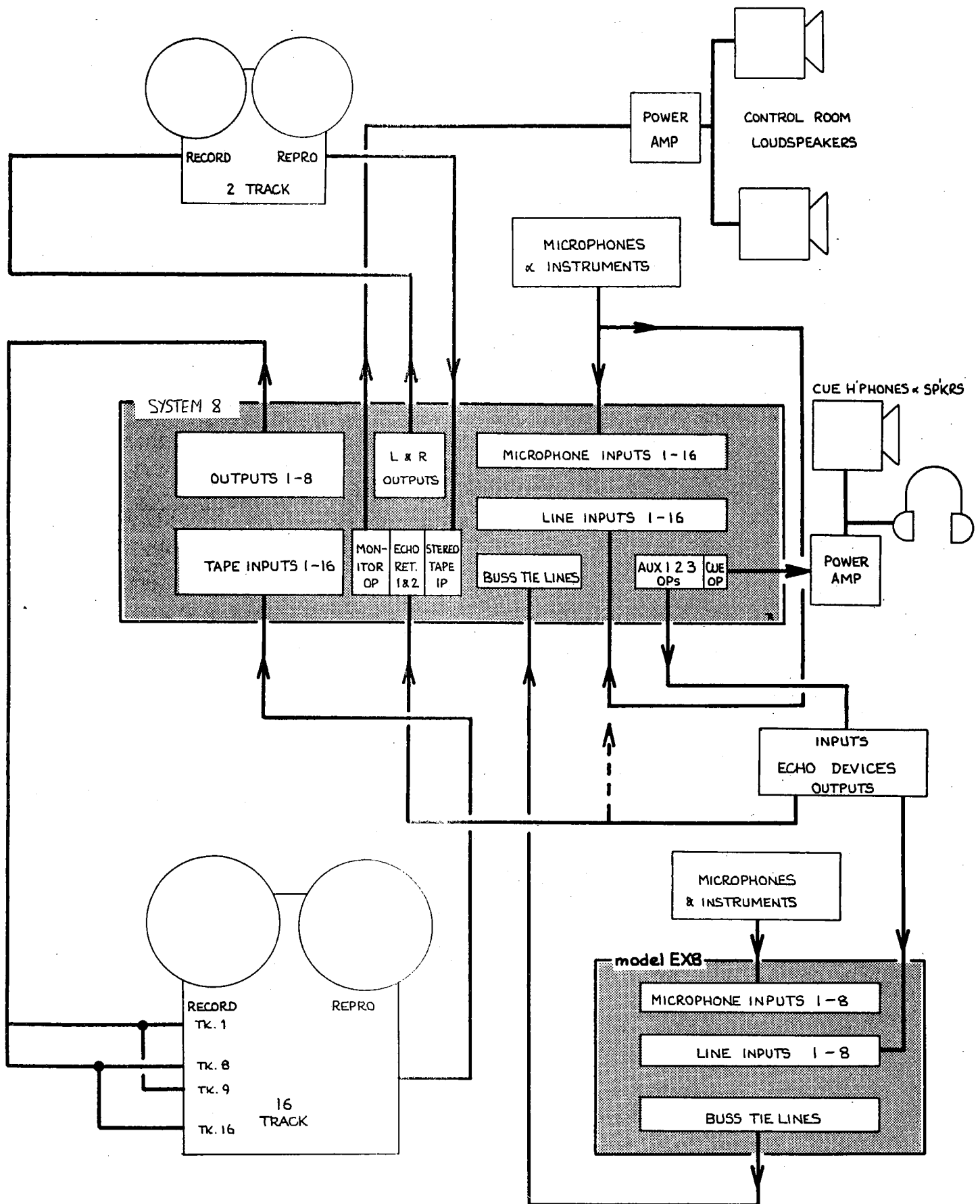


fig. 5

SYSTEM 8 16 TRACK RECORDING

CONNEXION OF STEREO RECORDER

Mixer main output L (Left) and R (Right) are the stereo output for the master recorder. The output level is adjustable by internal links for any tape machine level. For details refer to the OPTIONS Section on LEVEL MATCHING.

The outputs are unbalanced. Connect XLR pin 3 to the recorder input and connect the cable screen to pin 1.

Stereo machine outputs can be connected to the STEREO TAPE IP input jack which is an unbalanced stereo circuit. This input is independently adjustable for level matching. Connect recorder output Left to the input jack tip and recorder output Right to the input jack ring. Connect screens to jack case. Make a link connexion to a pair of jacks if you wish to have the stereo replay available on a pair of channel line inputs.

CONNEXION OF MONITOR POWER AMPLIFIER

The operator's monitor output is available on the $\frac{1}{4}$ " jack socket MONITOR. This is an unbalanced stereo circuit designed for use with stereo power amplifiers with sensitivity for rated output between 100mv and 1v approximately. Should the amplifier sensitivity fall outside this range refer to the OPTIONS Section.

Connect the tip (Left) of the monitor output jack to the power amp Left input, ring to Right and screen to case.

Mixer outputs generate switch-on thumps, which can strain loudspeakers. It is a wise precaution to switch the power amplifier off before switching the mixer on as this prevents switch-on pulses reaching the loudspeakers.

CONNEXION OF ECHO EFFECT DEVICES

Mixer outputs AUX 1, AUX 2, AUX 3 are for connexion to the inputs of echo devices. Each output is a mono unbalanced circuit. Connect the tip of the output jack to the equipment input, connect screen to jack case.

Other mixer outputs can be used to drive echo devices as required, e.g. outputs 1-8 (when not used for recording) and

channel insertion points. If you want to use a spare output for echo send, make sure the echo device can handle the output level that the mixer is set to for recording purposes. The use of insertion points for echo send is covered later.

CONNEXION OF CUE (FOLDBACK) AMPLIFIERS

The CUE OP is intended for connexion to power amplifiers for headphone and loudspeaker monitoring by performers. This is a stereo unbalanced circuit and includes TALKBACK when selected.

Connect each side of the CUE output to a power amplifier input, either one stereo or two mono units. Connect output jack tip to the left amplifier, ring to the right amplifier and screen to case.

For distribution of cue programme to a number of headphones, the power amplifier output should be divided by a suitable resistor network.

CONNEXION OF INSERTION POINTS

Each input channel, output group and the stereo output group all have circuit breakpoints for connexion of external equipment by $\frac{1}{4}$ " jack plug. The circuit is unbalanced at normal level and both output and input are on one jack. Audio signal flow through the mixer traverses these jacks when nothing is plugged into them. Insertion of a jack plug opens the switching contacts and directs mixer output to the jack ring circuit. Any signal applied to the tip contact proceeds to the following audio stage.

Connect external equipment such as compressors, limiters, noise gates, to the insertion points when you wish to have control over the signal by the external equipment. This can be done by wiring your external equipment to cables so that the input and output of the device is on one stereo jack plug. This is inserted wherever needed in the audio path by use of an insertion point jack socket.

Connect external equipment inputs to the ring of the jack, outputs to the tip of the jack.

A second use for insertion points is as an access point from which to 'borrow' the signal in the audio path. The normal flow of (say) mic input to track is not interrupted. The insertion jack only taps off the signal for use elsewhere, in an echo device for instance. To achieve this configuration connect a stereo jack plug tip and ring contacts together.

Connect the external equipment input to this join. Use this method to send an input signal to a number of different outboard units when the Auxiliary Sends are in use. It is the equivalent of the 'parallel' strip found on studio patch-bays.

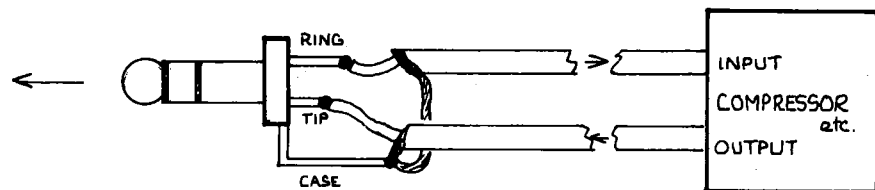


fig. 7 JACK PLUG for INSERTION POINT
"BREAK"

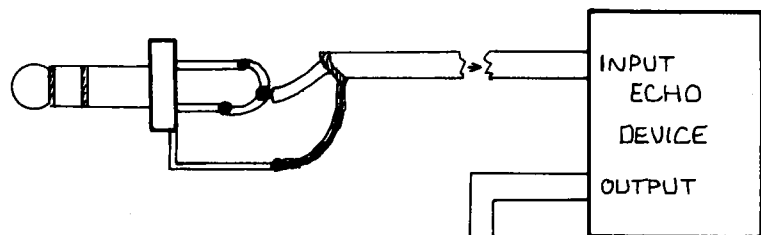


fig. 8

JACK PLUG for INSERTION POINT
"BORROW"

to ECHORETURN INPUT

This technique is safe to use since the channel insertion point is in the phase as the channel input and group output. So long as the external equipment preserves the signal phase, you mix the processed signal without cancellation effects. The output and stereo output insertion points are in the opposite phase to the channel inputs and outputs and should not be used to borrow signal for echo devices unless the echo device can also invert the signal phase (e.g. by reversing its balanced input connexions).

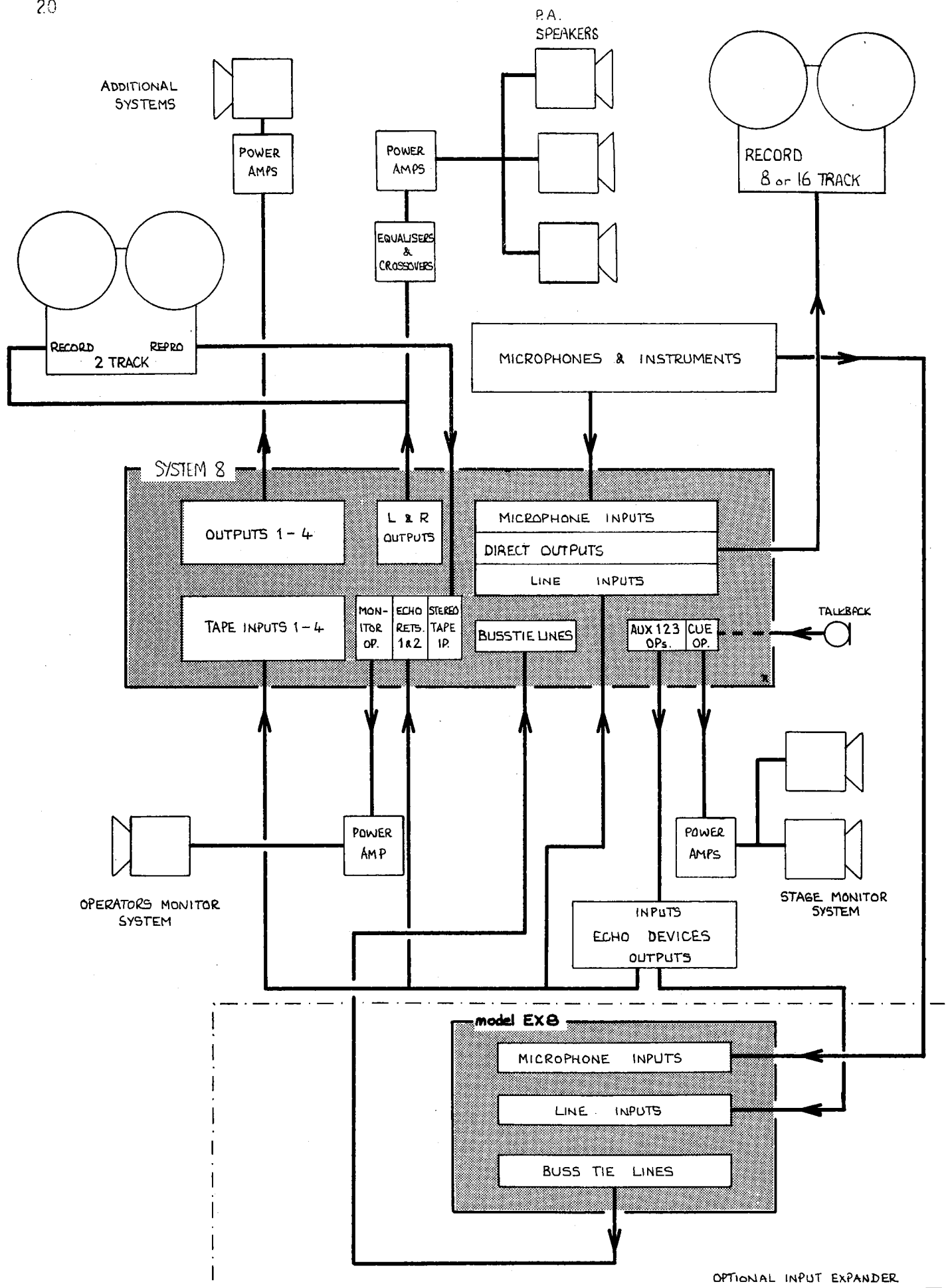


Fig. 9 SYSTEM 8 PUBLIC ADDRESS MIXING

SYSTEM 8 AND PUBLIC ADDRESS MIXING (P.A.)

Use of SYSTEM 8 for P.A. mixing will be described for connexion purposes in a system where the stereo output provides the main P.A. signal, cue is used for stage monitor and the track monitor sections used for additional echo returns to the stereo mix. Many other arrangements are possible to suit particular needs.

EQUIPMENT LIST:

SYSTEM 8 mixer
Microphones and instruments
Multicore snake
P.A. power amplifiers and speakers (main and stage monitor)
Echo effect devices

CONNEXION OF MICROPHONES AND INSTRUMENTS

CHANNEL MIC (microphone inputs) are low input impedance, high gain inputs for the connexion of all types of microphones. The input is balanced and provided with phantom power +48 volts when power supply MPS8P is used. The input withstands the 48 volt biasing present on the output of some types of external phantom power supply unit.

CHANNEL LINE inputs are high impedance, low gain inputs for the connexion of high level signals. The input is balanced. The discussion under connexions of microphones and instruments for multi-track use applies equally to P.A. use.

MULTICORE SNAKE

SYSTEM 8 models can accept connector bodies and wiring for multicore systems. Two methods are possible.

- (1) Use a multicore that terminates with XLR and jack connectors to suit the mixer connectors.
- (2) Pierce the meterpod rear cover to accept the connector body chosen and wire it to microphone and output connectors within the mixer. There is no standard for multicore connectors and the metalwork is left unpierced for fitment of the connector chosen.

Always use multicore cable with individually screened wire pairs for best quality results.

CONNEXION OF P.A. POWER AMPLIFIERS (MAIN AND STAGE MONITOR)

SYSTEM 8's stereo output provides the main stereo mix for P.A. work. It is also possible to use the individual group outputs 1-8 when using a multiple channel system with many power amplifiers feeding different areas. For P.A. work, it is recommended that the output LEVEL MATCHING is set for the higher level, 0 VU = +4dBv = 1.23v which will suit most power amplifier inputs. Refer to the OPTIONS Section for details. Outputs are unbalanced and the signal is on XLR pin 3.

When connecting mixer outputs to unbalanced inputs of power amplifiers, connect XLR pin 3 to amplifier input and screen to XLR pin 1.

When connecting to balanced inputs connect mixer XLR pin 3 to positive phase input, XLR pin 1 to negative phase input and connect screen to the power amplifier earth but not to mixer earth or pins 1 or 2.

Mixer outputs will drive any combination of power amplifier inputs whose combined impedance is greater than 600 ohms. Maximum output level is delivered into loads of 2000 ohms and more.

The stage monitor system that requires talkback facility from the mixer to the stage should be driven from the CUE output. This stereo circuit can be split to drive two separate mono systems. Refer to OPERATION section CUE SYSTEM for details.

The CUE output is unbalanced. Connexions on the stereo jack are tip = left, ring = right. Connexion details for the main outputs apply.

CONNEXION OF ECHO EFFECTS FOR P.A.

SYSTEM 8 outputs intended for echo send to external equipment are the auxiliary outputs AUX 1, AUX 2, and AUX 3. This assumes CUE is in use for stage monitor.

Auxiliary mix 1 is derived pre-fader from input channels and monitors.

Auxiliary mix 2 is selectable pre- or post-fader from input channels, and is post-fader from monitors.

Auxiliary mix 3 is derived post-fader from input channels and monitors.

Refer to OPTIONS Section if input Auxiliary Send without channel equalisation is required.

Auxiliary outputs 1 to 3 are high level, unbalanced outputs on $\frac{1}{4}$ " jacks. Connect jack plug tip to echo device input, connect screen to case.

When outputs 1-8 are not used for sub-groups or output to amplifiers, they can be routed to from channels, and connected to echo device inputs.

SYSTEM 8 inputs for echo return are of three types: Channel line inputs, echo returns one and two, and track monitor sections. These last are required, when sub-grouping, to mix the sub-group into the stereo mix. Spare monitors not used for sub-grouping can be used as echo return (without EQ) inputs to the stereo mix. The TAPE input jacks are unbalanced inputs and sensitivity is set high or low by the LEVEL MATCH option. Select whichever is appropriate for the echo device output level in use.

CONNEXION OF INSERTION POINTS

Generally the discussion under multi-track recording applies. Each input channel has an insertion point pre-equaliser. Each output has an insertion point pre-fader. The stereo output has insertion points pre-fader.

THE EX-8 EXPANDER UNIT

The add-on EX-8 unit expands your SYSTEM 8 instantly by eight channels. All EX-8 channels have *all* the facilities provided on standard SYSTEM 8 channels and identical layout. All buss outputs are mixed *before* the EX-8's output jacks. The EX-8 is really a small 8 buss mixer with no monitoring or metering.

Connexion of the EX-8 to your SYSTEM 8 is made using the tie-jack system. The EX-8 may be used in conjunction with the products of other mixer manufacturers. Outputs are unbalanced at 0 dB level (0 VU = 1.23v = 4 dBv). If in doubt about connexion details, refer to a qualified service technician.

INSTALLATION

The unit is self-powered with its own MPS8 (MPS8P) power supply unit. It is connected to a SYSTEM 8 main mixer Model 164, 128, 168 or 1616 by the BUSS TIE LINE system, using standard $\frac{1}{4}$ -inch jack connectors (3 pole type) and twin-core screened cable (*illustrated further on*).

Outputs from the EX-8 are at nominal 0 VU level (1.23v, +4 dBv) and are buffered.

The EX-8 is usually placed alongside the mixer. Alternatively, longer tie-line connectors will allow the EX-8 to be placed further away.

The EX-8 can be mechanically joined to the main unit as follows:

1. Remove bases of both units.
2. Remove left-hand side plate from the main unit and right-hand side plate from the EX-8.
3. Remove the wood trim from both these side plates.
4. Secure the side plates together, flat face to flat face, using bolts and nuts through the now vacant trim fixing holes.
5. Fit the combined side plate on the main unit. Place it face up on the workbench.

6. Place the EX-8 in position alongside, and locate the side plate in it. Replace the fixing screws to hold it.
7. Carefully turn the assembly over and refit both bases.

The strength of the assembly is sufficient for table-top and flight-case use. Floor stands should be arranged (when in use) to support both the main unit and the EX-8.

A stronger arrangement is made when a one-inch square tube is fitted across the assembly inside the armrest.

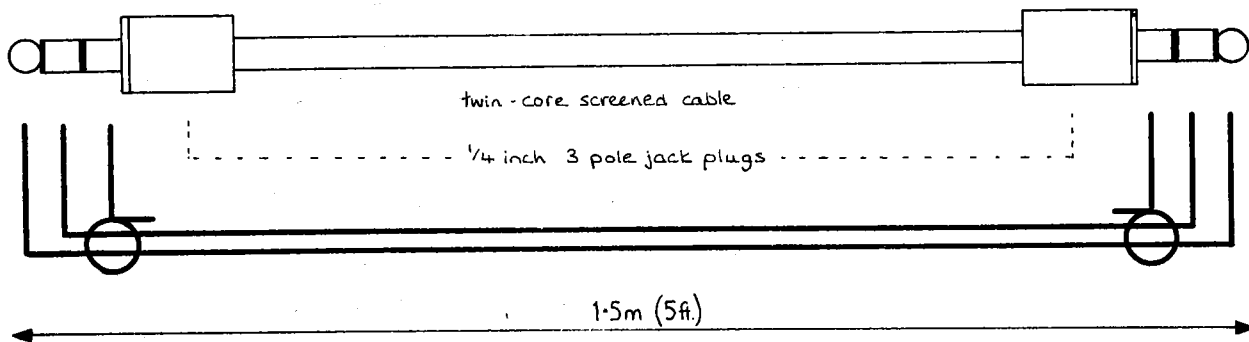
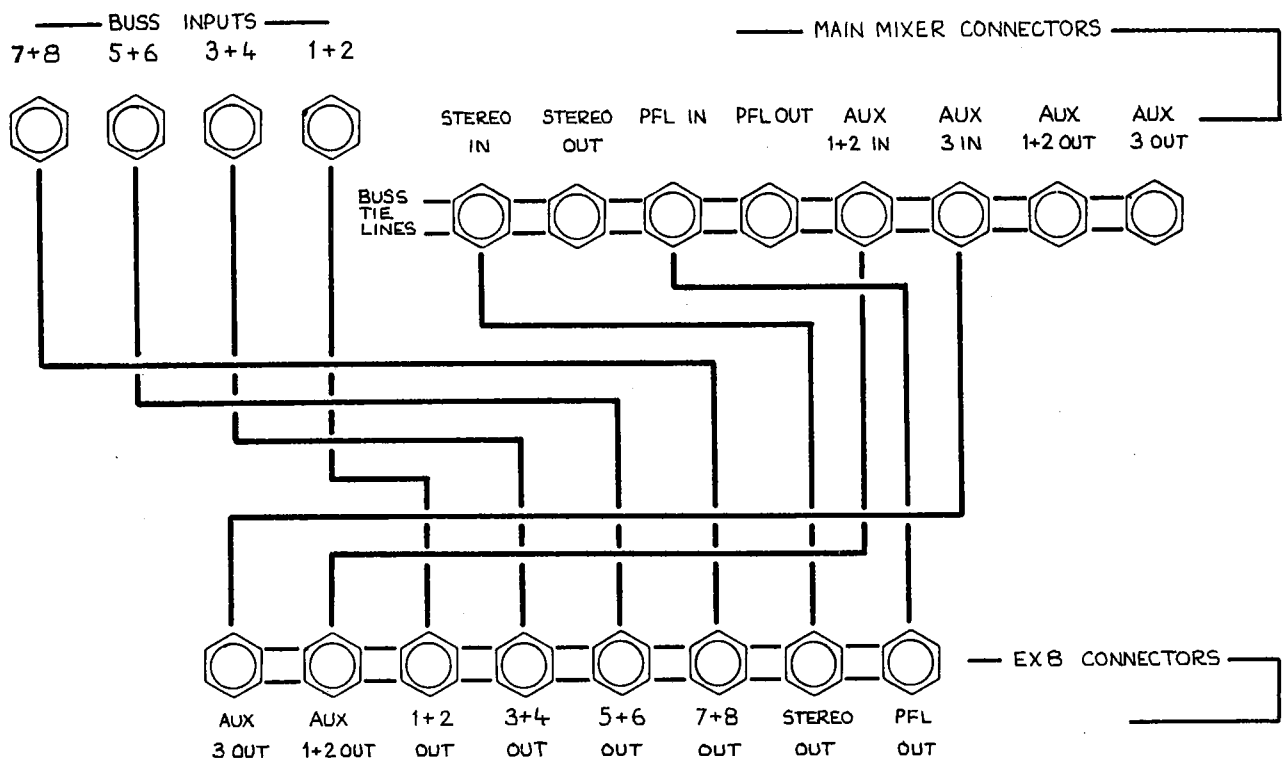


fig.10 **SYSTEM 8** Standard tie-line connector.



SYSTEM 8 EX8 connected by eight standard tie-line connectors.

TIE LINE SYSTEM

A system of input and output tie-jacks is provided to facilitate the connexion of the EX-8 expander unit, to allow SYSTEM 8 to be interconnected with other mixers, and to provide additional buss inputs and paralleled outputs. The system design allows free use of the tie-jacks as extra inputs without the need to sacrifice whole input and output sections. The CONNEXIONS part of this book gives full wiring details which use 3 pole $\frac{1}{4}$ -inch jacks throughout.

INPUT TIE-JACKS. Groups 1-8, Aux Sends 1, 2, 3, the L-R stereo mix and the PFL system, are all accessible via input tie-jacks. These enable connexion of the EX-8 expander unit, but can be used when, for any reason, you wish to patch a signal into the appropriate mix buss. *All tie-jack inputs are at 0 dB with no level-match option.* Other mixers' buss outputs may be patched into SYSTEM 8 via these input tie-jacks.

OUTPUT TIE-JACKS. These provide 0 dB outputs with no level-match option for Aux Sends 1, 2, 3, and the L-R stereo mix. Whilst effects units are connected to the main Aux Send 1, 2 and 3 outputs, the output tie-jacks of these Aux Sends can be used to connect cue systems, in addition to the cue system being fed from the mixer's main cue output. Or an effects device may be connected to give simultaneous access to two devices from one Aux Send. A tape-machine or cassette deck can be fed from the L-R stereo group output tie-jacks, so that a copy may be made in real-time with a master mix.

When the PFL system is cross-linked via the tie lines, you have access to the monitor output of the master mixer from the slave mixer and retain the one-shot ease of use.

System design preserves phase integrity throughout the tie lines. Outputs are buffered at line level (0 VU = +4 dBv = 1.23v) for use with screened lines of any length. Inputs are switched to prevent pick-up at unused tie line jacks.

An EX-8 connected to a main mixer increases your input capacity. Used in the straightforward method where corresponding circuits on the EX-8 and the main unit are joined one-to-one, you now have up to eight extra sources that can be mixed to any of the eight group outputs and the stereo output. This is the way to add echo return mixing facilities to a Model 1616, where all channels are used for tracks on remix.

This is not the end of the possibilities. It is quite acceptable for different hook-up systems to be created so long as you can follow the system and recall what you have done.

SOME RULES FOR TIE LINE CONNEXION:

Circuits identified OUT and OP are mixer and expander OUTPUTS.

Circuits identified IN and IP are mixer and expander INPUTS.

1. Only connect an output to an input, or vice versa, but NOT an output to an output or an input to an input.
2. Always use 3-pole (stereo) jacks, even if your application calls for only one of the two signals available.
3. When connecting outputs to external equipment, best performance is available when the external equipment input impedance is greater than 5000 ohms (5 kohms).
4. You won't have level control of incoming signals on the mixer panel, so adjust the signal level to a tie line input from the external equipment panel controls.

Section 3: CONTROL SUMMARY

SYSTEM 8 - INPUT SECTION CONTROLS

The Input Channel amplifies, equalises and routes incoming sound. The connectors are on the rear panel.

PAD operates before MIC input and cuts the input level by 20dB.

MIC/LINE selects the input source to be either MIC (XLR) or LINE (jack).

GAIN sets input gain for the channel and operates for MIC and LINE sources. Note the double lines at "5" and "7" indicating the unity gain ("5") and +12dB gain setting ("7").

EQ IN/OUT bypasses the equaliser section when pressed.

HF switch 12kHz/8kHz and HF + control the high frequency equaliser characteristics. Press the 12kHz/8kHz switch for higher frequency operation.

MID FREQ pot sets the mid-equaliser centre frequency in the range indicated.

MID + sets the cut or boost required.

LF switch 120Hz/80Hz and LF + control the low frequency equaliser characteristics. Press 120Hz/80Hz for higher frequency operation.

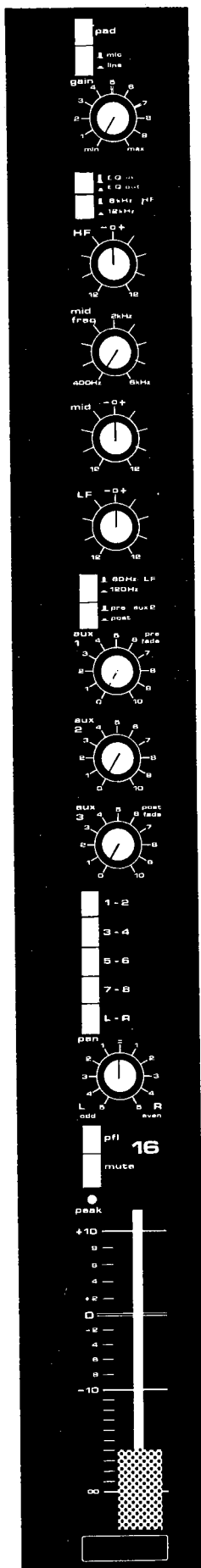
AUX 1 pot mixes input programme into auxiliary mix 1. Source is pre-fader, so the level doesn't vary with fader movement.

AUX 2 pot mixes input programme into auxiliary mix 2. Source can be pre-fader or post fader by selecting pushbutton PRE/POST.

AUX 3 pot mixes input programme into auxiliary mix 3. Source is post fader and varies with fader movement.

The auxiliary mixes are provided for cue mixing and echo mixing.

ROUTING 1-2, 3-4, 5-6, 7-8 and L-R pushbuttons switch the input programme to one or more mixer outputs.



PAN works with ROUTING to select ODD or EVEN mixer outputs. In the centre, equal signals go to ODD and EVEN numbered outputs. Also provides stereo mixing sound position.

PFL switches input programme to the mixer monitor output and main Left and Right meters for level and quality checking. The source is pre-fader and allows checks to be made whilst the fader is closed. Release PFL after use to regain normal monitor operation. PFLs on several inputs may be simultaneously selected.

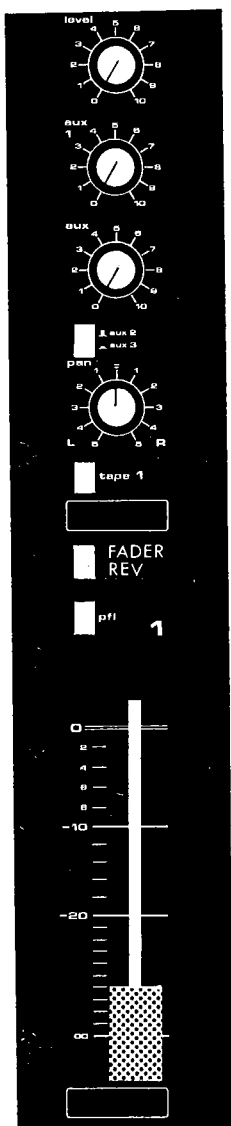
MUTE switches off the outputs of the input section that come after the fader, that is ROUTING, AUX 2 (POST) and AUX 3.

PEAK signifies high level signal levels that are approaching overload. Source is pre-fader and responds to equaliser adjustments.

FADER has a normal position at the double lines, and can move up or down to boost or cut level.

SCRIBBLE PATCH for identifying the input programme material.

INSERTION POINT (Rear Panel) provides access to the input programme after pre-amplification and before equalisation for use with signal processors.



SYSTEM 8 - OUTPUT AND MONITOR SECTION CONTROLS

The Monitor section mixes sound into the stereo output.

LEVEL pot controls the volume of the programme going to the stereo output. The maximum ("10") position provides a straight-through path with no change in level between the source and the stereo output.

AUX 1 pot mixes monitor programme with auxiliary mix 1. The source is before the LEVEL pot and doesn't vary with changes in level.

AUX pot mixes monitor programme into auxiliary mixes 2 or 3.

AUX 2/AUX 3 pushbutton decides which mix the AUX pot feeds. The source for the AUX pot is after the LEVEL pot, so the mix level to AUX 2 or AUX 3 varies with the level settings. *AUX 1 and AUX are provided for cue and echo mixing.*

PAN sets the programme position in the stereo mix. Centre position sends equal levels to the Left and Right halves of the stereo output.

TAPE pushbutton selects the programme source for the monitor section. Press to monitor the incoming signal on the rear-panel TAPE IP connector of the same number as the panel section. Release to monitor the mixer output of the same number.

SCRIBBLE PATCH provides space to identify the incoming TAPE IP programme.

MODEL 1616 ONLY has eight more MONITOR sections, so that tracks 9 to 16 of a 16-Track tape recorder can be connected and monitored.

METER with the same number as the panel section shows the signal level of the incoming programme source as selected by the TAPE switch position.

The Output Section combines sounds from Input sections for connexion to a power amplifier (for P.A. use) or Multitrack Recorder input.

FADER REVERSE pushbutton exchanges the OUTPUT FADER with the LEVEL pot to allow the signal from the multi-track recorder to be controlled by the fader during sync monitoring or overdubbing.

PFL switches output programme to the mixer monitor output and Left and Right meters for level and quality checking. The source is before the output fader, so that checks can be made with the fader closed.

OUTPUT FADER has a normal position at the double lines and fades the combined programme. Mixer output connectors are on the rear panel.-

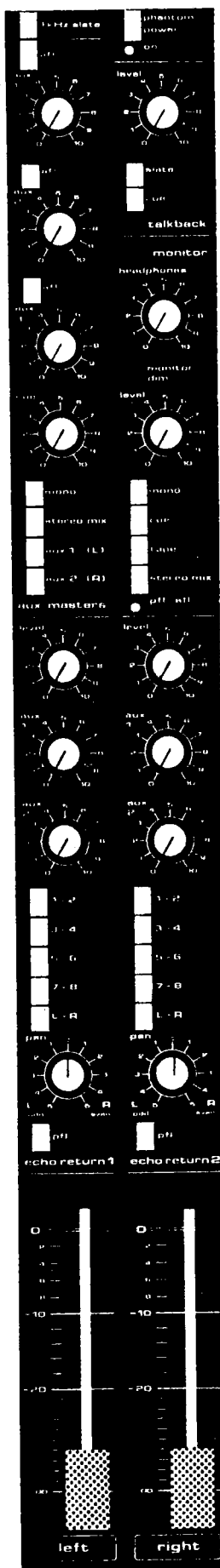
SCRIBBLE PATCH provides space to identify the output programme material.

INSERTION POINT provides access to the output programme after combining and before the fader for signal processing. PFL shows the incoming signal after processing when the insertion point is in use.

BUSS IP 1 & 2, etc., rear panel connector provides for direct injection of a signal to the combining section of an output. Each connector carries two circuits and is for use with the EX8 input expander or other equipment.

The mixer output level to tape or amplifier is shown on the corresponding meter when the MONITOR TAPE pushbutton is 'out'. See also LEVEL MATCHING in the OPTIONS Section.

SYSTEM 8 - MASTER SECTION CONTROLS



In the centre section of the mixer are the five sub-sections for control of the mixer master outputs, other than the multi-track outputs 1-8. These are **STEREO OUTPUT**, **AUXILIARY OUTPUTS 1, 2 and 3** and **CUE OUTPUT**, **MONITOR SECTION**, **TALKBACK**, and **ECHO RETURN**.

LEFT and RIGHT FADERS are the master controls for the level of the outgoing stereo mix to tape recorder or P.A. amplifier. Normal position is at the '0' double line. The output level can be set to high or low standards to match tape machine operation level. See **OPTIONS** Section. **INSERTION** connectors (rear panel) provide access for signal processing devices.

AUX MASTERS section. **AUX 1** is the master level control for the Aux 1 mix. Normal position is '8'. The **AFL** pushbutton selects the Aux 1 output to the monitor section for level and quality checking. **AUX 2** and **AUX 3** controls work for Aux 2 and Aux 3 mixes in the same way. Use for echo and cue mixing.

CUE is the master level control for the stereo cue output. The source for the cue programme is selected by the three pushbuttons **AUX 1 (L)**, **AUX 2 (R)** and **STEREO MIX**. The **MONO** pushbutton combines the selected source programme to be the same on the Left and Right cue outputs. See also **OPERATION** text for uses of CUE.

MONITOR section **LEVEL** control adjusts the volume to the **MONITOR** output for loudspeaker amplifiers. The **HEADPHONE** control independently adjusts headphone listening level from the internal headphone amplifier. Four pushbuttons select the monitor programme source. The selected source is displayed on the Left and Right meters and goes to the headphone and monitor outputs. When **PFL** or **AFL** is selected, the LED lights and the **PFL** programme (in mono) goes to meters, headphones and monitor output. Other mixer outputs are not affected. Normal setting of the four pushbuttons is the **STEREO MIX** selection. **TAPE** selects the stereo tape input (rear panel). **CUE** selects the stereo cue output. **MONO**

combines the Left and Right monitor programme, the result going to meters, headphones and monitor outputs.

MONITOR DIM reduces monitor output by 20dB, for quick level change without losing the LEVEL setting. Talkback automatically engages MONITOR DIM.

TALKBACK section LEVEL adjusts voice level to the selected Talkback destination. SLATE selects Talkback to mixer outputs 1-8

CUE selects Talkback to both sides of the stereo cue output and automatically dims the main cue programme for improved intelligibility.

ECHO RETURNS 1 and 2 are line level inputs for echo return. PFL selects the incoming programme to the monitor section for level and quality checking. PAN and ROUTING 1-8, L-R select the destination of the echo return programme. LEVEL adjusts the volume of echo return to the selected destination. AUX 1 and AUX 2 adjust the amount of echo added to the auxiliary mixes 1 and 2, and are not affected by other echo return settings.

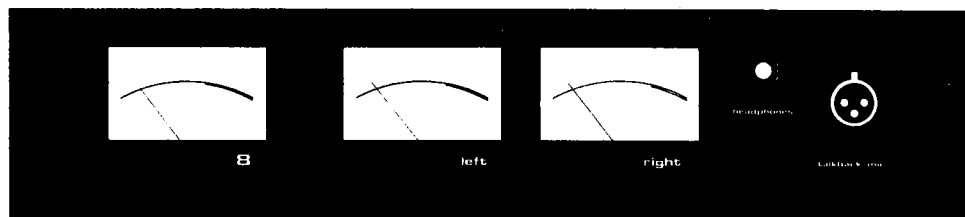
1kHz SLATE pushbutton controls the reference tone oscillator. All eight (on Model 164, four) main outputs receive reference level tone when the pushbutton is pressed. The 1000Hz tone at 0 VU is of high purity.

PHANTOM POWER pushbutton directs the external +48volt supply from Power Supply type MPS8P to the input Mic XLR connectors pins 2 and 3. See also OPTIONS section.

THE BUSS TIE LINE connector system (rear panel) provides extra jack connexions in addition to the normal input and output requirements. These jacks allow programme to be added to the mixer Aux 1 and 2, Aux 3 inputs, STEREO input and PFL input without using up input channels. This is the method of connecting in the circuits of an EX8 input expander unit. BUSS IP 1 & 2, 3 & 4, etc jacks in the output section accept programme for direct injection to the output mixes for the same purpose.

BUSS TIE LINE output jacks (Aux 1 & 2 op, Aux 3 op, STEREO

op and PFL op) provide outputs for connexion with the TIE LINE inputs of another SYSTEM 8 mixer to join the two together. When not used for this purpose, they provide additional plugging points for recording, amplifying and processing the programmes.

SYSTEM 8 - METERPOD SECTIONMETERS

1-16

These show the programme level of the mixer output or tape input as determined by the monitor section TAPE pushbuttons above each output fader. Press to read the incoming tape level, release to read outgoing output level. Meters read the internal level and are calibrated in VU, where 0VU is the maximum recommended programme level without overload distortion. The level match arrangement on outputs and tape inputs does not affect meter readings, so that 0VU on the mixer set for low level operation and connected to a low level tape machine input will produce 0VU on the tape meter. When set for high level operation, the meter reading and the signal levels at outputs and tape inputs agree at 0VU = 1.23v RMS = +4dBm.

METERS LEFT and RIGHT (All Models)

These show the programme level of the STEREO output or whichever monitor source has been selected in the master monitor section. PFL/AFL selection is also shown on the Left and Right meters in mono. Level matching arrangements for the stereo output and stereo tape input apply as for the 1-8 outputs above.

HEADPHONES jack presents the output of the internal headphone amplifier for use by the operator to check on headphones. The programme source is the selected master monitor source, either the STEREO MIX, CUE or TAPE.

TALKBACK MIC XLR socket accepts a microphone or microphone

gooseneck assembly for the Talkback system. Use a Dynamic type microphone as phantom power is not available and the unbalanced input will not accept phantom-powered microphones with bias voltage on the output connector.

Section 4: OPERATION

INTRODUCTION

Whether you're a newcomer to the world of multi-track recording or a seasoned veteran, you'll find your SYSTEM 8 logical and simple to operate from the word go. Feedback from studios and leading engineers has helped us to design into SYSTEM 8 all the features you need to make top-quality multi-track recordings and master mixes. These same features also make SYSTEM 8 a remarkably versatile stage mixer.

As you become familiar with SYSTEM 8, you'll discover how it's flexibility and your own creativity can make a great team. Working with SYSTEM 8 you'll discover many interesting possibilities and new ways of working, and we're sure you'll have a lot of fun, too!

So that you can derive maximum benefit from your SYSTEM 8's capabilities, we urge you to study the following pages carefully, getting to know the layout and function of its controls before putting it to work. You'll find that we have tried to make the operating instructions as clear yet as comprehensive as possible. Starting with the assumption that you have installed your SYSTEM 8 to work in conjunction with an eight-track tape machine, we shall outline the basic principles involved in the multi-tracking process and then go on to describe in detail all the controls and their respective functions. The mixer system diagram lets you follow the signal through from input to the various outputs.

MULTI-TRACKING OUTLINED

There are three basic stages in multi-track recording:

- (i) THE RHYTHM TRACK. A rhythm section typically consisting of drums, bass guitar, rhythm guitar, and perhaps a keyboard instrument is recorded onto several tracks of a multi-track tape machine.
- (ii) THE OVERDUBBING. Vocal and instrumental 'overdubs' are added, being recorded onto tape tracks which were not used when the rhythm section was recorded.

(iii) THE REMIX. The multi-track master tape is mixed down to the final stereo master.

MULTI-TRACKING WITH SYSTEM 8

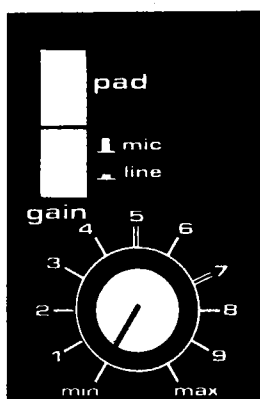
When you commence a rhythm track session with your SYSTEM 8, you will be passing the signals from your microphones through the INPUT CHANNELS, and routeing those signals to your multi-track machine via the OUTPUT GROUPS. You may want to modify or improve the sound of an instrument by adding some equalisation (EQ), and this you can do as the signal from its microphone passes through the EQUALISER SECTION of the relevant INPUT CHANNEL.

The MONITORING SECTION lets you listen on your control-room loudspeakers (or headphones) to a stereo 'monitor mix' of your recording, and you can send a mono or stereo 'cue mix' to the musicians' headphones via the AUXILIARY SENDS and CUE SYSTEM. 'Monitor echo' (echo you *don't* want recorded onto tape) can be added to sweeten your monitor mix and the musicians' cue mix (also known as the foldback mix) via the AUXILIARY SENDS and ECHO RETURNS.

THE INPUT CHANNEL

The input channel accepts a balanced or unbalanced input signal from a microphone or D.I. box and a line-level source, such as the output of a tape-track, synthesiser, or external signal-processing device such as an ADT unit or phaser. For eight-track operation, tape machine outputs 1-8 are internally wired to the line-inputs of channels 1-8.

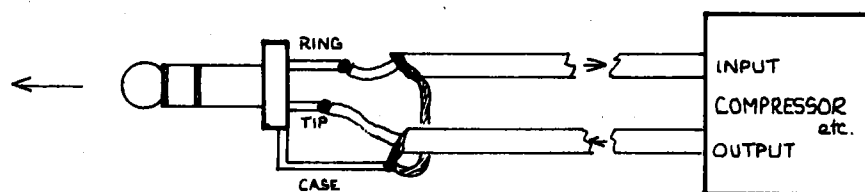
SYSTEM 8 is pre-wired for the +48v phantom powering of condenser microphones. The MPS 8 power supply with 'P' option provides the necessary voltage. The master PHANTOM POWER ON pushbutton switch and LED indicator are located at the top of the mixer above the Monitor Master and Talkback controls. When using phantom power it is important to avoid connecting an unbalanced source to a microphone input without first having disconnected the phantom power internal wire link on that channel (see OPTIONS section), or switching phantom power off.



Input source selection is made by the pushbutton MIC/LINE switch located at the top of the channel, below which is the Input GAIN pot which amplifies incoming microphone signals when MIC is selected, and amplifies or attenuates incoming LINE signals as desired. Control range for MIC input is +24dB to +64dB and for LINE input -11dB to +30dB

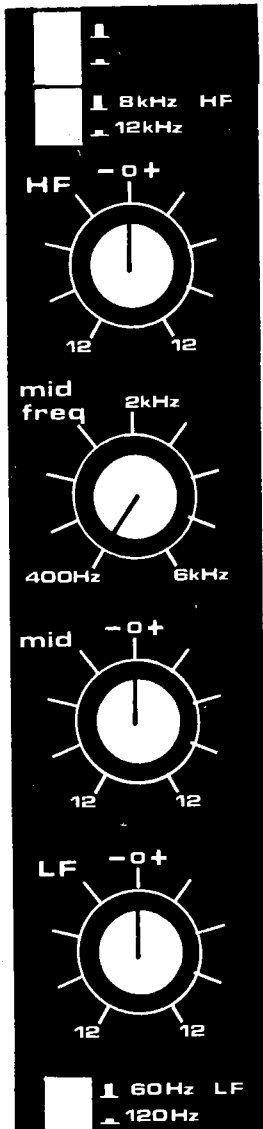
Depressing the pushbutton PAD switch prevents input overload from high-level microphone sources (e.g. bass drum, etc.) by attenuating the incoming signal by 20dB prior to the GAIN pot. The PAD switch is only operative in MIC mode. Unity gain (i.e. input level = output level) position of the GAIN control in LINE mode is at 5 on the calibrated scale, and when the line input is fed by a -8dBv source (e.g. output of a TEAC or FOSTEX tape machine) the GAIN pot should be set up to 7 on the scale to provide 12dB of gain to bring the signal to the mixer's standard operating level.

A CHANNEL INSERTION POINT is provided by a stereo jack socket (tip = return) on the rear panel, enabling you to insert an external signal processing device such as a limiter, compressor, noise-gate or graphic equaliser into the channel signal path after the GAIN pot, and before the equaliser section.



JACK PLUG for INSERTION POINT
"BREAK"

The 3-band EQUALISER Section (the 'channel EQ'), allows you to alter the tone and harmonic balance of the signal in the channel. You can also filter out unwanted high-frequency noise (hiss) or low-frequency noise (rumble and air-conditioning), and compensate for acoustic deficiencies or problems with your recording environment. The high and low frequency equalisers have a choice of two shelving characteristics, and the mid-range peak/dip equaliser is fully sweepable over the mid-range frequencies.



EQ IN/OUT. This control, when pressed, bypasses the complete equaliser section and is useful in allowing comparison between the equalised and non-equalised signal. Only when it is released do the equaliser controls have any effect.

HF (High-frequency) EQUALISER. This control, marked HF, boosts or cuts by up to 12dB at 8kHz, having progressively less effect at frequencies below 8kHz. At its centre-zero position it has a flat response, and therefore no effect on high-frequency signal content. Depressing the 8kHz/12kHz pushbutton switch immediately above the HF pot changes the control frequency from 8kHz to 12kHz.

MID-RANGE EQUALISER. This has two controls. With the upper pot marked MID FREQ, you can vary the equaliser centre frequency between 400Hz and 6kHz, and the lower pot marked MID boosts or cuts at that frequency by up to 12dB. The response of the equaliser tails off progressively each side of the selected control frequency. With the MID pot at its centre-zero position response is flat and mid-range frequencies are unaffected.

LF (Low-frequency) EQUALISER. This control, marked LF, boosts or cuts the channel signal by up to 12dB at 60Hz, having progressively less effect at frequencies above 60Hz. At its centre-zero position it has a flat response and therefore no effect on low-frequency signal content. Depressing the 60/120Hz pushbutton switch changes the control frequency to 120Hz.

Setting the equaliser controls to centre-zero produces a flat response, and the signal in the channel will be identical at the equaliser input and output.

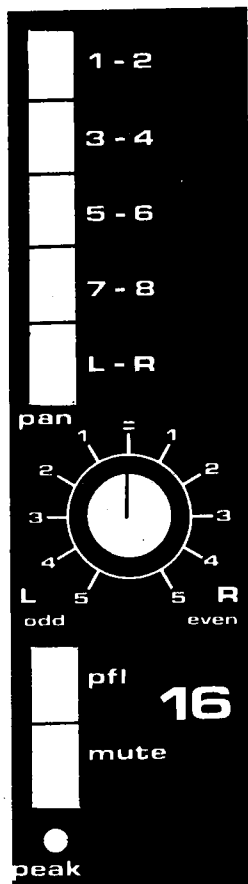
The CHANNEL FADER controls the channel output level. Fader calibration is nominally in dB, and '0' indicates the unity-gain position (fader input level = fader output level). At the top of the fader's travel (marked +10) the fader output level is increased by 10dB compared to its input level.

Immediately above the fader is the PEAK INDICATOR LED. This illuminates when the signal level in the channel reaches a point 3dB below the clipping threshold, indicating that too much input gain is being applied. This can happen when you are using equalisation to boost, rather than cut, frequencies present in the signal. (Remember when using equalisation that boosting results in a higher level at the equaliser output than at the input, so reduce input gain to compensate. The reverse also being true, you may need to add input gain when cutting at the equaliser stage!). It is good practice not to work too near the clipping point of a circuit, so that unexpected peaks do not cause the signal to distort. The distance between normal operating level and clipping, measured in dB, is referred to as *headroom*.

CHANNEL ROUTING AND PAN POT. After the fader, the channel can be routed directly to the L-R stereo mix by depressing the pushbutton marked L-R, and panned to any desired position in the stereo spread by turning the PAN pot. The channel is normally routed to L-R for mixdown, Direct-to-Stereo recording, and for stage work.

In the same fashion, the channel signal can be routed to (and panned across) the OUTPUT GROUPS, which you will normally have connected to the correspondingly-numbered tracks of your multi-track machine. The group routing pushbuttons are marked 1-2, 3-4, 5-6 and 7-8. To route to an odd-numbered group, depress the appropriate pushbutton and turn the PAN pot fully to the left (fully anti-clockwise). For an even-numbered group, turn the PAN pot fully to the right (fully clockwise). You can pan between any odd- and even-numbered pair of groups, and several pushbuttons may be depressed to select more than one pair of groups.

SYSTEM 8 has eight outputs for Tape and sixteen track monitor sections. Output 1 goes to Track 1 and Track 9. Output 2 goes to Track 2 and Track 10, etc. To route to Track 1, select 1-2 and pan left. To route to Track 4, select 3-4 and pan right. To route to Track 9, select 1-2



and pan left. On the tape machine select 9 in record, leaving Track 1 in sync. On the mixer select Tape 1 to hear Track 1 sync playback. When recording of Track 9 is completed, select Tape 9 to hear Track 9 sync playback.

CHANNEL PFL (Pre-Fade Listen). Depressing the PFL pushbutton located below the PAN pot enables you to 'solo' a channel signal on the monitor loudspeakers, whilst its level is displayed (in mono) on the L-R meters. The channel signal is 'sampled' by the PFL system at a point in the channel circuit *after* the equaliser and *before* the fader. Releasing the PFL button restores normal monitoring and metering. Using PFL has no effect on any signals going to tape at the time, and will not disturb either your monitor mix or the performers' cue mix. All PFL signals appear in mono on the monitoring, irrespective of the channel PAN pot position.

The PFL system is purely a monitor system, and is totally independent of mixer panning and routeing settings.

PFL is a really useful facility, allowing you to make quick checks and changes. For example, you may be hearing distortion on drums: By quickly depressing the input channel PFL switches you can pinpoint the offending channel. A glance at the PFL level on the L-R meters will tell you if you are overloading the channel by using too much microphone gain, or whether the problem is likely to be a dodgy microphone or cable. You may also, for example, wish to change the equalisation setting on a single vocal or tom-tom microphone being mixed into a group. PFL lets you make an instant quality check and re-equalise if necessary.

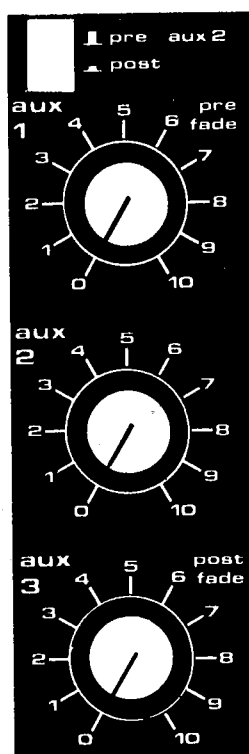
When any PFL button is depressed, the PFL/AFL INDICATOR in the centre of the mixer below the STEREO MIX button will illuminate.

CHANNEL MUTE SWITCH. This switch is located immediately above the fader. Pressing it 'kills' the output of that channel, and cuts off any POST-FADE Auxiliary Sends in use from that channel. The MUTE switch is located in the channel circuit *after* the fader and *before* the PAN pot and routeing buttons, so PRE-FADE Auxiliary Sends are unaffected

by its operation. Therefore the performers' cue mix will not change when you mute a channel feeding their cue mix via a PRE-FADE Auxiliary Send.

Remember that muted channels will not go either to tape or to the L-R stereo mix!

When you're re-mixing a song with a quiet intro section, you'll often find keeping channels muted until you actually need them turned on can help keep your mix cleaner and clearer by avoiding hiss from empty tracks.



CHANNEL AUXILIARY SENDS. Each channel has three Auxiliary Send (AUX) pots, feeding Aux Send masters 1, 2 and 3. The Aux Send outputs are for connexion to external effects and the Cue system.

AUX 1 is a fixed PRE-FADE feed to the Aux Send 1 mix. Its operation is unaffected by the position of the channel fader or the MUTE switch.

AUX 2 is a switchable Pre/Post-Fade feed to the Aux Send 2 mix. Depressing the pushbutton AUX 2 PRE/POST switch (located beneath the 60/120 Hz switch) changes AUX 2 from Pre-Fade (i.e. its output is unaffected by the channel fader or mute switch) to Post-Fade (i.e. its output will change proportionately to channel fader movements, and will be cut off when the channel MUTE switch is operated).

AUX 3 is a fixed POST-FADE feed to the Aux Send 3 mix. Its output will change proportionately to channel fader movements, and will be cut off when the channel MUTE switch is operated.

AUX SENDS 2 AND 3 PRE-EQ OPTION. The pre-fade feeds on AUX 1 and 2 Sends are taken from a point in the channel circuit immediately after the equaliser and before the fader. This enables performers to hear their voices and instruments in the cue mix enhanced by equalisation. Returns from effects devices retain the same equalisation characteristics as their source signal, which is desirable during mixdown.

Certain P.A. applications require pre-equaliser pre-fade sends, so SYSTEM 8 offers you this option on Aux Sends 1 and

2 from the channels. Simply change over a wire link as described in the OPTIONS Section of this Handbook.

THE OUTPUT GROUPS

If you are working in an 8-Track situation, it's convenient to think of an output as representing the input to the correspondingly-numbered tape track, as you will have connected these outputs to your tape machine inputs when you installed your mixer. Therefore, to assign an input channel to, say, Track 7 of the machine, route that channel to group 7 remembering, of course, to have the group fader up!

When you need to assign several channels to one tape track (for instance when mixing several backing voices together), select the appropriate group and mix them into the desired blend using the channel faders. You can now use the group fader to set the optimum recording level onto tape.

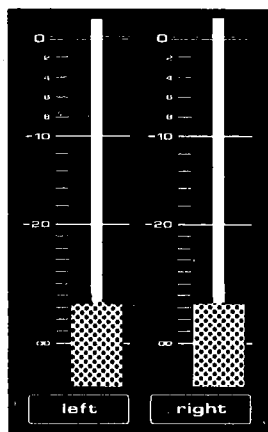
The OUTPUT FADER gives you control over the output level of the group, and it is calibrated '0' at the top of its travel (unlike the channel fader), indicating unity gain at that position. When mixing several channels into a group, you may find it useful to position the group fader at -6dB before getting a balance on the channel faders. In that position you can raise the level of your blend to tape as necessary, as well as being able to reduce it. Group output level is displayed on the corresponding meter.

A GROUP INSERT POINT is provided by a jack socket on the rear panel (tip = return), enabling you to insert an external signal-processing device such as a limiter, compressor, noise-gate, etc., into the output group signal path before the output fader.

GROUP PFL (Pre-Fade Listen). The Group PFL system functions in the same manner as the channel PFL system. Depressing a

Group PFL button sends the output pre-fader signal to the monitor loudspeakers and L-R meters without disturbing recording or auxiliary systems.

SYSTEM 8 works with all multi-track tape machine operating levels and group output levels should be set to match tape machine line levels as described in the OPTIONS Section.



THE STEREO OUTPUT

The main L-R stereo mix is used for multi-track monitoring purposes, for recording direct to stereo, and as the main stereo mix during mixdown. Output levels are controlled by the L-R faders, calibrated '0' (for unity gain) at the top of their travel, and metered on the L-R METERS located in the centre of the mixer.

L-R INSERTION POINTS are provided by stereo jack sockets (tip = return) situated on the rear panel, enabling the insertion of compressors, etc., into the stereo mix output.

SYSTEM 8 works with all stereo tape machine operating levels and the stereo output level should be set to match tape machine line level as described in the OPTIONS Section of this Handbook.

TRACK MONITOR PANEL

The track monitor panel is located on the right-hand side of the mixer (above the output faders) and enables you to set up a stereo monitor mix with cue and echo. There is a track monitor section for each output, and each section can be switched to monitor either an output group of the mixer, or the correspondingly-numbered tape track.

Each track monitor is provided with a LEVEL pot, a PAN pot, two Auxiliary Send pots, a Group/Tape select switch and a

pushbutton AUX 2/AUX 3 switch.

We shall now describe the controls and their functions of the eight identical track monitors:



TAPE SWITCH. This pushbutton is located at the bottom of each track monitor section, below the PAN pot. In its normal (up) position, the track monitor will monitor the appropriately-numbered output group of the mixer (i.e. track monitor 7 will monitor group 7). Depressing this enables you to monitor the correspondingly-numbered tape track (i.e. tape track 7). This switch also controls the source selection (group or tape track) of the correspondingly-numbered meter.

You will always be metering the monitor source you have selected.

LEVEL POT. The LEVEL pot is located at the top of the track monitor section. This controls the level in your stereo monitor mix of the monitor signal from the output group (or tape track), depending on the position of the TAPE switch.

FADER REVERSE. This pushbutton exchanges the LEVEL pot with the OUTPUT FADER to allow more accurate fader control of the level of the monitor signal in your stereo monitor mix. The group output level to the multi-track recorder is then controlled by the LEVEL pot. This facility is available for track monitors 1 to 8 only.

PAN POT. The PAN pot controls the sound position in your stereo monitor mix.

AUXILIARY SENDS. There are two Auxiliary Send pots and one pushbutton switch located above the PAN pot.

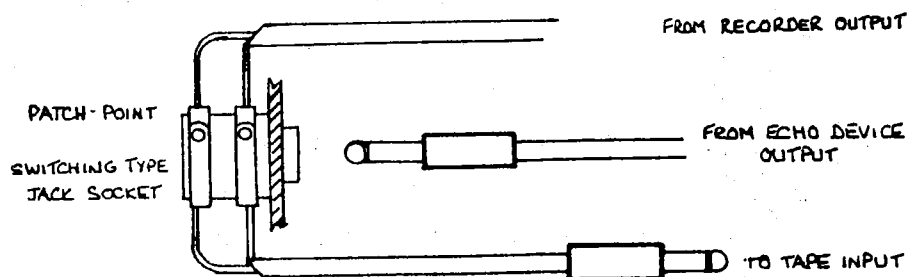
AUX 1 is a fixed PRE-FADE send, feeding the monitor signal to the Aux Send 1 mix from a point *before* the monitor LEVEL pot. This you would normally use as a send to the cue output.

The lower AUX pot is a fixed POST-FADE send, which can be assigned to either Aux Send 2 or Aux Send 3 mix busses, depending on the position of the associated AUX 2/AUX 3 pushbutton switch. Depressing this switch feeds the monitor signal to Aux Send 3. With the switch in its normal (up) position, the monitor signal feeds Aux Send 2. This Send you would normally use for Echo Send from tape.

The AUX 1 and AUX 2/3 pots will always be fed from the monitor source (Group or Tape Track) selected by the TAPE switch.

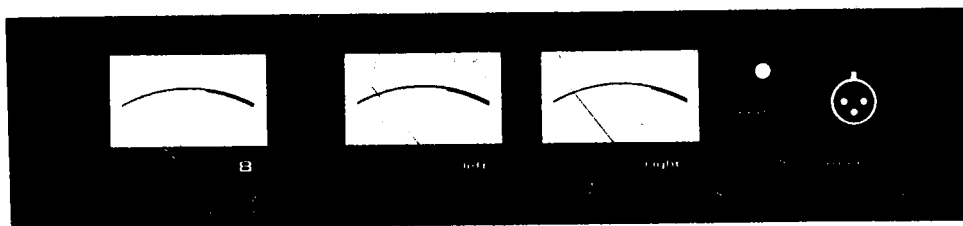
SYSTEM 8 Model 1616 has eight extra meters and track monitor sections so that sixteen tracks can be monitored simultaneously. When not selected to the TAPE position, the extra eight sections are monitoring outputs 1-8, just like the first eight sections.

Monitor sections can have another use when not needed for tape track monitoring. The output of an echo device (or any other signal wanted in the stereo mix) can be plugged into the tape input jack socket and treated like a track. You can make a simple patching system that removes the need to unplug the tape track jack plug first. For this you need a switching-type single-pole jack socket for each tape input. Wire the socket so that when an effect output jack is inserted, the normal tape-track path is broken.



When working SYSTEM 8 in P.A. applications, you can use the tape inputs and monitor sections as extra line inputs to the stereo mix for high level instruments and echo effects.

In this case the mix level may be controlled by the OUTPUT FADER when the FADER REVERSE pushbutton is selected. This applies to tape inputs 1 to 8 only.



MONITORING AND METERING

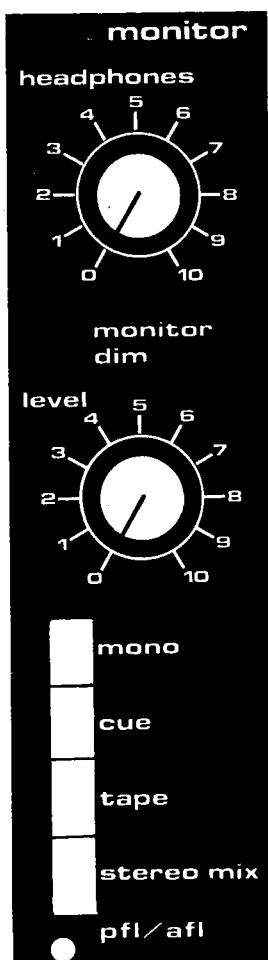
A comprehensive monitoring system is essential for multi-tracking. Under normal conditions you will be listening to a stereo monitor mix of your recording, but you may at any time need to monitor an individual input or output, Auxiliary Send, Echo Return, or the output of the cue system. SYSTEM 8 gives you the capability to switch over instantly to monitor and meter any of the above, without disturbing or interrupting your recording.

The Master MONITOR controls are situated in the centre of the mixer and control the output of the monitor loudspeaker amplifier.

MONITOR MASTER SELECT SWITCHES. These four pushbuttons are labelled (from top to bottom) MONO, CUE, TAPE and STEREO MIX.

Select STEREO MIX to monitor the L-R stereo mix. Stereo mix output levels will be displayed on the L-R meters. The main L-R faders must be up, as they are in circuit prior to the point at which you monitor the stereo mix. When you are recording rhythm tracks or overdubbing, the stereo monitor mix that you build up with the track monitor level pots and PAN pots directly feeds the L-R stereo mix. When you are remixing or recording from inputs direct to stereo, the channels are routed directly to the L-R stereo mix by means of the L-R routing buttons.

Except when replaying a stereo tape or listening to the output of the cue system, you will always be working with STEREO MIX selected. Simply pressing the next source you want over-rides STEREO MIX - you don't need to cancel it.



Select TAPE when you wish to listen to replay from your stereo machine. Tape levels will be displayed on the L-R meters. Selecting TAPE will not interfere with recording - you are merely switching monitor functions. The stereo tape input level should be set to match tape machine line level as described in the OPTIONS Section of this Handbook.

Select MONO and the stereo signal you are monitoring (STEREO MIX, TAPE or CUE) will be mixed to mono, and the resulting mono signal fed to both inputs of your monitor power amplifier. This facility is useful for checking mono compatibility of a stereo mix, as level relationships alter when a stereo mix is heard in mono. This is due to the fact that anything panned dead-centre in the stereo increases in level by 6 dB. Hence a stereo record may sound very different over AM radio! With MONO depressed, both L-R meters will read identically.

The L-R stereo meters will always display the monitor source you have selected, except if you have depressed a PFL or AFL button, when they will display the appropriate PFL or AFL level.

MONITOR MASTER LEVEL POT. This control is located immediately above the monitor master select switches. Use it to set your loudspeaker listening level.

MONITOR DIM SWITCH. This is a pushbutton mounted above the monitor master LEVEL pot. Depressing it attenuates the monitor output to your monitor power amplifier by 20 dB; useful when you need to temporarily reduce listening level without having to reset the monitor LEVEL pot. Operation of the talkback system will automatically 'dim' the monitoring, to allow two-way conversation without feedback problems.

HEADPHONE LEVEL POT. This control is located above the monitor DIM switch, and allows you to set a suitable level when monitoring on headphones. The headphone output of the mixer automatically follows the monitor master select switches. You can use headphones with impedances from 8 ohms upwards, and the jack socket (located on the meterpod) is a stereo output. Use the headphone output to check the quality of the cue output as the performers hear it, and for monitoring

mixer outputs under any conditions when loudspeakers are not suitable.

MASTER SECTION

The central section of the mixer has the master controls and facilities needed for ECHO and CUE mixing, and for TALKBACK.

AUXILIARY SENDS. SYSTEM 8 has three Auxiliary Send mix busses, for cue mixes and to provide access to outboard echo and effects devices.

AUX 1 is a fixed PRE-FADE Send, fed by the AUX 1 pots on the channels and track monitors.

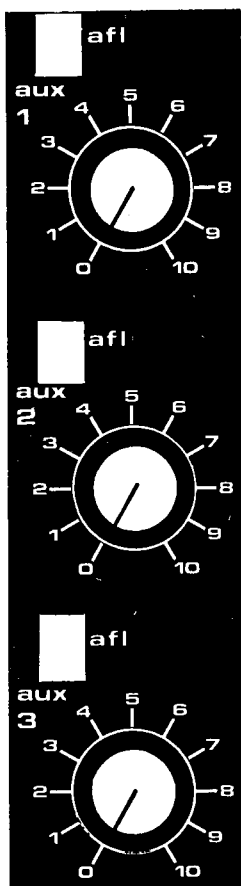
AUX 2 is a fixed POST-FADE Send, from the track monitors, but is fed from the channels either pre-fade or post-fade depending on the position of the individual AUX PRE/POST switches.

AUX 3 is a fixed POST-FADE Send, from the channels and track monitors.

The master output level controls for the three auxiliary mixes each have a pushbutton, marked AFL, which selects the output to the monitor output for level and quality checking. This works in the same way as a PFL pushbutton, and lets you hear the outgoing mix after the master level control.

Normally speaking, *pre-fade* Sends are used for cue mixes, in order that performers should not be disturbed by the changes in their headphone mix as you alter fader positions on the mixer. *Post-fade* Sends are used to feed external or effects devices such as echo units, etc., where it's normally desirable that the echo signal changes proportionately as the channel fader position is varied (i.e. a voice or instrument going to echo retains the same proportion of direct-signal to echo-signal as that voice or instrument is raised or lowered in the mix).

You will notice that the channels have three Auxiliary Send controls, and the track monitors two. The PRE-FADE AUX 1 mix is common to both channels and track monitors, but the second POST-FADE monitor auxiliary mix is individually switchable between AUX 2 and AUX 3. You can use AUX 1 as a



pre-fade track monitor cue mix, and be operating two separate monitor echo or delay devices on AUX 2 and AUX 3. Thus some track monitors would be sending to the echo device on AUX 2 Send, and other track monitors to the device on AUX 3 Send. However, no track monitor could feed AUX 2 and AUX 3 simultaneously.

The arrangement of rear panel output connectors for Cue and Echo circuits deserves attention. It is designed specially to assist in quick operation by minimising the re-arrangement of the connexions to the various outboard echo devices and amplifiers in the typical control room. You will find three jack sockets, marked AUX 1 OP, AUX 2 OP and AUX 3 OP. These are the main outputs from the three auxiliary mixes on which cue and echo output signals are prepared.

There is also the CUE output stereo jack socket and the AUX tie line connectors indicated within the double line connector group. The full versatility of the Cue system is described later. For the moment, the important point is that with CUE going to the headphone or stage foldback amplifiers, all of the three AUX 1 OP, AUX 2 OP and AUX 3 OP connectors are free for permanent connexion to Echo Device inputs, ready for use when overdubbing and mixing. Of course, you may have a patching facility between mixer outputs and echo devices, and the same advantage still applies. You won't need to reach round the back to re-plug in order to use all the auxiliary outputs for echo mixing - they can be plugged up from the start.

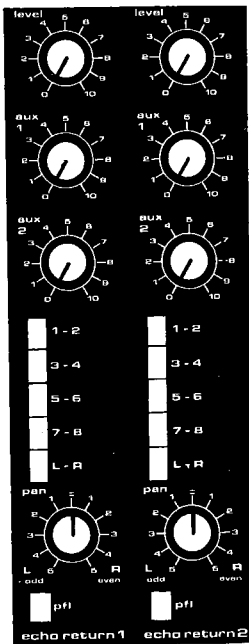
In case you are wondering how AUX 1 can be used as an Echo Send when the input channel AUX 1 control is pre-fade, the answer is that you can also send to AUX 1 from a track monitor section. You simply route to any spare output, open the fader, select the TAPE pushbutton 'out' and open the AUX 1 control in the track monitor section. Keep the LEVEL control closed unless you also wish to add the sound to the stereo mix.

Tie line connectors are multi-purpose, and when not used to parallel the mixing circuits of two mixers the jack sockets are available as duplicate outputs of AUX 1, 2, 3, etc., for connexion in whatever way suits your equipment (e.g. to more

echo device inputs). Refer to the table in the CONNEXIONS section for jack configurations.

ECHO RETURNS

There are two separate returns, located above the L-R faders. Use them to return the stereo output of a reverb plate, or the outputs of two separate echo or effects devices, to the L-R stereo mix or to tape. The routing buttons and PAN controls are identical to those on the channel, and function in exactly the same way. To return an echo device or effect to your mix (or to tape when recording direct to stereo), merely select L-R and pan the return, setting the return levels with the LEVEL pots situated underneath the cue select switches (ER1) and master monitor source select switches (ER2). To record the output of the echo device onto multi-track tape, de-select the L-R button and route the return signal to tape via the group select button and PAN pot.



Each echo return has two Auxiliary Send pots, the upper (marked AUX 1) feeding Aux 1 mix, and the lower (marked AUX 2) feeding Aux 2 mix. Both Aux Sends are PRE-FADE and are not affected by the LEVEL control setting. Echo Return PFL provides the usual quality and level check facility.

Here are several examples of how you can use the Aux Send pots on the echo returns:

EXAMPLE ONE: (This example presupposes that you are the proud possessor of a reverb unit, and either a delay device or a spare tape machine with vari-speed.)

You are remixing, and want to add delayed reverb to a lead vocal track rather than sending it straight to the reverb unit. You cannot plug a tape delay or electronic delay line between the mixer's Aux Send output and the reverb unit because other instruments in the mix require 'straight' reverb without pre-delay.

1. Connect the output of AUX 2 to the input of your reverb unit, and the reverb unit output to ECHO RETURN 1, returned to the mix via the L-R routing button at an appropriate level, and panned wherever you like in the

stereo mix. We suggest that you use AUX 2 as a Send to a reverb unit or other echo device because this does give you slightly more flexibility when all your channels are in use and you wish to send the outputs of effects devices to reverb.)

2. Connect the output of AUX 3 to the input of your delay unit, and return the output of the delay unit to ECHO RETURN 2, returned to the stereo mix via the L-R routeing buttons. (If you possess a spare mono or stereo tape machine, this will function well for delay purposes - and even better if it has a vari-speed control!)

3. Feed your vocal to the delay unit via the vocal channel AUX 3 Send pot at an appropriate level. Set the desired amount of delay return to the stereo mix using the ECHO RETURN 2 LEVEL pot.

4. Finally, on ECHO RETURN 2 open the AUX 2 control. This sends delayed vocal back out to the reverb unit. ECHO RETURN 1 from the reverb unit now has 'straight' reverb on vocals fed out to it from the AUX 2 mix, and delayed reverb on vocals fed out to it from the AUX 3 mix and then the Echo Return AUX 2 control.

Using the above set-up, the delay output is returning to the mix as well as feeding the echo plate. De-select the ECHO RETURN 2 L-R pushbutton if this is not desired. You can do the same thing with channels used as echo return inputs, and you have the advantage of equalisation for the delay and reverb returns.

EXAMPLE TWO: REPEAT ECHO. Connect the output of AUX 2 to the input of a tape machine with vari-speed facility. Connect the replay output of the tape machine to either of the Echo Returns. Use the vari-speed control to set the delay-time for the first reflection, and then use the Echo Return AUX 2 pot to feed the echo return signal back to the tape input, thus giving you multiple delays. Any vocal or instrument added to Aux mix 2 will get the repeat echo treatment. This set-up is also known as 'spin' echo.

If you run out of Auxiliary Sends during remix, for example if you wish to use 4 different effects, remember that you can always route a channel to a group, as well as to the L-R stereo buss, and then use the group output as a send to your effects device. If you do this, pick a group appropriate to the channel pan position *i.e.* if you wish to send to a group when mixing and your channel is panned hard left in the mix, then you must choose an odd-numbered group as an effect send.

If you need more echo returns, do not despair! You can return your effect to the L-R mix through a channel line input. If all channels are in use, you can return your effect to the mix by connecting your effect output to a tape input on the mixer's rear panel, and use the correspondingly numbered track monitor section. With the TAPE switch depressed, of course.

THE CUE SYSTEM



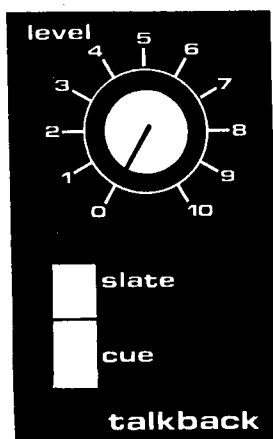
It is essential that performers feel happy with their cue mix (sometimes called a 'Foldback' mix), otherwise their performance may suffer. Often you will be called upon to provide several different cue mixes simultaneously, as musicians on the same session may need totally different mixes in their headphones. Use a good quality power amplifier for your cue system - *Bad foldback quality = unhappy musicians. Unhappy musicians = problems!*

SYSTEM 8's cue system enables you to provide two separate mono cue mixes or one stereo cue mix, and a 'talk to cue' facility allows you to communicate with the performers. Cue mixes may be set up from the channels, the track monitors, a combination of both, or the L-R stereo mix may be selected as a cue source. This can be useful when overdubbing, providing you with an instant stereo cue mix containing any monitor effects you are using on the L-R mix.

Connect the Cue output to the input of a stereo power amplifier. If only a mono cue system is required, connect the left Cue output to the power amplifier. You may also connect the Cue output to two different mono amplifiers for separate use.

Cue system controls are located next to the Monitor Master Select Switches, and consist of the Cue Source Select push-buttons, labelled STEREO MIX, AUX 1 (L) and AUX 2 (R) and MONO, and the Cue Master level pot labelled CUE. Use the Cue Source Select pushbuttons to select either or both of the pre-fade mixes Aux 1 & Aux 2 in MONO, or in STEREO (Aux 1 = L, Aux 2 = R), or select the L-R stereo mix. The following truth table shows you the cue outputs possible using the Cue Source Select buttons singly or in combination:-

<u>CUE SOURCE SELECTED</u>	<u>CUE OUTPUT</u>	
	<u>LEFT</u>	<u>RIGHT</u>
Stereo Mix	L	R
Stereo Mix + Mono	L & R	L & R
Aux 1	Aux 1	-
Aux 1 + Mono	Aux 1	Aux 1
Aux 2	-	Aux 2
Aux 2 + Mono	Aux 2	Aux 2
Aux 1 + Aux 2	Aux 1	Aux 2
Aux 1 + Aux 2 + Mono	Aux 1 + 2	Aux 1 + 2



TALKBACK TO CUE

The talk to cue button (marked CUE) is located above the Master Monitor section, below the Talkback LEVEL pot.

The Talkback LEVEL pot sets the voice level to suit the microphone in use, the speaking distance, and variations in voice loudness. This is how you set the various controls for easy use:

Start by playing some music into the mixer and set up the stereo output to be peaking at 0 VU. Select CUE source, STEREO MIX, and adjust the CUE LEVEL for comfortable listening level in the cue headphones or loudspeakers. This may require adjustment of the power amplifier input controls if fitted. The maximum cue output level is 1.23v (+4dBv) for a 0 VU sine wave. Now select Talkback to CUE and make your announcement. Adjust TALKBACK LEVEL for a comfortable voice level over the cue headphones or loudspeakers. The cue programme automatically dims during Talkback to improve intelligibility. Now when you select any other cue source

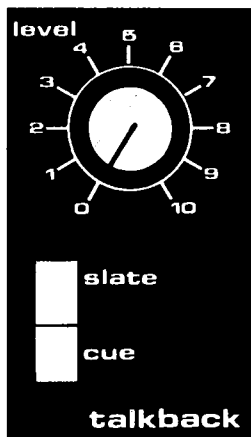
such as any auxiliary mix, the voice level during Talkback is correct for a standard level auxiliary output, which you can check using AUX AFL. You can always check the quality of the CUE output yourself by listening on the monitor headphone output as this is not dimmed during Talkback and is a replica of the mix being heard over the cue system itself.

POINTS TO REMEMBER:

- (i) When using AUX 2 as a channel cue send, ensure that the AUX 2 pre/post buttons on the channels are in the PRE position.
- (ii) When using AUX 2 send pots on the track monitors as a cue send, remember that they are *fixed post-fade*, and any subsequent change in track monitor LEVEL pot positions will produce co-responding level changes in the AUX 2 cue mix!
- (iii) The AUX 1 and 2 mixes are fed to the cue system from a point *after* their master LEVEL pots. Hence when using AUX 1 and 2 as two separate mono sends simultaneously, use their master LEVEL pots as individual cue send masters, having first set the master CUE pot at a nominal level (say 8 on the calibration). When used as left and right sides of a stereo cue system, set AUX 1 and 2 master LEVEL pots at an equal nominal level (say 8 on the calibration), and use the CUE pot to set the master stereo cue output level.
- (iv) Effects devices connected to AUX 1 and 2 outputs need not be disconnected when these Aux mixes are used as cue sends, but check that their return inputs are closed.
- (v) AUX 3 may be used to feed an external cue system, but without Talkback. AUX 1 and 2 may be similarly used as extra cue outputs when the cue system output has the stereo mix selected as its source.
- (vi) To facilitate setting up and checking cue mixes, you can monitor cue system output by selecting CUE as the master monitor source, with the option of headphone monitoring via the headphone output.

(vii) The cue output, selected to STEREO MIX, can also be used to drive an amplifier and pair of small speakers for reference purposes when mixing.

(viii) The cue system outputs can be used to drive a Studio Playback system.



TALKBACK SYSTEM

An XLR socket is provided on the meter panel for the connection of a Talkback microphone. Use any type of dynamic microphone on the Talkback input, or a condenser microphone having no polarising voltage on the audio output. Talkback controls are located above the Master Monitor controls in the centre of the mixer, and consist of a LEVEL pot, and two non-latching pushbuttons labelled SLATE and CUE.

Push SLATE and the Talkback signal will appear on groups 1-8 before the faders, enabling you to vocally 'ident' recordings. This will assist accurate location of takes later.

Push CUE to talk to the performers via their cue mix(es).

Operation of the Talkback system automatically dims the monitors to prevent feedback. For more information see CUE description.



1KHz OSCILLATOR SLATE

The latching 1KHz SLATE pushbutton places a 1 KHz tone at 0 VU onto groups 1-8 at a pre-fade stage. Ensure group faders are at the top of their travel when using the

oscillator to align tape recorders. This facility enables you to make a quick tape machine record/replay level check. To route the oscillator to the L-R stereo buss, first route the oscillator to groups 1-8 by depressing the SLATE button, then monitor two of these groups on the track monitors, panned hard left and right across the stereo buss.

The 1 kHz tone is preset in level, has low distortion and can be relied on for all level matching purposes associated with tape recorders. A well-aligned tape machine will show 0 VU replay level on its own meters when the record level was also 0 VU. To show the tape-machine output level on the mixer meters, select track monitor TAPE for each track. If your machine operates at 0 VU = 4 dBv (1.23v), the mixer internal level match system should be set in the high position. Outputs 1-8 then give +4dBv (1.23v) output level at 0 VU. If your tape machine operates at 0VU = -8dBv (0.31v) or 0 dB (0.775v) the mixer level match should be set to the low position. Refer to the OPTIONS section for details.

USING SYSTEM 8 FOR P.A. AND SOUND REINFORCEMENT

In a live-sound situation, you will be driving your main P.A. from the L-R stereo outputs. Channels can be routed direct to the L-R stereo mix, or routed to groups if you wish to assemble sub-groups. When sub-grouping, remember to have the corresponding track monitor TAPE buttons in the 'up' position. Sub-groups will be routing to the L-R stereo mix via the track monitor LEVEL pots, so you may prefer to work with these pots fully open and the group faders around the -10 dB mark rather than working with the groups up and having to mix sub-groups together using the track monitor LEVEL pots.

The cue system will provide your stage monitor sends, with talkback. Read the earlier section covering the cue system, and refer to the truth table for possible cue outputs.

Track monitors not being used for sub-group purposes may be used as effects returns by switching them to monitor TAPE,

and returning the effects outputs to the mixer via the tape inputs on the rear panel.

If it is desired to limit the stereo output in the interests of loudspeaker protection, limiters can be inserted across the L-R stereo mix using the stereo group insertion points.

The headphone output can be used to monitor PFL and AFL signals, and the PFL and AFL levels are of course displayed on the L-R meters. Aux Sends may be used as additional cue sends without talkback and as Echo sends.

The EX-8 input expander unit is self-powered and connexion to the main unit is at high level via screened cable. The expander can be sited at any suitable distance from the main unit. A connexion schematic is shown under INSTALLATION section. EX-8 outputs contribute to the main mixes in the same way as input sections on the main mixer. Signal phase is preserved.

Two SYSTEM 8 mixers, when joined for expanded capacity using the BUSS TIE LINE system will have the master controls (group faders, auxiliary masters) in circuit between inputs mixed on the subsidiary unit and inputs mixed on the main unit. You can use these master controls to create sub-mixes with master level control. The subsidiary unit retains its own auxiliary output connectors which are free for connexion to cue and echo systems independent of the main unit.

LEARNING YOUR WAY AROUND

[IMPORTANT - Several tape-machine manufacturers' products operate at a line level of -8dB (TEAC, FOSTEX, etc.). When using SYSTEM 8 with one of these tape-machines ensure that you have set the level-match options correctly (as detailed in the OPTIONS Section). When replaying -8dB line level tape tracks through the channels in Line mode, set the input gain pot to 7 on its calibrated scale.]

So that you can learn your way around SYSTEM 8, get a friendly musician to be your guinea-pig!

First, after reading the previous section explaining the controls and their functions, set the controls of the mixer to a neutral state in the following manner:

- (i) Set all input GAIN pots, AUX SEND pots and track monitor LEVEL pots to their minimum positions,
- (ii) Set all PAN pots and equaliser controls to their mid position,
- (iii) Ensure that all pushbuttons are 'out' (undepressed),
- (iv) Set all channel faders to minimum and group faders and stereo group faders to the top of their travel,
- (v) Set AUX SEND master pots and the CUE master pot to around 8 on their calibrated scales,
- (vi) Select STEREO MIX on the Master Monitor Source pushbuttons.
- (vii) Set the master monitor LEVEL pot to around 6 .

The mixer is now in a neutral, or 'flat', state. Get into the habit of 'flattening' the mixer before every session, thus preventing the previous session's settings from interfering with your recording, and possibly causing havoc!

Now plug a microphone into Channel 1, and place it near your friendly musician! Select MIC and depress the PFL button. You will now hear your sound source over the monitor speakers

and read PFL level on the L-R meters. With the PFL button depressed, adjust the input GAIN pot to give a level on the L-R meters around '0' VU. Adjust the master monitor LEVEL pot to give a comfortable listening level.

Route the channel to the L-R stereo buss by depressing the L-R button, having first released the PFL button. Bring the input fader up to its '0' mark and use the PAN pot to pan your signal across the monitoring and L-R meters. Experiment with the equaliser controls to familiarise yourself with their response. Release the L-R button and assign the channel to a group or groups using the routing buttons and PAN pot. With the channel routed to a group, monitor via the track monitor LEVEL pot and PAN pot, first checking that the group fader is up and the TAPE button is out on the track monitor. Observe group output level on the relevant meter (you can also depress the group PFL button and monitor group PFL on the L-R meters). Now select CUE on the Master Monitor Source Selects and you will be monitoring cue system output. Depress one or both of the AUX 1 and 2 cue source Selects. You can now rehearse sending to the cue system from the Aux Send pots on the channels and track monitors, and cue levels will be displayed on the L-R meters.

Remember that the Aux Send master LEVEL pots and the cue master LEVEL pot will affect cue level. Depress the AFL buttons on the Aux Sends so that you can meter Aux Send outputs on the L-R meters. Release these after use.

If you have an echo unit or similar device hooked up to the output of either AUX 2 or AUX 3, feed the signal to the appropriate send and, returning to STEREO MIX mode, bring the device output back to the monitoring via an echo return routed to L-R. Adjust echo return level using the LEVEL pot, and pan across the monitors.

Next, record your signal source onto a track of the 8 track machine and replay it (with the TAPE button depressed on the relevant track monitor when you want to listen back off tape). Now turn down the LEVEL pot on that track monitor and route the tape track playback signal to the L-R stereo mix using the

corresponding channel in LINE mode, with the L-R button depressed.

You have now observed for yourself practically the basic mixer functions - you should now do some recording!

PRACTICAL 8-TRACK RECORDING HINTS

The following section is intended to help you by passing on some practical tips, but we strongly advise you (should you not already possess one) to invest in one of the several excellent recording manuals currently available. We can especially recommend THE RECORDING STUDIO HANDBOOK by John M. Woram, Editor of dB magazine.

KNOW YOUR MONITORS. Every monitor speaker has a sound of its own, and so does the room in which it is located. Listen to as many good discs as possible over your monitor system (you can replay a disc pre-amp output through the mixer's line inputs). Learn what top engineers and producers consider to be good instrumental and vocal sounds by listening to their recordings on disc. You MUST learn what a good recorded snare drum sounds like in order to make good snare sounds yourself, for example. Remember that different styles of music require different sounds and recording approaches. Your tight, funky pop drum sound will not be appreciated by the heavy-metal rock band you may be recording tomorrow!

In general, commercial pop music, disco and funk require a tight, direct sound. Rock, Jazz and classical music lend themselves to a more ambient treatment. However, stealing a trick from one recording style and using it in another can be very effective! REMEMBER, THERE ARE NO RULES. The ears must be the ultimate judges over any recording.

MICROPHONES. Use as few microphones as possible in any given situation. Phase cancellations occur when multi-miking takes place, causing signal deterioration. As a rule of thumb, keep

the distance between microphones three times the distance between the microphones and their sound sources.

Always listen to an instrument acoustically before placing a microphone on it, to see where it sounds nicest as you walk around. Place the microphone at that point. Always get the best possible sound from an instrument amplifier or a drumkit BEFORE you go for the equalisers.

CARDIOMD mics have a tight pick-up pattern, but off-axis leakage from other instruments can result in muddy sounds especially if you are using screens. *OMNI-DIRECTIONAL* mics have a smoother response and are less prone to 'popping'. Unlike Cardioids, they have no proximity-effect bass boost, so they may be used close-up without sounding 'boomy'. However, for some applications they may pick up too much ambience.

Experiment with mics and mic placement. Let your ears be the judge and you will soon discover mics you can relate to.

CONDENSER mics have a high output and good transient response, but *DYNAMIC* mics are rugged and can withstand high sound-pressure levels, making them ideal for drumkits and loud guitars!

MONITOR MIXES AND CUE MIXES. Always make the monitor mix as good as possible. When overdubbing, try to work with a monitor mix as close as possible to your conception of the final mix. This will help you equalise overdub instruments 'in context'. Make sure that you have enough bass and drums in the monitor mix and cue mix - this will ensure 'good vibes' on playbacks, and good performances from the artistes. Check that the artistes are happy with their cue mixes, as happy performers sound better! This is a well-known psycho-acoustic phenomenon!!

TAPE MACHINES. The importance of correct tape-machine alignment on multi-track and stereo tape machines cannot be over-emphasised. Use a reputable brand of tape, have your machines aligned for that brand, and stick to it.

Incorrect tape-machine alignment of bias and record/replay

levels and frequency response causes increased noise and distortion and a false frequency response (i.e. sounds come back off tape sounding different and invariably worse). The popular compander noise-reduction system must see a correct sync/replay level, or their decode systems will multiply level anomalies through severe tracking errors.

Try and invest in multi-track and stereo test tapes and an oscillator, and learn to align your machines yourself. Do it regularly, and clean the tape paths and heads at least once a day to avoid wow and flutter and undue wear.

If your tape-machine has automatic mode switching and remote switching controls, use your SYSTEM 8 with the TAPE buttons depressed, so that all monitor mode switching is performed by the tape-machine. You can still monitor the groups at any time by releasing the TAPE buttons (for example, when you wish to repair a track previously recorded on tape and need to compare levels for compatibility purposes, or when you want to rehearse an overdub or punch-in without recording).

TRACK ASSIGNMENT. When initially planning track assignments, try to plan ahead as far as possible to avoid chaos later! Decide if your rhythm instruments will remain on the tracks you select, or if you will be 'bouncing' them down to two tracks or so before overdubbing. Note in advance that you cannot transfer a track already on tape over to an immediately adjacent track. More about that later

A typical input channel set-up for a rhythm section session might well look like this:

<u>INPUT CHANNEL</u>	<u>TRACK ROUTED TO</u>
1. BASS AMP or D.I.	2
2. BASS DRUM	3
3. FLOOR TOM	4
4. HANGING TOM	4
5. SNARE	4
6. HI HAT	4
7. CYMBAL (L)	4
8. CYMBAL (R)	4
9. GUITAR AMP	5

10. PIANO - Low end	6
11. PIANO - High end	6
12. GUIDE VOCAL	1

This set-up leaves tracks 7 & 8 free for overdubs, which can be bounced to track 1 after the guide vocal has been erased, freeing tracks 7 & 8 again for more overdubs.

If you are intending to bounce your rhythm section tracks to stereo before overdubbing, you could record the drumkit across four tracks of the machine and the piano in stereo across two, sending the guide vocal to the cue mix without recording it onto tape.

When bouncing rhythm tracks down to stereo, you have two choices. You can record a maximum of five tracks and bounce to two tracks of the multi-track machine, keeping one track as a barrier track (to avoid adjacent bouncing!) OR you can fill up all eight tracks with rhythm instruments and bounce them down to a stereo tape-machine and then copy them back onto a clean piece of multi-track tape. You would then have the ability to use up some of the remaining six tracks with overdubs, after which you could bounce some of these overdubs together, thus freeing some tracks for more overdubs

Develop tidy routines when doing lots of bouncing, as this can tighten and clean up your recordings considerably

AN ALTERNATIVE METHOD. During recent years, a new system of recording has evolved which has great relevance for 8-track users - despite the fact that it was mainly developed on 24-track sessions! It could almost be thought of as 'multi-tracking backwards', and offers you as an additional bonus unlimited separation acoustically and strong creative control over the recording. Interested? Then read on

Basically, a 'drum' track is laid down on one track using a rhythm box or drum machine, followed by a guide instrument, e.g. a guitar or keyboard. and then, say, a bass guitar track and a guide vocal track.

At this point, the track sheet will look something like this:

TRACK 1	-
TRACK 2	-
TRACK 3	BASS
TRACK 4	RHYTHM BOX
TRACK 5	GUIDE GUITAR or KEYBOARD
TRACK 6	GUIDE VOCAL
TRACK 7	-
TRACK 8	-

Then, a *first* vocal harmony part could be recorded on track one, double tracked on track 2, and the two tracks then combined together onto track 8.

A *second* vocal harmony would then be recorded onto track one, double-tracked on track 2, and these tracks combined together onto track 7.

Now a *third* vocal harmony can be recorded onto track one and double-tracked on track 2, and these tracks combined together onto track 6 (after the guide vocal on that track has been erased). You could regard tracks 1 and 2 as 'working tracks'.

The track-sheet will now look like this:

TRACK 1	-
TRACK 2	-
TRACK 3	BASS
TRACK 4	RHYTHM BOX
TRACK 5	GUIDE GUITAR or GUIDE KEYBOARD
TRACK 6	BACKING HARMONY VOX 3 (Combined)
TRACK 7	BACKING HARMONY VOX 2 (Combined)
TRACK 8	BACKING HARMONY VOX 1 (Combined)

At this stage the harmony backing vocals can be bounced to track one, and the drums now overdubbed on tracks 6, 7 and 8, allowing the rhythm box on track 4 to be then replaced by a master rhythm guitar track. The master keyboard instrument can now be recorded on track 5, and the lead vocal onto track 2. An instrumental solo might then be dropped into track 2 which would be heard in a section of the music where the lead vocal had nothing to sing. Track one might now be overdubbed in part with a percussion track, leaving the completed track-sheet looking like this:

TRACK 1	PERCUSSION and BACKING VOX
TRACK 2	LEAD VOX and SOLO
TRACK 3	BASS
TRACK 4	RHYTHM GUITAR
TRACK 5	KEYBOARD
TRACK 6	DRUMKIT
TRACK 7	SNARE DRUM
TRACK 8	BASS DRUM

If you are more ambitious, you might have delayed recording the drums, and instead used tracks 6, 7 and 8 to record some synthesiser parts, bouncing them together before proceeding with more overdubs.

When two or more instruments are sharing a track, you will find at the mixdown stage that they normally need different control settings. Make up a suitable parallel connecting cable to enable the tape track to be fed into the mixer along two channels, one for each instrument. Then, during mixdown, swap between channels so that as one channel is feeding the mix, the other is muted.

BOUNCING TRACKS. You should by now be familiar with the principles of track-bouncing, so here's the procedure for combining tracks. Your aim is, of course, to combine several tracks already down on tape and transfer them (having combined them across to another track of the tape-machine, or panned across a pair of tracks for a stereo bounce).

- (i) Ensure the track you are bouncing to is really vacant, and not a vital overdub.
- (ii) Ensure that the track you are bouncing to is clean (erase it anyway, to make sure!).
- (iii) Bring the tracks you wish to combine back from tape via the channel LINE inputs, and check that the track monitor LEVEL pots on those tracks are turned to minimum, to avoid monitoring duplications.
- (iv) Route the channels to the desired tape track(s) via the relevant group(s) and PAN pot(s), equalising the channel signals if desired. You will be monitoring the track monitor(s) switched to the group(s) you are using,

with the TAPE button(s) out. Set a good level to tape with the group fader(s).

(v) Keep your bounce clean by marking the group fader level, and then pulling the group fader down. When you enter record mode, only lift the group fader to its mark for the sections you require. If you goof, rewind a few seconds and punch in during a gap.

(vi) Mute the channels being combined before replaying the master track you have just made up, to avoid possible monitoring errors.

(vii) If all is well, place your master track in "safe" and erase the source tracks.

When you return tape tracks to the L-R stereo buss via the channels when overdubbing, you have the advantage of using them for monitoring and cue purposes, enhanced with equalisation. This is referred to as 'dubbing through the board'.

CUSTOMER OPTIONS

INTRODUCTION - Available Options.

1. PHANTOM POWER SUPPLY:

Individual mic input phantom power selection.

2. TAPE LEVEL MATCHING:

(a) Outputs to multi-track/inputs from multi-track.

(b) Outputs to stereo recorder/inputs from stereo recorder.

3. BALANCED OUTPUT OPTION:

(a) Balanced group outputs.

(b) Balanced stereo outputs.

4. AUXILIARY SEND CONFIGURATION:

(a) Aux 1 pre- or post input equaliser.

(b) Aux 2 pre- or post input equaliser.

5. INPUT EXPANDER SYSTEM EX8:

(a) Fitment.

(b) Connexions. (See INSTALLATION Section of this book).

These options are included to make SYSTEM 8 right for your operation. Options 3. and 5. involve additional parts which must be ordered and fitted by a competent technician. Options 1., 2. and 4. are straightforward and re-configure the standard system. Some need only the use of a screwdriver, others will need the skill of a competent technician to make solder connections.

Instructions follow for the successful completion of these important options.

OPTION 1 - PHANTOM POWER

Standard mixers are shipped with the 'PP' link on each input pcb assembly fitted so that when switched on the power reaches all microphone input terminals.

Individual microphone inputs can be isolated from the power by removal of this link as follows. You will need soldering iron, Pozi 2 screwdriver.

Disconnect mixer power.

Set mixer upside down or on end on the workbench. If on end, put the output end lower and support the unit to prevent damage.

Release all base plate screws and remove base plate.

Release three front panel screws holding the left hand side plate to provide access to input one if necessary.

Link 'PP' is located immediately behind the switches PAD and MIC/LINE one each input pcb assembly. See figure

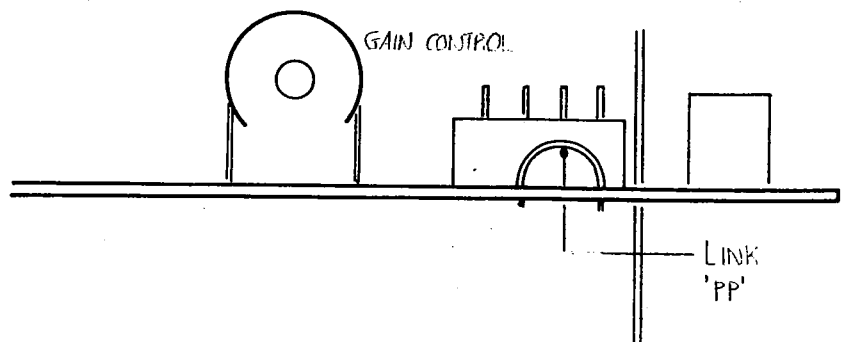
De-solder each end of the links to be removed, and discard them.

Check for solder bridges and debris around each link area worked on, and clear if necessary.

Make a note of the channels isolated for later reference.

Replace side plate and base plate.

To be safe re-check your work by measuring for voltage on the channels worked on.



OPTION 2 - TAPE LEVEL MATCHING

2(a) Multi-track level matching

Internal links adapt the group outputs 1-8 and tape inputs 1-16 to give correct signal level for record, listening and metering from the multi-track machine. Standard mixers are factory set for operation at OVU = -8dBv (0.31v).

The alternative level of operation is 12dB higher OVU = +4dBv (1.23v) and matches many tape machines in the high cost bracket.

To change the level matching option you will need only a screwdriver, Pozi No 2. The procedure is as follows:

Disconnect mixer power.

Set mixer upside down or on end on the workbench. If on end, put the output end lower and support the unit to prevent damage.

Release all base plate screws and remove base plate.

Level match adapter plugs are located halfway down each output PCB assembly. See figure 13.

Unplug adapter and replace on alternative set of pins. This resets the group output and both tape inputs associated with that group to the new level. Repeat for all eight groups.

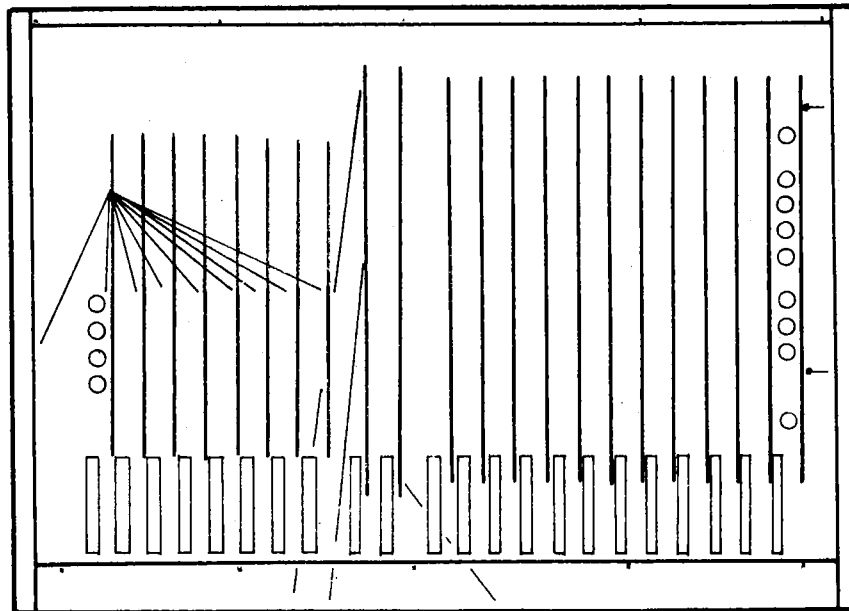
Check each plug for correct placement over the pins. Replace base plate.

Reconnect and check out by selecting slate 1kHz (master section), put output faders up and record a section of tone. Check replay on mixer meters (select tape 1, etc.) and adjust tape machine output level controls if necessary for OVU readings.

Tape machines operating at OVU = 0dBv will work best when the mixer is set for OVU = -8dBv operating level. Adjust tape machine record and replay level controls for mixer OVU on replay.

OUTPUT PCB ASSYS. INPUT PCB ASSEMBLIES
MASTER PCB ASSYS.

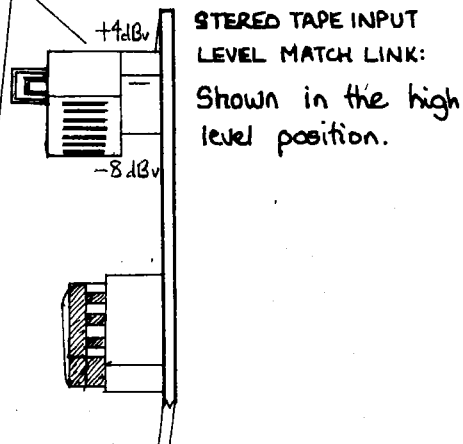
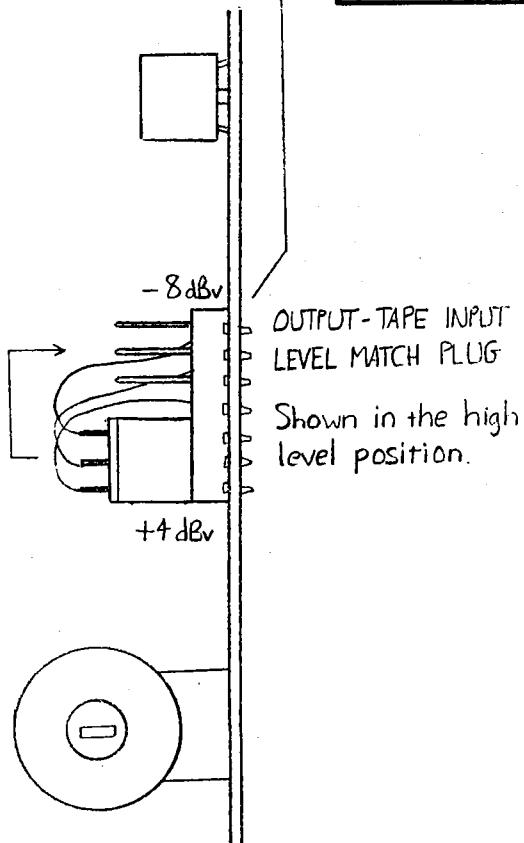
internal view



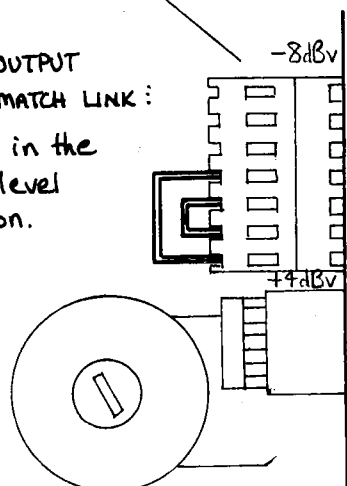
OPTION 1
Phantom Power

OPTION 4
Aux sends

OPTION 2
Level matching



RIGHT OUTPUT
LEVEL MATCH LINK:
Shown in the
high level
position.



LEFT OUTPUT LEVEL MATCH LINK:
Shown in the high level
position.

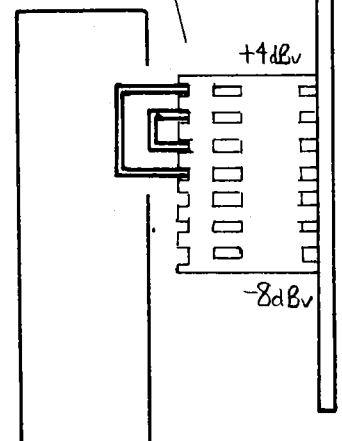


Fig 13.

2(b) Stereo tape recorder level matching:

Internal links adapt outputs left and right and stereo tape inputs left and right to give correct signal level for record, listening and metering of the master stereo tape recorder. Standard mixers are set at time of manufacture for operation at $OVU = -8dBv$ (0.31v)

The alternative operating level is 12dB higher at $OVU = +4dBv$ (0.775v).

The level matching for stereo output and stereo tape input are independent and so can be set for any permutation of high or low level output and high or low level input.

To change the level matching option you will need only the correct screwdriver pozidrive No 2. The procedure is as follows:

Disconnect mixer power.

Set mixer upside down or on end on the workbench. If on end, put the output end lower and support the unit to prevent damage.

Release all base plate screws and remove base plate.

The level matching links are located on the Master Left and Master Right pcb assemblies and are shown on figure

One link P1 on each pcb sets the output level either L or R. Lift and replace it the other way round to change the setting.

One link P2 on the Master Right pcb assembly sets the input matching for stereo tape input L and R. Lift and replace it the other way round to change the setting.

Check the links you have changed for correct placement over the pins.

Replace the base plate.

Check out the change using the mixer 1kHz slate oscillator with the stereo recorder.

Select 1kHz slate on.

Raise output fader 1 to maximum. Check meter one reads 0 VU.

If there is no reading, check that TAPE 1 pushbutton is out. Repeat for output 2.

Set monitor LEVEL 1 and 2 to maximum. Set monitor PAN 1 hard left and monitor PAN 2 hard right.

Select monitor source STEREO MIX and raise Left and Right faders to maximum.

Check L and R meters to read 0 VU.

On tape recorder, check meters and adjust if necessary on the recorder input controls for 0 VU reading.

Record a section of tone.

Replay tape and select monitor source TAPE.

Check that mixer L and R meters read 0 VU and adjust recorder replay if necessary.

This completes the check out.

OPTION 3 - BALANCED OUTPUT OPTION

3(a) Balanced group outputs

The standard group outputs are unbalanced on XLR connectors. All SYSTEM 8 MK3 models can be ordered with the balanced group output option by specifying at the time of order. This is achieved by adding one PCB assembly per group to provide electronic transformerless balancing to the existing XLR outputs.

It is also possible to retro-fit output balancing to all (or a few) groups. The parts required to do this are available in a kit. Order as follows:

SYSTEM 8 GROUP BALANCE KIT (order qty 1 to 8)

The work required to fit and commission this option is best left to a qualified service engineer. Your Sales Agent will be pleased to quote for this to be done.

To fit the option it is necessary to disconnect the mixer and remove the base plate to provide access to the group PCB assemblies and XLR connectors. Full fitting instructions are provided with the kit.

3(b) Balanced stereo outputs

The standard stereo outputs are unbalanced on XLR connectors. All SYSTEM 8 MK3 models can be ordered with the balanced stereo output option by specifying at the time of order.

It is also possible to retro-fit the balance option to the two stereo outputs. The parts required to do this are available in a kit. Order as follows:

SYSTEM 8 STEREO BALANCE KIT (order 1 pair)

Fitting this option is as fitting the group balance option. See notes above.

OPTION 4 - AUXILIARY SEND CONFIGURATION

Input Auxiliary Send controls are wired at time of manufacture to receive equalised signal from the input preamp. Certain P.A. applications call for the auxiliary mix to be equalised independently of the treatment given to input channels. This requires the Auxiliary Sends from the input preamp to receive un-equalised signal. On each input pcb assembly a link is provided for the pre-fader Auxiliary Sends 1 and 2 to be reconnected pre-equaliser. The procedure is as follows:

You will need a screwdriver (Pozidriv™ No. 2), soldering iron, long nose pliers.

Disconnect mixer power.

Set the mixer upside-down or on-end on the workbench.

If on-end, put the output end lower, and support the unit to prevent damage.

Release the screws securing the base plate and remove base plate.

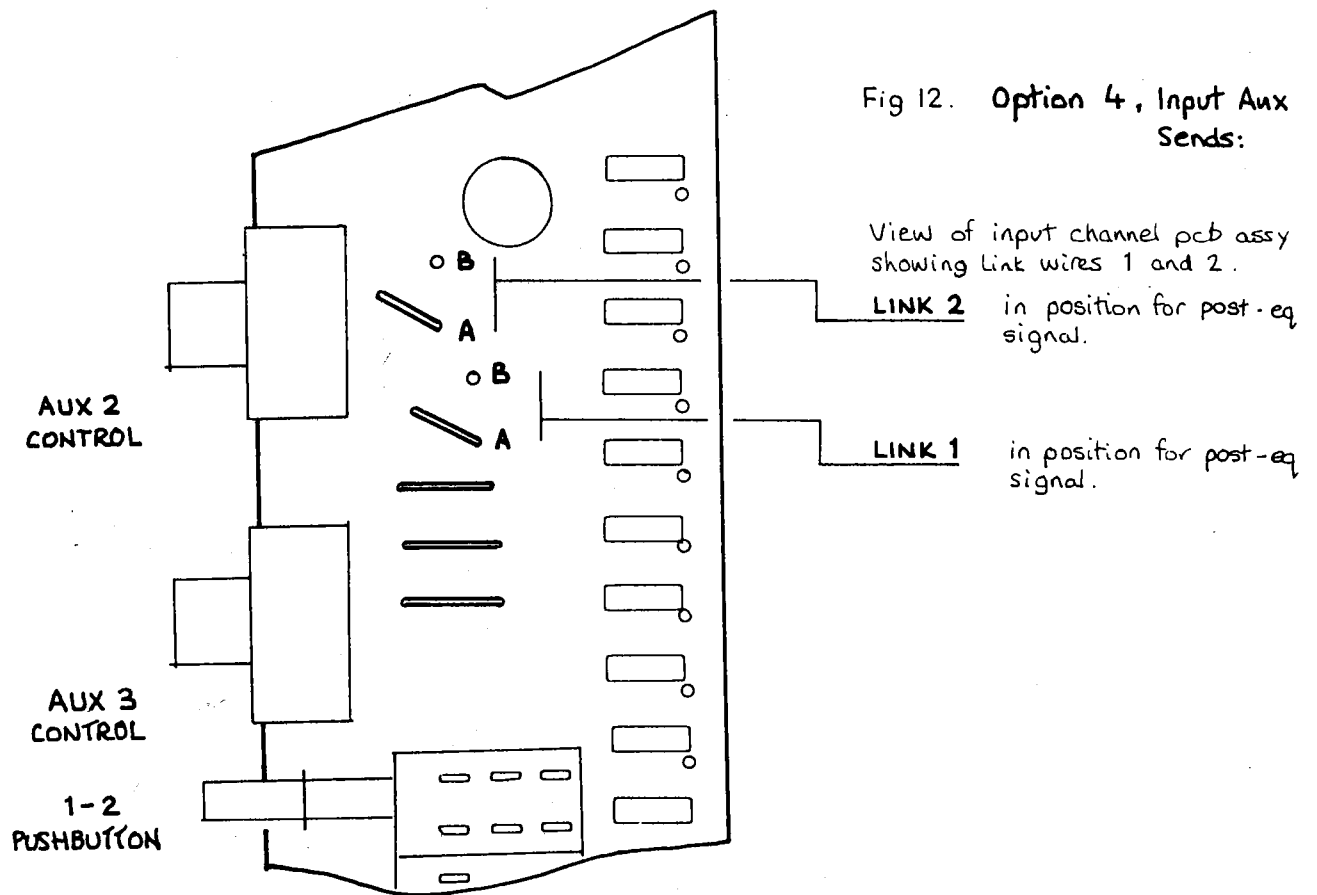
If input 1 is to be modified, also release the three screws securing the side plate at that end.

Link 1 (Aux Send 1) and Link 2 (Aux Send 2) are bare wire loops behind Aux 2 control (see figure

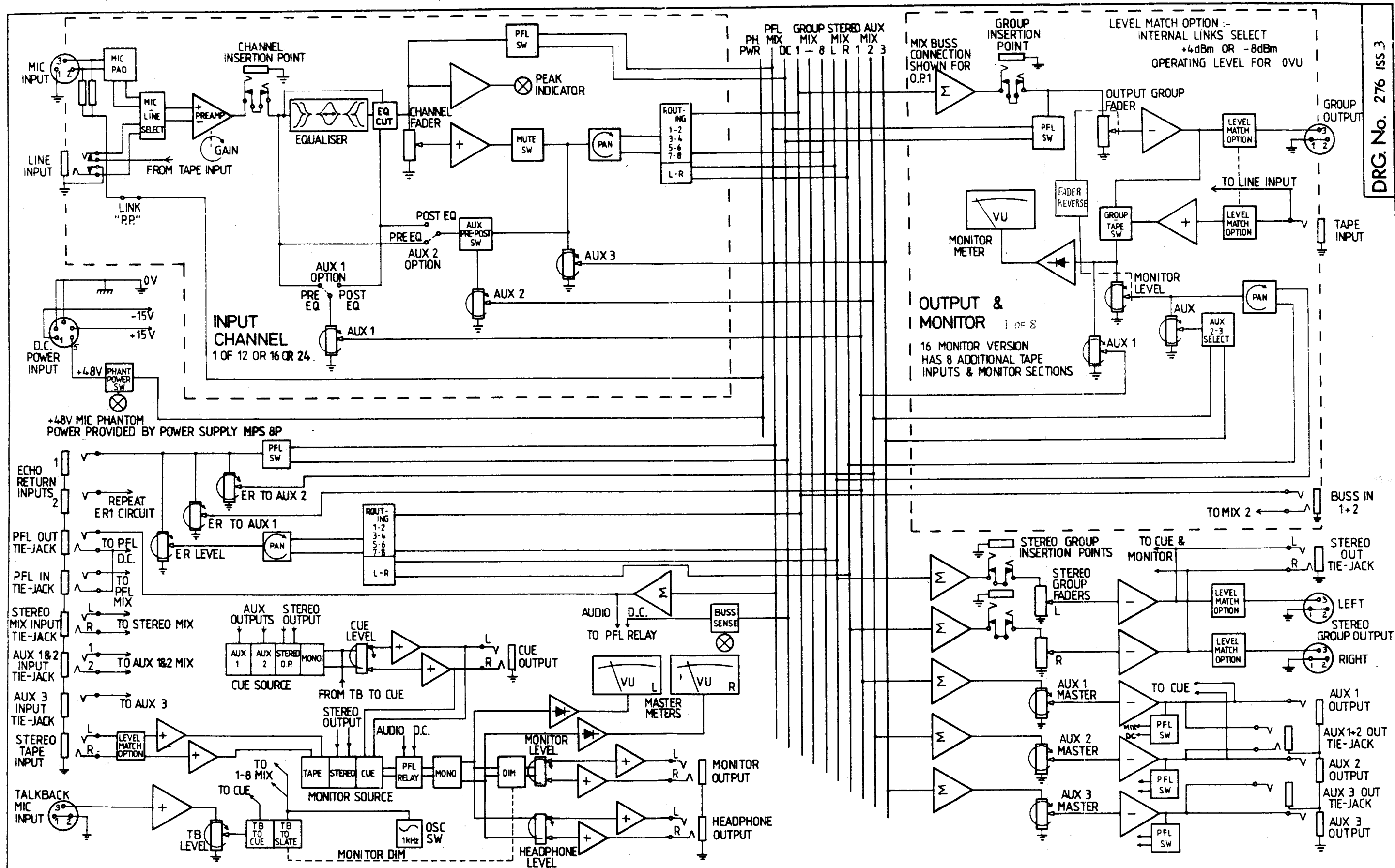
Use the soldering iron to melt the joint at pad 'A' on one end of each unit. Bend the link across to pad 'B' and put it through the hole. Re-solder in position 'B'. Long-nose pliers are needed to reach and bend the links.

Check each modified link and clear any solder debris.

Replace side plate and base plate.

OPTION 5 - INPUT EXPANDER SYSTEM EX8

See INSTALLATION Section page 24.



CD	MK2 FADER REVERSE	ISSUE 2	6-686
No.	DESCRIPTION	CHKD APPD.	DATE
REVISIONS			

SYSTEM 8: SYSTEM SCHEMATIC mk 3

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DRAWN	TRACED	CHECKED	APPROVED	DATE	SCALE
R	MA	R	R		
DRAWING No. 276 ISS 3 6-86					