FORMULA SOUND

PM-90

PRODUCTION MIXER

USERS MANUAL

PM-90 INTRODUCTION

The PM-90 is a compact stereo production mixer of modular construction which combines exemplary technical performance with the highest standards of mechanical engineering. A choice of modules allows for each mixer to be individual to the needs of the user.
3 chassis sizes are available - 4 channel, 8 channel (19" rack mounting) and 12 channel.

The system is totally modular, each module being complete in itself. The connections to the various modules are via a flexible 26 way ribbon cable featuring computer grade gold plated connectors. The chassis contains no electronics, therefore upgrading to a larger chassis size is quite straightforward.

As standard the output module is fitted to the right hand end of the chassis with the input modules arranged in any order to the left. Blank modules are inserted where input modules are not required.

POWER SUPPLY

The mixer requires an external power supply unit providing regulated supplies of ±18V & 6V.

A suitable power supply unit, the PSU4, is available. This is short circuit protected and features slow turn on for extra reliability. The unit is externally switchable to operate from 220-240V AC or 110-120V AC.

This power supply must be connected to a suitable earth and adequate ventilation provided. Do not cover.

Check that the voltage selector switch is set to the correct mains voltage before connecting power.

DAMAGE MAY RESULT IF THE UNIT IS CONNECTED TO THE WRONG SUPPLY VOLTAGE

CONNECTING THE POWER SUPPLY

Ensure that the power supply unit is disconnected from the mains supply. Using the captive lead connect the mixer to the power supply unit ensuring that the CONNECTOR IS FULLY MATED and that the RETAINING CLIPS are FITTED BEFORE CONNECTING MAINS POWER.

MONO INPUT MODULE

This module will accept either low impedance balanced microphone, or balanced line input signals. A switch on the rear selects mic or line inputs.

CONNECTORS (Rear Panel)

Mic/line inputs	3 pin XLR	PIN	1 Screen
			2 Phase
			3 Non Phase
Insert (Send/Return)			1/4" 3 Pole Jack
Control			3.5mm 2 pole Jack

INSERT (Send Return)

An insert break jack is accessible on the rear panel to provide the facility to use extra outboard equipment to process the signal. Virtually any signal processing equipment currently available requiring line level signals may be connected. Usually a 'Y' lead will be required for this connection. Twin individually screened cable should be used. The figure 8 type is readily available. (See Fig 1). When connecting between the PM-90 and other equipment take care to avoid ground loops. (See the section on inter-connections and good wiring practice).

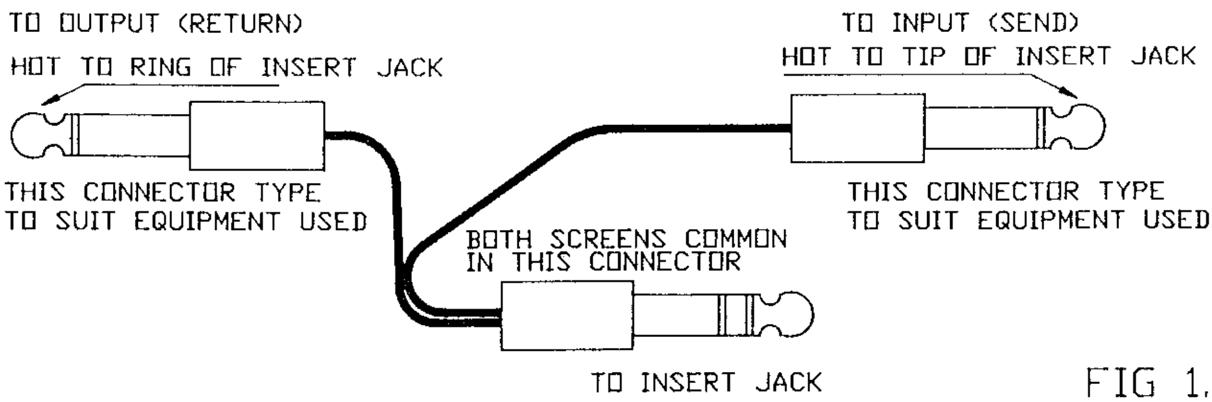


FIG 1.

The insert wiring for the main output is identical.

CONTROL OUTPUT

A 3.5mm jack socket is fitted along side the insert socket. This provides a DC control output. It may be used in conjunction with other modules and equipment to provide switching functions controlled by the channel fader. e.g. Auto-muting of local monitors to prevent howl round etc. ONLY CONNECT APPROVED EQUIPMENT TO THIS SOCKET.

MONO INPUT FRONT PANEL

1 - Gain control with adjacent clip led indicator (see section on gain).

1 - Treble control ±10db @ 10kHz shelving characteristics.
1 - Mid control ±10db @ 1.5Khz bell characteristics.
1 - Bass control ±10db @ 120Hz bell characteristics.
The e.q. range is easily restricted to ±5dB via internal links. (See options section).
1 - Pan control used to position the signal anywhere in the final stereo picture. Centre position equal to left and right. Fully clockwise right hand channel only. Fully anti-clockwise left hand channel only.

1 - Channel identification area.

Use the labels provided with each new mixer to label channel function. Make sure that the area is clean and not greasy for best results. Carefully remove the appropriate label from the backing paper, position on the module and press into place. Transferring the label on the tip of a penknife blade or similar can help considerably.

4 Illuminated push button switches

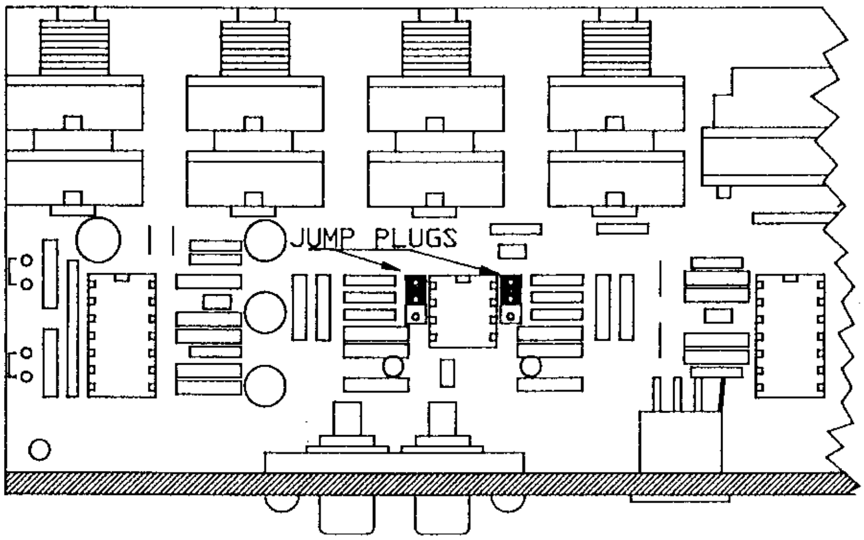
1 - E.Q. OUT. When depressed e.q. section and any outboard equipment connected to the insert socket is by-passed
2 - MUTE. When depressed the channel is turned off.
3 - AUX. When depressed the signal from this channel is mixed with any other source selected. The resulting mix is available at the aux. outputs on the master module. This signal is not cut by the mute switch but it is dependent on the channel fader. Therefore, it is possible to send a channel signal to the aux. output only, if required.
4 - PFL (Pre fade listen) When depressed connects the channel signal to the PFL buss and switches the headphones and meter to monitor the pre-fade signal (irrespective of the fader position). This position is also used to check the setting of the gain control. Any number of channels may be selected and the sum of the channels will be monitored.

FADER

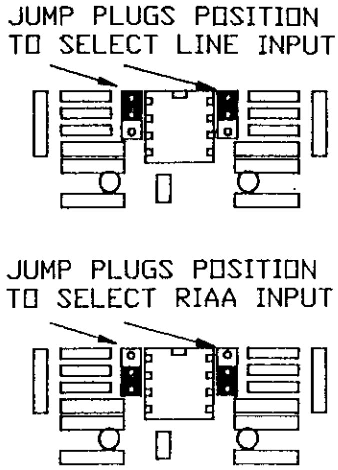
1 - 60mm studio quality linear fader. Is used to fade up and down the volume of the channel. N.B. the fader should not be holding much gain. For normal microphone operation set the fader at No. 7 and adjust the gain for normal volume. Fine control is now available on the fader and the channel may be turned on and off by the mute switch.
IF THE FADER IS BELOW NO. 7 DURING NORMAL OPERATION TURN DOWN THE GAIN CONTROL AND TURN UP THE FADER. THIS WILL GIVE THE BEST PERFORMANCE.

STEREO INPUT MODULE

This module will accept virtually any stereo input signal likely to be required. e.g. Tape or cassette recorder, CD player, video, tuner, mixing desk, etc. Phono cartridge (RIAA) needs to be selected internally. This is a simple operation which will require a pair of snipe-nosed pliers or tweezers. The position of 2 jumper plugs determines whether the RIAA stage is in or out. (See below)



STEREO MODULE BACK



CONNECTIONS (Rear Panel)

2 - Gold plated phono connectors accept left and right input signals.

REMOTE START/STOP

1 - 5 pin din socket provides connection to the remote switch.

Connections PIN 1 Normally open
2 Common
3 Normally closed

OUTPUT MODULE

The output module contains all the electronics required for the various mix and output stages along with the monitoring, switching and VU meter sections. Connections are on the rear of the unit.

FRONT PANEL

Visual monitoring is provided by a stereo 24 segment led bar graph meter. The meter normally reads main output but can be switched to read record output, PFL monitoring signals, and split cue which allows both program and PFL signals to be monitored simultaneously. The headphone monitoring follows the VU meter. A peak hold facility is provided with an adjacent on/off push button switch. This facility makes the viewing of peak levels very easy.

POWER INDICATOR

2 red led indicators confirm the status of the connected power supply unit (PSU). If both of these leds are not illuminated a fault has occurred with the PSU or connecting cable. Disconnect the PSU from the mains supply, check that the interconnecting cable is mated properly and the retaining clips are in place. If the fault persists seek qualified technical assistance.

DO NOT LEAVE A FAULTY PSU CONNECTED. IT MAY CAUSE DAMAGE TO THE MIXER. DISCONNECT IT FROM THE MAINS AND THE MIXER.

CONTROLS

ZONE This control adjusts the level of the zone output. It is a stereo or mono programme output independent of the master fader. A push button on the rear panel selects stereo or mono.

AUX This control adjusts the level of the aux output. It is a stereo or mono output, and is the sum of the various channels switched to aux. A push button on the rear panel selects stereo or mono. Mono input modules feed equally to left and right.

L BAL R This control allows the operator to adjust for slight imbalance in the stereo signal. The normal setting would be centre, only the main output is affected by this control.

PHONES This control adjusts the level of the headphone output available on the adjacent jack socket. (Recommended headphone impedance 50 ohms or greater).

N.B. THIS IS A STEREO OUTPUT AND ONLY A STEREO JACK PLUG CONNECTOR MUST BE USED. PROLONGED USE OF A MONO JACK TO CONNECT HEADPHONES WILL CAUSE PERMANENT DAMAGE AND INVALIDATE THE WARRANTY

STEREO INPUT FRONT PANEL

1 - Gain control with adjacent clip led indicator (see section on gain).
1 - Treble control ±10dB @ 10kHz shelving characteristics.
1 - Mid control ±10dB @ 1Khz bell characteristics.
1 - Bass control ±10dB @ 60Hz shelving characteristics.
The e.q. range is easily restricted to ±5dB via internal links. (See options section).
1 - Rotary 3 position switch selects crossfader A ¦ B. The centre position selects both A & B, therefore the channel will be fed to the output no matter where the crossfader is set. Any number of stereo channels may be assigned to any position on the crossfader. This makes this section very flexible, complex stereo mixes and fades may be performed easily.

1 - Channel identification area.
This is identical to the mono input module and may be labelled in the same manner.

4 Illuminated push button switches
1 - REMOTE switch connects to the remote socket for controlling ancillary equipment. The switch is supplied as a latching push button but may be changed to momentary action (See options section).
2 - MUTE When depressed the channel is turned off.
3 - AUX When depressed the signal from this channel is mixed with any other source selected. The resulting mix is available at the aux outputs on the master module. This signal is not cut by the mute switch but it is dependent on the channel fader. Therefore, it is possible to send a channel signal to the aux output only, if required.
4 - PFL (Pre fade listen)
When depressed connects the channel signal to the PFL buss and switches the headphones and meter to monitor the pre-fade signal (irrespective of the fader position). This position is also used to check the setting of the gain control. Any number of channels may be selected and the sum of the channels will be monitored.

FADER

1 - 60mm studio quality linear fader. Is used to fade up and down the volume of the channel. N.B. THE FADER SHOULD NOT BE HOLDING GAIN. For normal production operation set the fader at No. 9 and adjust the gain for normal volume. The fader should be used to fade in the signal at the required time taking the fader to the top. This lets you concentrate on your timing, confident that the volume will always be correct. If you want to operate so that you have some gain in hand, (for instance, during a mixing session) operate with the fader set at No. 7. Remember always to keep the gain control as low as possible to achieve the desired results. (See section on gain).

MASTER FADER This is a 60mm linear fader and controls the level of the main stereo output.

CROSSFADE This is a 45mm linear fader which fades smoothly between stereo channels selected to X-fade A and B.

PUSH BUTTON SWITCHES

4 Illuminated push button switches located centrally perform the following functions:-

1 - **MUSIC DIM** When depressed the signal from the stereo channels is attenuated by approx 20dB. This function may also be remotely activated by shorting the 3.5mm jack on the PSU4. The switch will also be illuminated if remotely activated. (The switch may be modified to momentary action - see options).

2 - **SPLIT** Use to assist in cueing and synchronisation of programme material. When depressed the headphone monitoring and VU meter function is split to provide mono PFL signals on one side and mono program on the other.
The program source will be main output or record output depending on the position of the MON REC O/P switch. The PFL source is whatever is selected to PFL.

3 - **MONO** When depressed the main output will be switched to mono.

4 - **MON REC O/P** (Monitor record output). When depressed swaps the monitoring source from main output (post main fader) to rec output (pre main fader). This function is also useful to compare levels as PFL and record output should be the same when the channel fader is used at the full up position.

OUTPUT MODULE REAR PANEL

Main outputs (balanced)	3 pin XLR	PIN 1 Screen 2 Phase 3 Non Phase

For unbalanced operation strap pins 1 & 3

Zone outputs (unbalanced) } Aux outputs (unbalanced) }	3 pin XLR	PIN 1 Screen 3 screen 2 Hot

Record outputs (stereo) 1/4" Mono Jack
This provides programme output which is not affected by the main fader.

Music Mix outputs (stereo) 1/4" Mono Jack
This provides a mixed output of stereo channels only.

Main output inserts (send/return) 1/4" 3 Pole Jack
(The insert sockets are provided to allow extra outboard equipment to be inserted in the main output chain, i.e. compressor, equaliser etc.). (See Fig. 1, Page 2)

N.B. Zone and Aux outputs are independently switchable to mono or stereo by an adjacent push button switch.

Sub Bass output - mono 20Hz - 80Hz.
Post master fader. Adjacent multi-turn preset controls level.

Output socket 1/4" 2 pole jack.

TRIG (Trigger output) - mono.
Pre-master fader. Adjacent multi-turn preset controls level.
Transformer isolated to ensure safe connection to lighting systems.

Output socket 1/4" 2 pole jack - fully floating. (For audio use see options section).

Sub-sonic filter.
Operational on all mixed program outputs to prevent sub-sonic damage to loudspeakers. -3dB at 20Hz; slope 18dB per octave.

PSU connector to power supply 14 pin IEEE

ALWAYS ENSURE THAT THE RETAINING CLIPS ON THIS CONNECTOR ARE USED.

CONNECTIONS AND GOOD WIRING PRACTICE

The installation of professional audio systems should be left to experienced engineers wherever possible. The interconnection of audio systems can be fairly complex and is obviously well outside the scope of this handbook. However, we have included a few basic points for anyone who is new to audio systems.

Good wiring practice should be observed when connecting any audio equipment. Good quality connectors and screened cable should be used for all audio connections. Twin screened cable should be used for all balanced lines particularly microphone connections. Always ensure cable clamps are fully tightened and gripping the outer sheath. Good strain relief and mechanically sound connections will increase reliability at virtually no extra cost.

GROUND LOOPS In our experience this is the most common problem encountered when connecting together different items of audio equipment. It is also the most common cause of hum (50Hz noise) on a system and is caused by incorrect system grounding.
When unbalanced connections (i.e. 2 connections, single screened cable, etc) are used to connect several items of audio equipment together, the signal common connection is the screen and this will be connected to mains earth at some point.
If several items of equipment have their signal common connected to mains earth this will form a loop (hence ground loop). Current will flow in this loop and appear in the form of hum (50Hz mains frequency) added to the audio signal. The problem is aggravated if the equipment is located some distance away as the loop is larger.
It is possible to have several ground loops within a system.
The solution is to connect the system to mains ground <u>ONLY ONCE</u>.
This is usually done at the mixer.
You will need to investigate the various items of equipment which you are using and isolate their signal common from mains earth.
Many manufacturers fit a ground lift switch for this purpose. On some equipment this is in the form of a removable link.
Unfortunately with some equipment you have to get inside to identify where the connection is and remove it.

YOU MUST NOT DISCONNECT THE MAINS EARTH WIRE FROM THE MAINS PLUG OF ANY EQUIPMENT. THIS IS FITTED FOR SAFETY REASONS AND MUST BE CONNECTED TO ENSURE THAT THE CASE IS EARTHED.

The power supply unit PSU4, has an internal link connection to mains earth which may be removed if this is the only way to solve the problem. Disconnect the mains supply and remove the case cover. Trace the earth wire from the mains input connector to the circuit board. Locate and cut the un-insulated wire link directly to the side of this wire. Refit the cover.
Often fitting an audio transformer to isolate different parts of the system may be the solution.

BALANCED LINES Balanced line connections are less prone to interference pick-up and are advisable for low level signal connections, i.e. microphones, and long cable runs. Balanced connections and floating outputs (transformer isolated) are usually connected using twin screened cable (3 terminal connectors, XLRs, etc). It is usual for balanced output circuits to be connected to balanced input circuits but this is not always the case. Often transformer balanced outputs and balanced inputs are useful to isolate ground loops. (But this may not be possible with electronically balanced output stages).

NOTE ABOUT THE USE OF GAIN

In an audio mixer different levels from various items of equipment need to be amplified to a common level so that they can be added or mixed together. These levels may differ considerably depending on the equipment. The signal from a microphone for example may be 1000 times smaller than that from a CD player.

For a mixer to be as flexible as possible and to accept signals from a variety of equipment it is desirable for the first or early stage in an input circuit to be a variable gain amplifier. The gain of this amplifier is set by the gain control.

All PM-90 input modules have variable gain. When amplifier gain is introduced in a circuit, noise is also introduced (this is a fact, you cannot have one without the other). The PM-90 has been designed to keep this noise as low as possible using the latest technology. For the best performance set the gain control as low as possible to achieve the desired output. Gain introduced into the system and then held on the channel fader is a waste of performance. Too much gain could result in overloading the first stage causing distortion and clipping.

A red clip indicator has been included next to the input gain control on all PM-90 input modules. This indicator flashes when the signal level is close to clipping. It monitors the signal at various points in the input module signal chain, therefore gain introduced by the equalisation or tone controls is also considered. IF THE CLIP INDICATOR ILLUMINATES DURING USE TURN DOWN THE GAIN CONTROL.

REMOVING AND RE-FITTING MODULES

REMOVING MODULES. To remove an input module from the chassis first remove 2 securing screws top and bottom. Carefully ease the module out of the chassis and remove the edge connector. (This may be easier through the space left in the rear when the module is eased forward).
To remove the output module, first remove the two input modules to the immediate left to provide some slack on the ribbon cable. Remove the 4 securing screws, then remove the output module from the chassis and disconnect the 2 connectors. The last connector is held by latches which will eject the connector when moved outwards. Reassemble in the reverse order. ENSURE THAT THE CONNECTORS ARE MATED CORRECTLY.

The modules have clearance of approx 0.25mm. This is deliberate to make the modules easy to remove and refit. Use a strip of thin card (postcard) to set the spacing between the modules when reassembling. Do not push all the modules tightly together as this will make them difficult to remove and leave an unsightly space. Remember 0.25mm is only 10 thousandths of an inch. This is the normal spacing clearance. But in an 8 channel chassis, if you push the modules tightly together the resulting space will be 2.5mm which is 1/10" which looks unsightly.

SPECIFICATIONS

Frequency response (measured with e.q. set to flattest position)

AUX outputs and Music only outputs	20Hz - 20kHz ±0.5dB
Program outputs	50hz - 20kHz ±0.5dB (Sub-sonic filter -3dB at 20Hz; slope 18dB per octave)

Distortion (THD) freq 1kHz Any input to any audio output	O/P @ +20dBV better than 0.01%

Output capability Excl. Trigger O/P	+20dBV into 600R load

Noise measured 20Hz-20kHz	
Mono module	
Microphone i/p (standard) Ref 150R	Equiv. input noise < -125dBU
Microphone i/p (special)	" " " < -128dBU
Stereo module	
Line input (unity gain)	" " " < -90dBU
RIAA stage	
Ref 5mV at 1kHz Input shorted	80dBV 'A' weighted

Gain	
Mono module	
Microphone max	70dB
Line max	30dB
Stereo module line i/p (max)	20dB

Maximum input level	
Mono module	
Microphone input (min gain)	+2dBV
Line input	+25dBV
Stereo module	
Line input min gain	+20dBV
RIAA input	-8dBV 400mV

Input impedances	
Mono module	
Microphone input	>2K ohms active balanced
line input	>10K ohms active balanced
Stereo module	
Line input or RIAA	47K ohms unbalanced

All balanced inputs and outputs are self-compensating. Either terminal may be grounded for unbalanced operation.
N.B. when wiring balanced circuits for unbalanced operation left and right stereo channels should be identical to maintain phase.

OPTIONS

INSTALLERS MAY WISH TO REMOVE THE OPTIONS PAGES TO PREVENT THIS INFORMATION BEING USED BY NON-TECHNICAL OPERATORS.

The PM-90 has been designed to be as flexible in operation as possible. Several optional facilities have been included. Most of these require fitting or removing links on the printed circuit boards.

STEREO MODULE

REDUCING E.Q. RANGE

Remove the module from the chassis. Identify 4 raised links on the PCB - 2 are between and behind the gain and treble pots, 2 are directly behind the remote socket. Cut all 4 links. This will halve the e.q. range to ±5dB. (Fig. 2)

INCREASING GAIN

The normal gain range is ±10dB. This may be modified to the range Unity to +20dB. Remove the module from the chassis. Identify 2 pairs of pads on the circuit board at the top of the module farthest from the fader. Carefully solder 2 links as per Fig. 2.

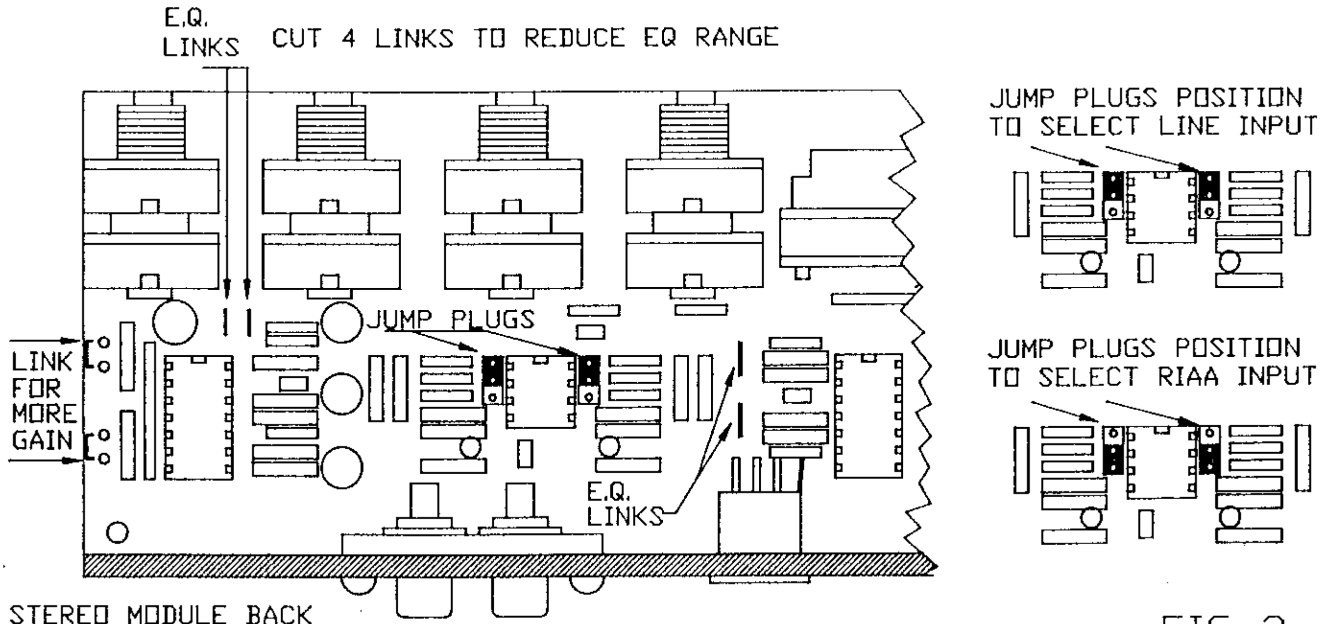


FIG 2.

MONO MODULE

REDUCING E.Q. RANGE

Remove the module from the chassis. Identify 4 pads behind bass pot. Carefully solder 2 links as per Fig. 3.

MICROPHONE PHANTOM POWER

To modify the module for a microphone which requires phantom power remove the module from the chassis. Identify 2 pads on the circuit board behind the microphone socket. Carefully solder link as per Fig. 3.

LEVEL CONTROL MODULE

A level control module is available to prevent systems from being over-driven or to control the maximum noise level in a venue. (This is often a legal requirement to conform to the environmental noise legislation).
If a level control module is fitted it is there for a good reason. Do not try to by-pass this unit or remove it - once fitted the mixer will not work without it. The internal presets within the unit are there to null the distortion of the circuit and can only be set with the aid of test equipment. Any tampering of this unit will degrade the superb audio performance of this unit.

OPERATION

The operation of the unit is very simple. If the operating level of the mixer is kept below the threshold, which usually means keeping below the red section of the output VU meter, everything is fine and the level control module has no effect. If this level is exceeded the module will automatically reduce the output level of the mixer. The level is reduced in discreet steps indicated by the bar graph meter on the module. The more one tries to increase the level the more the module will turn it down. It will, in fact, reduce the level by more than you are trying to increase it. This situation will continue until the mixer runs out of headroom, which will sound dreadful, but it will be quiet. Best and loudest results will be achieved by not having any of the leds lit on this module.

The action of this module is slow acting so as not to impair the dynamic range, i.e. it is not a compressor.

Dimensions (Exc knobs & connectors)

Mixer

Width		Height	Depth
4 channel	330mm (13")	267mm (10½")	76mm (3")
8 channel	483mm (19")	" "	" "
12 channel	635mm (25")	" "	" "

External PSU - PSU4

Width	Height	Depth
175mm (6.9")	75mm (3")	215mm (8.5")

CHASSIS

The chassis is contructed from custom aluminium extrusions and is available in 4, 8 and 12 input module configurations.
Finish is black anodised aluminium.
Module metalwork is also constructed of custom aluminium extrusions and is finished in black anodised aluminium with integral etched notation which will not rub off in use.

All Formula Sound products are designed and manufactured in our own factory which enables us to maintain strict quality control at every stage of manufacture. This attention to detail has helped to win us severel awards and has earned us a reputation throughout the world for the high quality and reliability of our products.

Formula Sound reserve the right to alter specifications at any time without notice.

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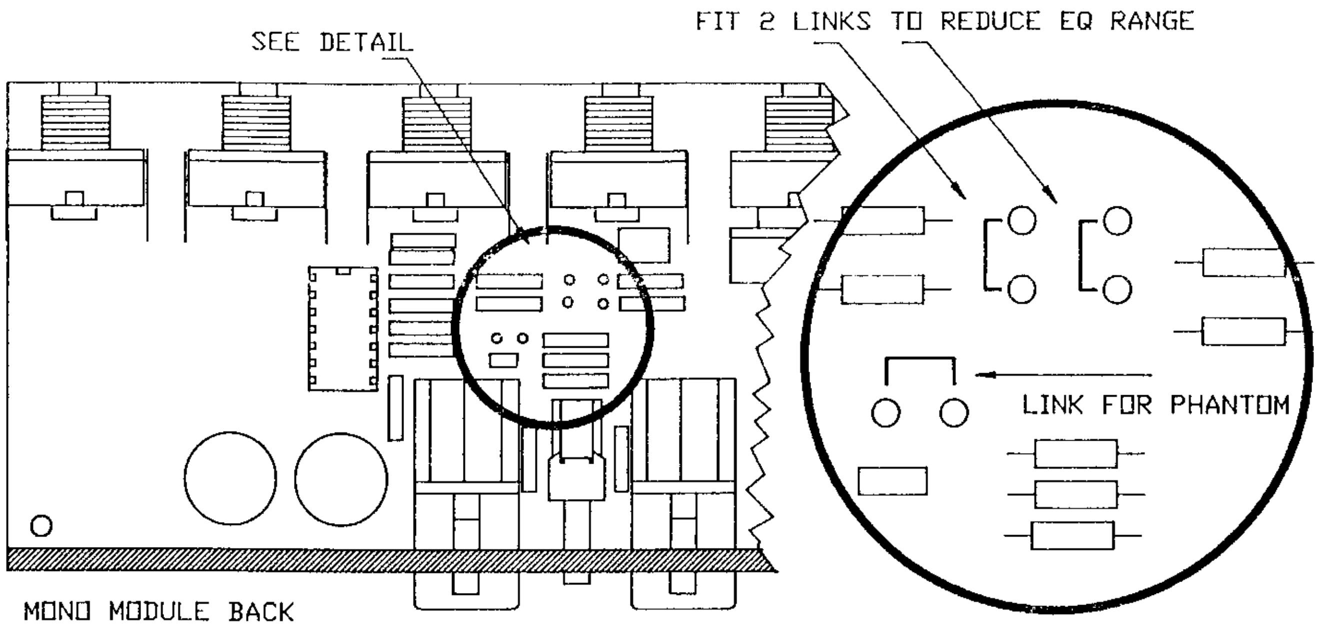

FIG 3.

TRIG (TRIGGER OUTPUT)
If this mono output is required for audio use it should be noted that the performance of this output is determined only by the size of the internal isolating transformer. Output levels below 0dBU should be acceptable. For high level signal operation it is recommended that the internal isolating transformer is removed. Contact Formula Sound Technical Dept for more details.

PSU CONNECTOR
The PSU connector (14 pin IEEE) also carries the connection to the music dim facility accessible via 3.5mm jack socket on the power supply. Connect jack tip to body or ground, this activates music dim.

This connector also has a connection to the mono PFL signal. This may be accessed by carefully connecting miniature screened cable to pin 9 (hot), screen pin 13. Only attempt this connection if you have suitable experience. Shorts on this connector may cause serious damage and void warranty.

CONVERTING PUSH SWITCHES
It is possible to modify the push button switches from latching action to momentary action. This may be required on some forms of remote start switching. This modification removes the latching mechanism. ONCE MODIFIED THE SWITCH CANNOT BE CHANGED BACK without fitting a new switch so be sure that you want this type of switch before proceeding. With the module on a flat surface remove the fader (2 screws) and hinge out of the way. Identify the switch to be modified. Using a sharp knife or scalpel remove the raised rectangular section of plastic on top of the switch. This is easily done by cutting along one side and hinging up. Remove the flat spring and the wire latch with a pair of tweezers or snipe-nose pliers. Refit the fader.

INSTALLATION SECURITY
A security kit is available to effectively cover the rear of the mixer with the intention of preventing unauthorised persons from connecting or tampering with input and output connections. Contact Formula Sound for details.

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CE

E. U. CERTIFICATE OF CONFORMITY

We declare that the products listed conform to the following directives and standards

89/336/EEC amended by 92/31/EEC and 93/68/EEC

BS EN 50082-1 BS EN 50081-1

PRODUCT TYPE
PM-90 MODULAR MIXING SYSTEM

The CE mark was first applied in 1995

Signed ....

R. A. Cockell Managing Director

ATTENTION

The attention of the specifier, purchaser, installer, or user is drawn to the fact that good wiring practice must be observed when connecting the above equipment. Good quality connectors and screened cables must be used for all audio connections. Twin screened cables should be used for all balanced lines.

THE EQUIPMENT MUST BE EARTHED

CONSULT THE USERS MANUAL FOR TECHNICAL DETAILS