FORMULA SOUND

PM - 100

MODULAR PRODUCTION MIXER

USERS MANUAL

PM - 100 Introduction

The Formula Sound PM-100 mixer is essentially the successor to our award winning PM-80 and PM-90 mixers.

We believe that the experience we have gained over the past 21 years of producing the PM-80 and PM-90 has produced a mixer that is equal to the challenges of the new millennium. We have incorporated new features and innovations that have been in our file of wish lists from operators around the world. We have tried to provide a tool that can be individual to the operator, We can provide advice on how to achieve the best performance out of this equipment but when it comes to mixing it becomes personal taste so we have provided the tools to do the job with choices where possible. We believe that the equipment should not restrict the creative talent but help it.

General Description

The PM-100 is a compact modular production mixing system which combines exemplary technical performance with the highest standards of mechanical engineering.
The system is totally modular each module being complete in itself. The connections to the modules are via a flexible 34 way ribbon cable and gold plated computer grade connectors.
The chassis contains no electronics therefore upgrading to a larger chassis size is quite straight-forward.

There are only three types of module, a universal input channel module, a blank module to fill unused space in the chassis and the output module containing all the mixing, monitoring, and output electronics.

The mixer is factory assembled with the output module towards the right hand end of the chassis and the input modules to the left. But in fact modules may be positioned in any format required. Specific layouts may be ordered or engineers can easily change layouts to suit individual requirements. Blank modules where fitted may be easily replaced by input modules at a later date if required.

The chassis is available in 3 standard sizes, 4 channel, 8 channel (19" rack mount) and 12 channel. Larger sizes can be offered to special order.

Power Supply

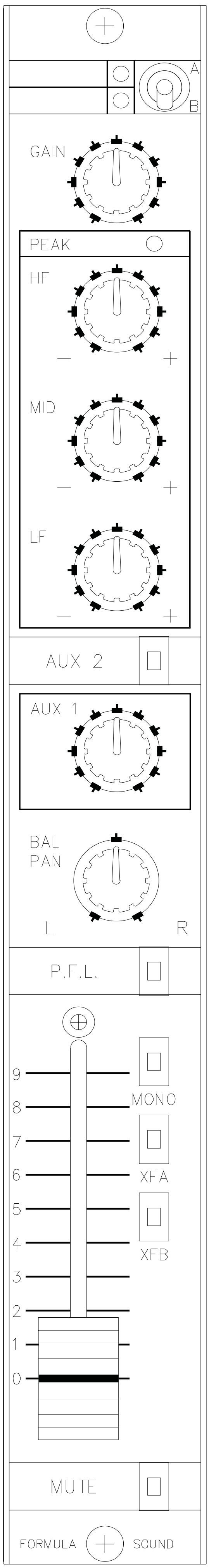
The mixer requires an external power supply unit providing +/-17.5V for the audio electronics and 12V for the working illumination socket.

A suitable power supply unit, the PSU5, is available. This unit is designed to be surface mounted remotely from the mixer and features slow rise time characteristics to minimise turn on thrumps.

The unit is internally switchable to operate from 220-240V AC or 110-120V AC. Check that the voltage selector switch is correctly set for the mains voltage available before connecting power.

Damage may result if the unit is connected to the wrong supply voltage.

It is ***important*** from a performance and safety aspect that the power supply is connected to a reliable and ***suitable earth***. Remember that this earth connection may also be providing the earth for input sources connected to the mixer, turntables, microphones etc. For safe and reliable operation adequate ventilation must be provided. ***Do not cover the power supply***.
Mount the power supply in a suitable location bearing in mind that the captive connecting lead length is 1.5 meters. Connect the mixer to the power supply ensuring that the connector is fully mated and the securing screws fastened. (Do not over tighten the securing screws).



PM - 100 UNIVERSAL INPUT MODULE

Input Sources. The design of this module allows almost any input source to be used and further allows the operator to switch between any two input sources.
This is achieved by providing a balanced low impedance low noise microphone pre amplifier; a stereo RIAA equalised pre amplifier which also may be configured as a normal flat response line level input; and a normal line level input stage.

Input Connectors Rear panel mounted 1 - XLR Mic input socket, 1 - 3 pole jack insert socket (provides connection for external equipment into the microphone channel).
Two pairs of gold plated RCA phono sockets are provided for the stereo inputs.

Input **A** may be the RIAA input or a normal line input as selected by 2 internal jumpers. The **B** input may be a microphone input or a stereo line input as selected by a further internal jumper. Rear panel gain trims are provided to set the maximum gain available on any input.

INPUT SELECTION A front panel mounted toggle switch allows rapid selection of the **A** or **B** input. Red and Green led's provide clear visual indication of the input selected.
Adjacent areas are provided to label the input sources.

GAIN A control allows the operator to compensate for variations in programme material. (See later section on gain)

PEAK A red led indicates when the peak signal level in the module is just below clipping.

Three band equalisation

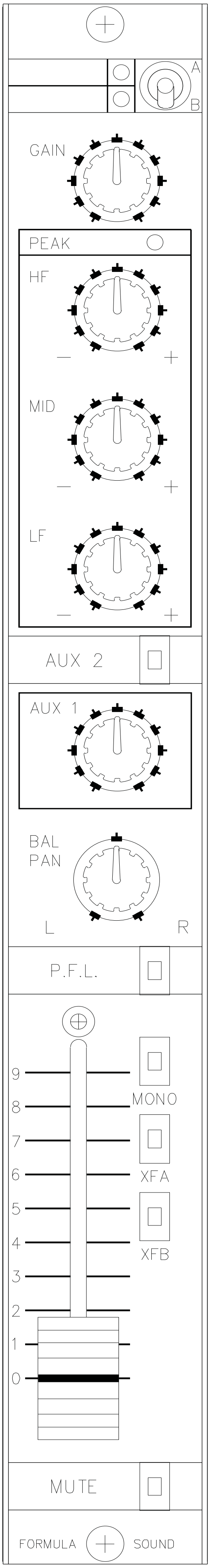
HF (High Frequency or Treble)
This control provides +6dB boost -20dB cut @ 10kHz with a shelving response.

MID (Middle Frequencies)
This control provides +6dB boost -20dB cut @ 1kHz with a bell response.

LF (Low Frequency or Bass)
This control provides +6dB boost -20dB cut @ 100Hz with a bell response.
This response has been chosen so that sub sonic frequencies are not boosted by this control.
All E.Q. controls have a centre detent at the (E.Q flat) centre position.
The E.Q. function may be disabled by the setting of internal jumpers (see diagram).

AUX 2 Push button with led illumination.
When depressed the signal in this module is mixed with any other source selected. The resulting mix is available at the Aux 2 outputs on the master module. Internal jumpers allow the source to be selected pre of post (before or after) the channel fader. (See output section for more options on Aux2)

AUX 1 has a similar function to Aux2. A control replaces the switch so a separate level mix of channels is available at the Aux1 outputs on the master module. Internal jumpers are also provided for pre post selection.



PM - 100 UNIVERSAL INPUT MODULE continued.

Bal - Pan (Stereo balance & Pan)
With a stereo source selected this control provides adjustment to balance stereo signals.
With a microphone source selected or the mono switch depressed this control allows the source to be positioned anywhere in the stereo image.
A centre detent is provided at the control centre position.

PFL (Pre Fade Listen)
Push button with red led illumination.
When depressed the signal in this module is connected to the PFL Mix Buss. Any number of channels may be selected simultaneously, the sum of which will be available in the master module for monitoring. An indicator led below the Vu meters also illuminates if any PFL switch is depressed. (See monitoring section on output module for more information)

MONO Push button with red led illumination.
When depressed this switch simply links left and right signals and allows stereo signals to operate in mono.

XFA (Crossfade A select) Push button with red led illumination.
Depressing the switch routes the channel signal via the VCA controlled crossfader channel A. XFA will take priority if both buttons are depressed.

XFB (Crossfade B select) Push button with green led illumination.
Depressing the switch routes the channel signal via the VCA controlled crossfader channel B.

Red and green led switch illumination along with similar illumination on the crossfader gives good visual indication of routing status.

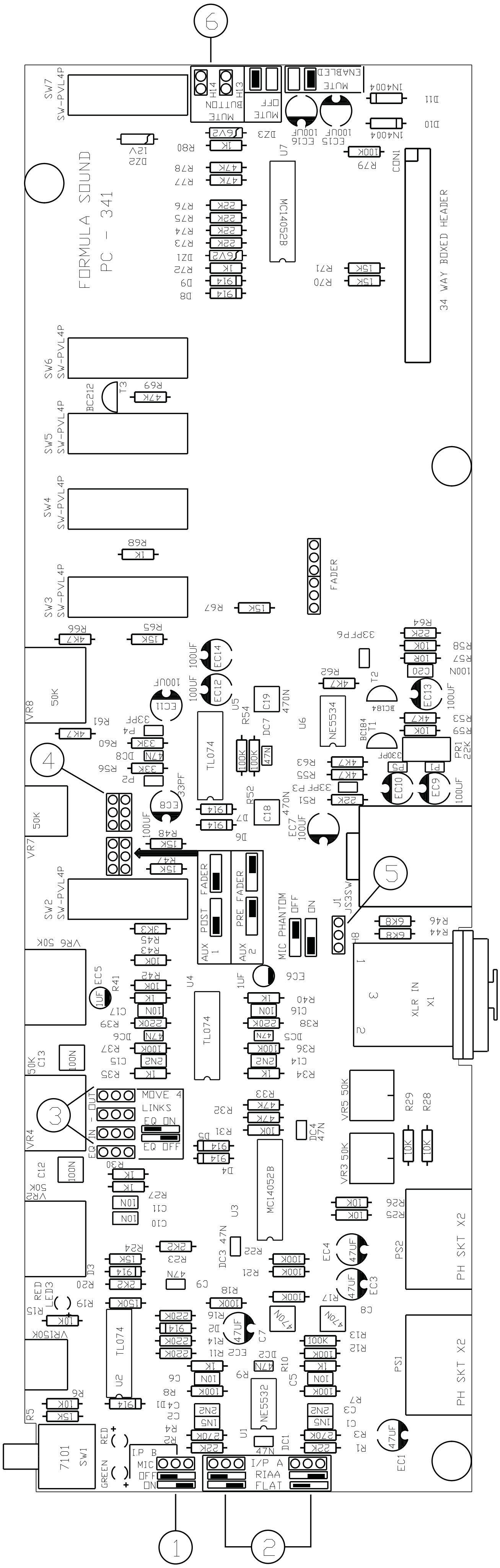
If neither of the crossfade assign switches are depressed and a stereo source is selected the signal will be routed via the stereo buss to the output section bypassing the crossfade circuitry.

If a microphone source is selected the crossfade A&B assign switches will not function and will not illuminate if depressed. The signal will be routed to the output section via the microphone mix buss.

Fader
A 60mm studio quality linear fader is provided. This should be used to fade up and down the volume of the channel. For normal production operation the fader should be set at the top (No.9) and the gain control adjusted 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 the timing confident that the volume will be correct. If you want to operate with some gain in hand (e.g. mic mixing) operate with the fader set at No.7.
Always operate with the gain set as low as possible to achieve the desired results. (See the section on gain)

MUTE
Push button with red led illumination.
When depressed signal feeds to the programme busses are muted. AUX1 and AUX2 are not affected. An internal jumper allows this switch to be disabled.

(Factory option - the mute switch may be rewired to provide a low voltage remote start switch wired to a 3.5mm jack socket on the rear panel.)



PM - 100 UNIVERSAL INPUT MODULE

Internal Settings

Jumper links are used to select and change functions.
1-5 uses 3 pin headers and the link should join the centre pin to one of the end pins.
6. Two pairs of pins - the link joins one pair of pins. The various link positions are also shown on the PCB.
If you are changing link positions ensure they are seated correctly and all necessary links have been moved

1. ***1 Jumper link.*** The position selects input B to microphone or stereo line input (in the microphone position the Xfade select switches are disabled.)

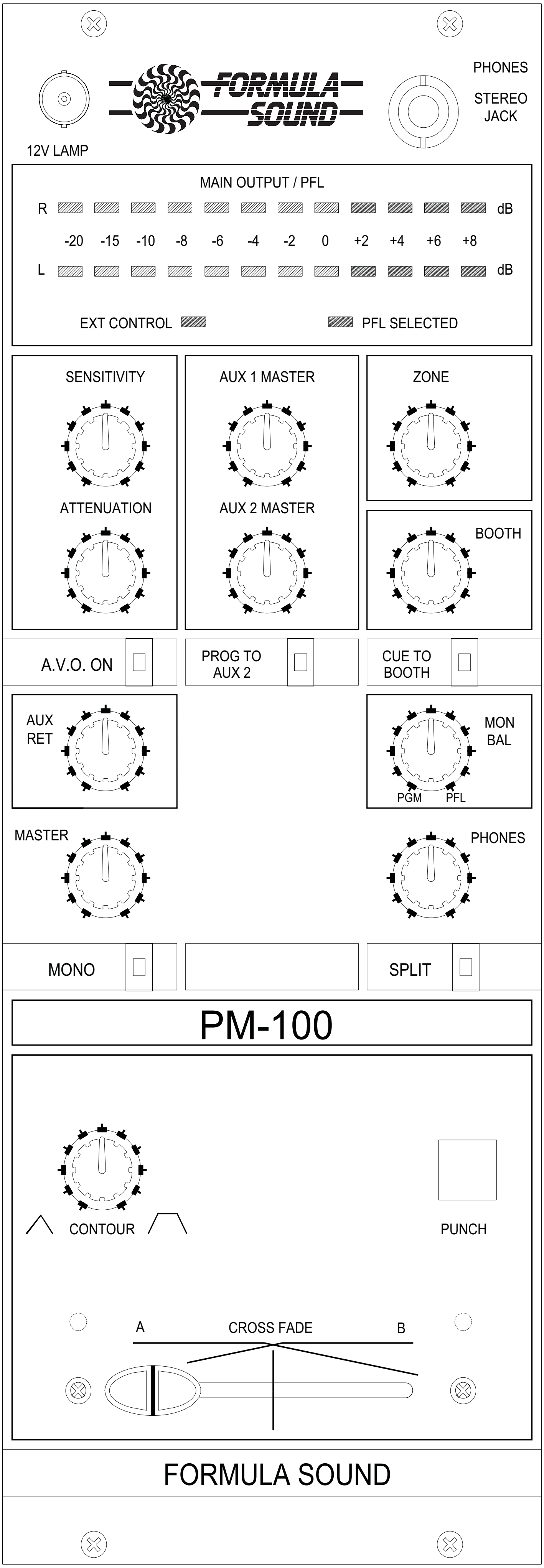
2. ***2 Jumper links***. The positions select input A to be a Magnetic phono cartridge equalised to RIAA or a normal flat response line input. Move both links.

3. ***4 Jumper links***. The positions select equalisation on or off (i.e. tone controls disabled). It is vital that all four jumpers are moved.

4. ***2 pair of jumper links*** select Aux1 and Aux2 to be pre or post fader. (i.e. if the selected aux function is affected by the channel fader.) Move 2 links for AUX1 and 2 links for Aux2.
E.G. an auxiliary set up for use as an echo send on a microphone channel would normally be set to post, so if the fader is closed no signal would be sent to the echo system.
If the auxiliary function was in use as a foldback or monitoring feed it would normally be set to pre fader.
Aux sends can be used for many tasks.

5. ***1 Jumper link*** selects Phantom power on or off for microphone channels.

6. ***1 Jumper link*** The position of this link determines if the channel mute button is enabled or disabled.



PM-100 Master Module

The master module contains all the electronics required for the various summing amplifiers, output stages, VU meters, crossfade control circuits, etc. The majority of connections are on the rear of the unit.

Monitoring

The monitoring section of the PM-100 is very comprehensive. Visual monitoring is provided by a stereo 24 segment led bar graph meter.
Two powerful stereo headphone amplifiers are incorporated designed to drive headphones of 32 ohms or greater. Connections are via 3 pole jack sockets. The main phones output is located at the top right of the front panel, the second output is on the rear panel.
The meter and the main phones output are both connected to the same signal so what you see is what you hear.

Mon Bal (monitor balance)

It is important that the function of this control is fully understood.

The output from this control is fed to the monitoring section (the led bargraph meter and via the PHONES volume control to the main phones output).
With the control turned fully clockwise towards ***PFL***, the meter will read the PFL signal of any channel that has its PFL button depressed, or the sum of channels if more than one is selected. As a further visual aid a red led beneath the meter also illuminates if any PFL button is depressed. If no PFL buttons are depressed the meter reads the main output level which is dependent on the MASTER control setting.
Turning the monitor balance control counter clockwise towards ***PGM*** feeds the programme signal (pre master) to the monitoring section. Therefore the operator can quickly pan or mix between the PFL and program.
It also facilitates the monitoring of a recording being made with the master turned down.
A SPLIT button is also provided (which is the preference of some operators). This provides a mono PFL signal on one side of the monitor system and a mono program signal on the other side. The Mon Bal sets the balance of the two signals in this situation. Practice and use will determine the best mode of operation.

PM-100 Master Module continued

A.V.O. (Automatic Voice Over)
This function is enabled by the illuminated push button A.V.O. ON. When turned on any microphone signal present will dim the music signals to allow talk over. The amount the music is dimmed is determined by the attenuation control. The adjacent sensitivity control sets the triggering level of the AVO system.

AUX RET (Auxiliary Return)
This is a basic stereo input configured to accept balanced or unbalanced input signals via 2 X 1/4" 3 pole jack sockets. (For unbalanced operation 2 pole jack plugs may be used). For mono operation feed equally into both inputs. The input impedance is >20K ohms balanced and 10K ohms unbalanced. This input is intended for simple no frills connection to the stereo buss for effects return etc. If more facilities are required it is normal to use an input channel.

MASTER - This control sets the output level of the main balanced output available via 3 pole XLR connectors on the rear panel. This output is monitored by the main phones and VU meter when the MON BAL control is set to PFL and no PFL is selected. The VU meter is factory calibrated 0dB = 0dBu (775mV)

MONO - An illuminated push button switch located below the master control allows the master output to be operated in mono mode. This switch only affects the main output.

ZONE -The zone output is almost identical to the main output in all respects except that the signal is not available to the monitoring section and no insert sockets are provided. The output may also be used in mono or stereo as set by a rear mounted push switch located between the XLR sockets.

AUX 1 MASTER - This master control adjusts the output level of the signals derived from the Aux1 sends on the input modules. The outputs are via 2 1/4" jack sockets a mono signal may be obtained by using the left output socket only.

AUX 2 MASTER - This master control adjusts the output level of the signals derived from the Aux2 sends on the input modules. The outputs are via 2 1/4" jack sockets a mono signal may be obtained by using the left output socket only.

AUX 2 OPTIONS
An illuminated push button switch located below the AUX 2 Master control **(PROG TO AUX 2**) routes the programme mix to the AUX 2 output. This is provided so that the AUX 2 output can be used as yet another zone output should this be required.
Alternatively if the second phones output is set (via internal jumpers located on PC-344) to listen to the AUX 2 output a second monitoring system will then be available which is totally independent of the main phones output. This would provide for a second operator to have independent monitoring facilities. In this mode AUX 2 jumper links on the channel modules would normally be set to pre fade. The AUX 2 master control would set the phones level.

BOOTH - This control adjusts the level of the booth output which is a mix of music channels only. The outputs are via 2 1/4" jack sockets. A mono signal may be obtained by using the left output socket only.

CUE TO BOOTH - An illuminated push button switch located below the booth control. When depressed this switch routes the mains phones signal to the booth output, so therefore all the phones monitoring facilities become available on the booth outputs. Operators should be aware that in this mode it is possible to have microphone signals present on the booth outputs which could be the cause of acoustic feedback.

PM-100 Master Module continued

CROSSFADE
The XF-A and XF-B busses are fed to the crossfade circuitry. The crossfader is a 45mm fader which can be used to smoothly fade between the XF-A and XF-B busses.
The crossfade system uses studio quality V.C.A.'s (voltage controlled amplifiers) to Improve crossfader life and also ensure good attenuation at the ends of travel.
Red and green leds positioned at the fader ends correspond to the XFA and XFB routing switch illumination colours which gives good visual routing status.

CONTOUR
A contour control is provided which allows the operator to adjust the crossfader characteristics to suit personal taste, from a gentle fade between A and B to a more aggressive fade towards the ends of the fader.

PUNCH
A fast momentary action push button is fitted to provide rapid switching effects. The action of this switch is to turn on both XF-A and XF-B busses irrespective of the fader position. Depending on how channels are routed to crossfader this can provide several effects for the creative D.J.

INSERTS
Insert sockets in the form of 3 pole 1/4" jacks are provided for the connection of external signal processing equipment. Located on the rear panel the connections are:-
Tip = send (mixer output)
Ring = return (mixer input)
Body = common (ground)
These sockets have switching contacts so that if a plug is not inserted the signal path is maintained.

Insert sockets are provided for the following functions:-
XFA left and right for the connection of filters etc. into the XFA signal chain.
XFB left and right for the connection of filters etc. into the XFB signal chain.
MIC left and right for the connection of signal processing equipment, compressors, limiters etc. into the microphone signal chain.
Master Insert left and right for the connection of signal processing equipment into the main output chain.
Individual microphone inserts are also provided on the input channel modules.

RECORD OUTPUTS 2 pairs of gold plated phono sockets are provided for the connection of stereo recording equipment. One pair provides the complete programme mix; the other pair provides the mix of music sources only. AVO does not affect this output. All record outputs are unaffected by the master controls.

SUB BASS OUTPUT A mono output via a 1/4" jack is provided for driving a sub bass loudspeaker system to enhance low frequency performance. The filter allows frequencies below 70Hz to pass through. An internal jumper located on PC-344 allows higher bass output if required.

LIGHTING TRIGGER A mono fully floating transformer isolated trigger output is available on a 1/4" jack to provide safe connection to lighting equipment.

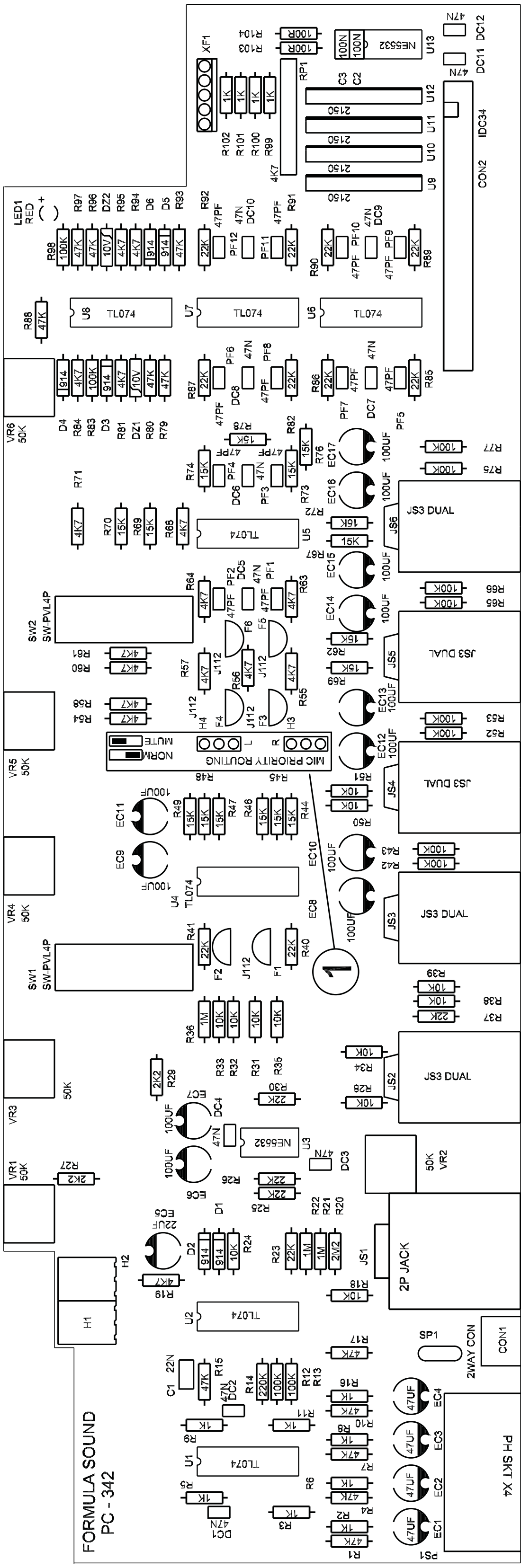
PRIORITY
A priority input with adjacent level trim control and control socket are provided to help comply with fire regulations. The fire alarm panel needs to provide a fully floating pair of contacts to join the two pins on the control connector.
Internal jumpers located on PC-342 provide the option to mute all signals or leave mic signals active and intoduce the priority signal.
The priority input may not be required if existing microphone inputs are to be used for evacuation. This will depend on local fire regulations.

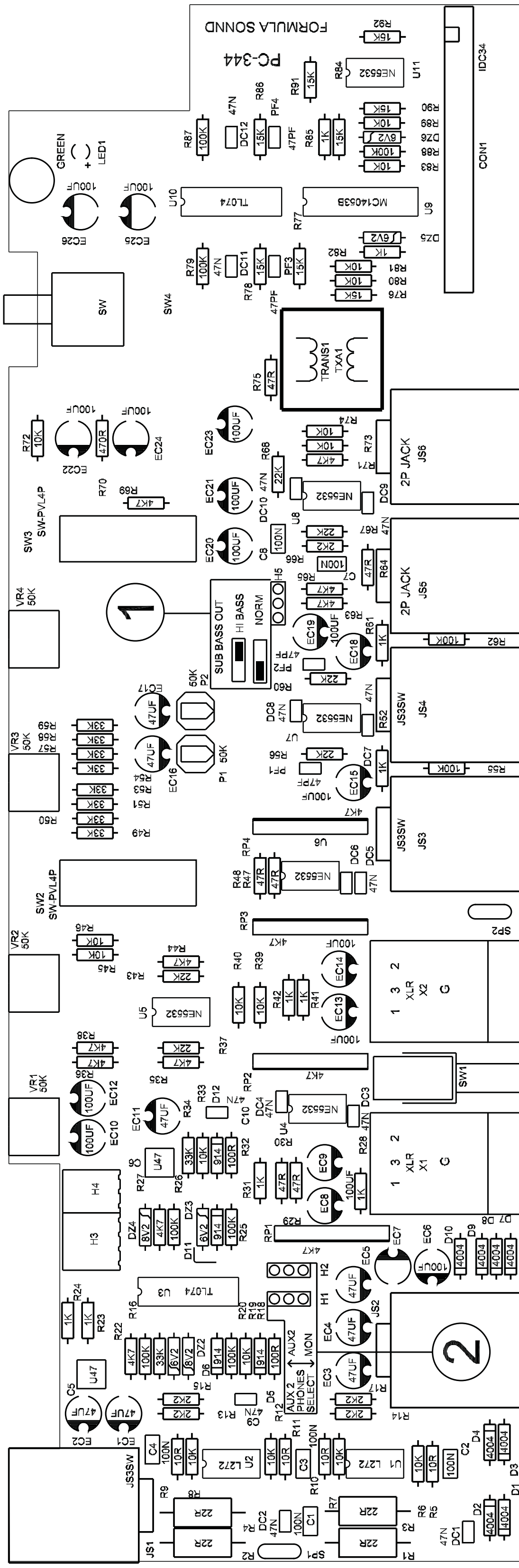
PM-100 OUTPUT MODULE
INTERNAL OPTIONS JUMPER LOCATIONS

PC- 342 1 ***2 JUMPER LINKS***
The position of these links selects microphone status when priority is activated. Move both links.
NORMAL position microphone channels are active. Priority acts as music mute.
MUTE position microphone channels are muted. Connect evacuation message system to priority input

PC - 344 1 ***1 JUMPER LINK*** selects sub bass output level. (Access by removing back panel)

PC - 344 2 ***2 JUMPER LINKS*** selects AUX 2 phones to follow AUX2 output or follow main phones output. (Access by long nose pliers or tweezers from open end)





Gain

In an audio mixer different signal levels from various items of equipment need to be amplified or attenuated to a common level so that they can be added or mixed together. These signal 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 necessary for the first or early stage in the input circuit to be a variable gain amplifier. The gain of this amplifier is set by the gain control.

The PM-100 universal input module has a combination of variable gain amplifiers to get the input signals to the required operating level. 3 gain trim presets are fitted - 1 for each input. These are accessable through holes on the rear of the module and are labelled accordingly. The stereo input gain trims are conventional single turn controls. The Mic preset is a multi-turn trimmer. These trims should be adjusted to set the maximum gain available for any input. The front panel gain control works in conjunction with these presets. The system may be considered as coarse gain control on the rear presets and fine adjustment on the front gain control.

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-100 has been designed to keep this noise as low as possible by using the latest technology. For the best performance set the gain control as low as possible to achieve the desired output level. 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 peak indicator has been included and is located below the front panel gain control. This indicator flashes when the signal level is close to clipping. It monitors the signal at various points in the signal chain, therefore, gain introduced by the equalisation or tone controls is also considered.
IF THE PEAK INDICATOR ILLUMINATES DURING USE TURN DOWN THE GAIN CONTROL (OR INPUT TRIM PRESETS).

INSTALLATION, 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 depending on the type and size of system and obviously well outside the scope of this handbook. We have included a few basic points for information 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 in connectors 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 the most common cause of hum (50Hz noise) on a system and is caused by incorrect system grounding.
When several items of audio equipment are connected together with unbalanced connections (i.e. 2 connections, single screened cable, etc.) 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 a distance apart 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 only once***. This is usually done at the mixer. You will need to investigate the various items of equipment 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.

PM-100 Specifications

Frequency response	
E.Q. set flat	20Hz - 20kHz +/-0.5dB
Maximum output level	
Main output and Zone output active balanced.	+20dBV
Distortion	
Stereo, auxiliary, mic busses etc. to related O/P's	< 0.01% typically <0.005%
Noise measured 20Hz-20kHz	
Stereo inputs e.q. flat gain set to max	EIN < -98dB
RIAA stage ref. 5mV 1kHz Input shorted	-80dBV "A" weighted
Microphone input (ref 150R)	EIN < -124dBU
Gain	
Stereo inputs rear trim @ max	+/- 15dB
Stereo inputs rear trim @ min	+ 0dB -20dB
Microphone input max	+ 75db
Microphone input min	+ 0dB
Maximum input level	
Stereo input A (line)	+20dBV
Stereo input A (RIAA)	-8dBV 400mV
Stereo input B	+20dBV
Microphone	+4 dBV
Input Impedances	
Stereo input A	> 47k ohms
Stereo input B	> 10k ohms
Microphone input	> 2k ohms active balanced

Equalisation

3 band e.q. per channel	+5dB -20 dB @	Treble 10kHz
		Mid 1kHz,
		Bass 100Hz

Internal channel jumpers allow the e.q. stage to be disabled if required.

Dimensions
(Exc. Knobs & connectors):

8 Channel	Width 483mm (19")
4 Channel	Width 330mm (13")
12 Channel	Width 635mm (25")

Height 312mm (12.25"-7RU)
Depth 110mm (4.33")

External PSU - PSU5 Width 175mm Height 75mm Length 230mm

Installation Cut-out
Height 312mm Width:- 4 channel 280mm. 8 channel 433mm. 12 channel 585mm

Balanced outputs For unbalanced operation strap pins 1&3 to ground and use pin 2 hot. This will result in no loss of output level or performance.

Chassis
The chassis is constructed from custom aluminium extrusions and is available in 4, 8 and 12 input module configurations. Finish is gun-metal grey anodised aluminium.
Module metalwork is also constructed of custom aluminium extrusions and is silver anodised. The module colour and notation is by inset reverse printed polyester film, which provides a hard wearing attractive finish.

Removing and Re- fitting Modules.

A removable cover on the rear of the mixer provides access to the ribbon cable which connects to all the modules. This cover is secured by 2 screws. Although modules can be removed and replaced from the mixer front when necessary removing the rear connector cover makes disconnecting modules easier particularly if more than one module is being removed.

To remove an input module from the chassis first disconnect all connections from the module and ensure that the power supply is turned off. Input modules are secured by 2 screws, removing the screws will release the module. Carefully ease the module out of the chassis and disconnect the ribbon cable connector.
The output module is secured by 4 screws and connected by 3 connectors to the ribbon cable assembly. We recommend that the rear connector cover is removed to facilitate disconnection of the ribbon cable if this module is removed.
The master module is normally factory fitted to the right hand end of the chassis although the design of the ribbon cable allows modules to be fitted into the chassis in any position. The output module is the width of 3 input modules and uses 3 of the connectors for its operation.
Ensure that the connectors are fully mated before refitting the rear cover.

The modules have a clearance of approx. 0.25mm. This is deliberate to allow the modules to be easily removed and refitted to the chassis. 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 1/1000”. 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” and looks unsightly.

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

All Formula Sound products are designed and manufactured in our own factory which enables us to maintain strict quality at every stage of manufacture. This attention to detail has helped to win us 14 industry awards to date and has earned us a world wide reputation for the high quality and reliability of our products.

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