Studiomaster Mixdown

Owners Manual

Contents

1 Introduction

1:1 GENERAL INTRODUCTION TO MIXDOWN

Please read this manual thoroughly from front to back, then you can be sure that you are getting the best from your MIXDOWN.

There are two sizes of MIXDOWN mixing console: 16.4.8 and 16.8.16. This manual has been written to cover both consoles (differences between the two are identified in Chapter 2).

Although they have been designed to meet the demands of today's studio environment, MIXDOWN mixing consoles can be used equally well in recording and live applications.

Many studios are now using more and more MIDI based outboard equipment and instruments. A normal console would require such instruments to be recorded via multitrack or use up valuable input channels to pass them to stereo. This would unnecessarily, but unavoidably, add noise to signals which are virtually perfect. The MIXDOWN consoles can avoid this as they have basic input sections routed direct to stereo and with monitoring controls. They are able to utilise the multitrack for "live" sounds (vocalists, guitars, brass etc.) only, while monitoring the sequenced MIDI keyboards, drum machines etc. connected to the AUXILIARY LINE INPUTS. At the 2-track mixdown stage, the multitrack recording would be mixed with the MIDI sequenced instruments from the basic input sections. This could make it possible for a smaller multitrack to be used, because the multi-output drum machines and keyboards - which would have normally consumed several tracks and made bouncedowns inevitable - could now bypass the multitrack operation totally.

The 16.4.8 has 26 inputs in total at mixdown and the 16.8.16 has 34.

DO NOT switch on the console until this section has been read:

Important!

1:2 READ THIS

BEFORE PROCEEDING: DO NOT CONNECT THE AC SUPPLY LEAD TO THE CONSOLE OR EXTERNAL POWER SUPPLY

The following section relating to the different supplies fitted to a MIXDOWN mixing console MUST be read before switching on. The standard console is available with either an internal power supply, or an external power supply.

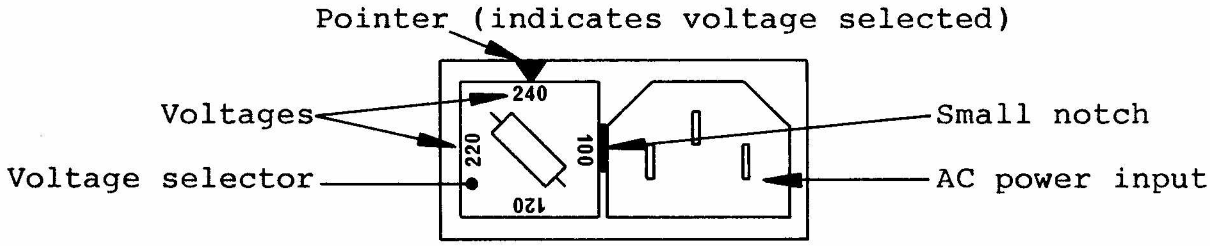
1) An **INTERNAL POWER SUPPLY** model is identified by the 3-pin power input socket found on the rear of the console.
2) An **EXTERNAL POWER SUPPLY** model is identified by a 6-pin (XLR type) DC connector found on the rear of the console. An external power supply Type EP4 would be supplied as standard with the console.

NOTE: The internal power supply and the EP4 external power supply are sufficient to allow expansion of a MIXDOWN by a further 8 input channels (2 expanders), a total of 24 inputs.

3) The **EXTERNAL POWER SUPPLY** model may be expanded by more than 2

expanders if used with the more powerful EP3 rack mounted external power supply (available separately).

The internally powered console and both the external power supplies have the same voltage selector/AC input/AC fuse assembly:



The voltage selector (the square shaped part which can be levered out) contains the AC fuse.

NOTE: External power supplies with their AC lead connected straight into them (no socket and connector) are fixed at 110 to 120V AC input and cannot be changed.

On the internally powered MIXDOWN, the voltage selector is fitted to the rear panel. On both external power supplies, it is at the back.

The voltage selection procedure is the same for all three variations. Check that the pointer (see diagram above) is pointing at the local AC supply voltage (100, 120, 220 or 240V stamped on the voltage selector). If it is, then go ahead and connect AC power to the console/external power supply.

If the voltage indicated is wrong, then lever out the voltage selector (there is a small notch for this purpose between the selector and the AC power input). Turn the selector round until the pointer is pointing to the correct voltage, then press it back in.

You will have noticed a fuse while the selector was out. This is the console's or power supply's AC fuse.

Internally powered consoles are now ready to be switched on. Externally powered models should now have the DC power lead from the power supply connected to the 6-pin socket on their rear panel. The power supply is now ready to be switched on.

REMEMBER, on an external power supply model, 48V Phantom Power has to be switched ON at the power supply before it can be switched ON at the console. 48V Phantom Power is explained in 4:11.

WARNING: ONLY use STUDIOMASTER MIXDOWN consoles with STUDIOMASTER external power supplies.

2 Control Identification

2:1 INPUT CHANNEL (Referring to FIG 1)

The 16.8.16 input channel is shown, features unique to the 16.4.8 are shown in shaded areas.

1 **MIC input** Electronically balanced female XLR wired Pin 1 = Ground, Pin 2 = -ve phase, Pin 3 = +ve phase. The channel may be driven with a low impedance balanced microphone. Gain range 15 to 60dB. Input impedance typically 2kohm (balanced). Maximum input level 0dBm.

2 **LINE input** Electronically balanced stereo 0.25" jack wired Sleeve = Ground, Ring = -ve phase, Tip = +ve phase. For unbalanced use, use a mono jack plug wired Sleeve = Ground, Tip = Signal. Gain range -15 to +30dB. Input impedance greater than 60kohm (balanced), greater than 30kohm (unbalanced). Maximum input level +30dBm.

3 **TAPE input** Unbalanced 0.25" jack wired Sleeve = Ground, Tip = Signal. Drives the input channel or TAPE monitor section with an unbalanced signal from the multitrack tape recorder. On expanders, this is marked "LINE B", for use as an extra LINE input. Input impedance typically 10kohm. Maximum input level +30dBm.

ON 16.4.8 (not shown)

3a **TAPE input (Inputs 1 to 8) Unbalanced 0.25" jack wired Sleeve = Ground, Tip = Signal. Drives the input channel or TAPE MONITOR section with an unbalanced signal from the multitrack tape recorder. Input impedance typically 10kohm. Maximum input level +30dBm.**

3b **LINE input (Inputs 9 to 16 and on expanders) Unbalanced 0.25" jack wired Sleeve = Ground, Tip = Signal. Input impedance typically 10kohm. Maximum input level +30dBm.**

4 **SEND/RETurn** Stereo 0.25" jack wired Sleeve = Ground, Ring = Return, Tip = Send, post-EQ and pre-fader. Return impedance 5kohm, nominal level +4dBm.

5 **DIRECT OUT** Mono 0.25" jack wired Sleeve = Ground, Tip = Signal. Post fader. Nominal level -10dBV. Minimum load (for reference level) 25kohms.

6 **+48V** button. When depressed, applies +48V phantom supply to Pins 2 and 3 of XLR MIC input for use with condenser microphones.

7 **TAPE** button switches the input channel from the MIC XLR or LINE jack to the TAPE jack when depressed, so that tape returns can be left connected thus eliminating the need for re-

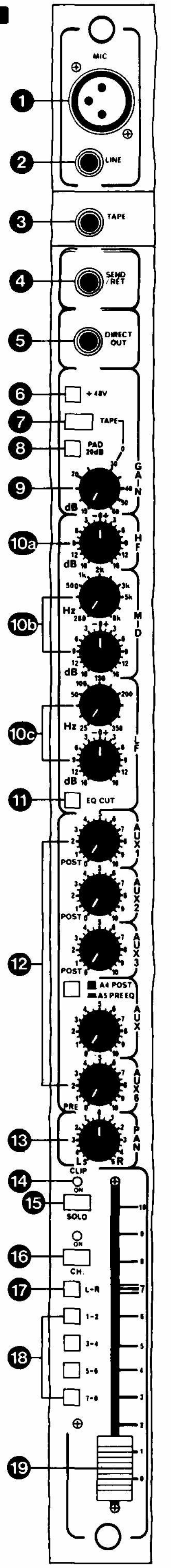


FIG 1

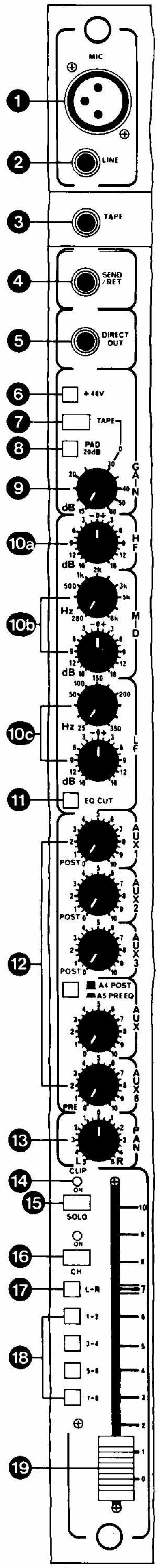


FIG 1

patching during remix. On expanders, this button selects LINE B input when depressed.

ON 16.4.8 (not shown)

7a **TAPE** button (Inputs 1 to 8) switches the input channel from the MIC XLR to the TAPE jack when depressed, so that tape returns can be left connected thus eliminating the need for re-patching for remix.

7b **LINE** button (Inputs 9 to 16 and on expanders) switches from MIC XLR to the LINE jack when depressed. This allows any input channel to be used with either a low impedance balanced microphone or unbalanced LINE level signal.

8 **PAD** switch provides 20dB attenuation to MIC inputs when depressed. It can be used to attenuate an incoming signal which, after turning the GAIN control to minimum, is still too high to be usable.

9 **GAIN** control with 45dB range ensures that the input - be it MIC, LINE or TAPE is at a suitable level to drive the channel. It is used to set the nominal channel signal level by varying the pre-amp gain, preserving maximum headroom and minimum noise without loading the input signal.

10 **EQUALISATION.** A 3-band semi-parametric design comprising of:

10a High Frequency control is an active shelving-type filter providing 16dB of cut or boost at 12kHz. The filter is a 2-pole design with a slope of 12dB/octave.

10b **MID** frequency network consists of a continuously variable centre frequency control with a sweep from 280Hz to 8kHz. A separate control provides 16dB of cut or boost at the chosen centre frequency. The filter is a 2-pole design with a peaking response, and a "Q" value of approximately 1.5.

10c Low Frequency network consists of a continuously variable shelving frequency control with a sweep from 25Hz to 350Hz. A separate control provides 16dB of cut or boost at the chosen shelving frequency. The filter is a 2-pole design with a slope of 12dB/octave.

11 **EQ CUT** switch bypasses the equalisation circuit for a flat frequency response.

12 **AUXILIARIES.** There are six auxiliary busses, five of which are simultaneously accessable on the channel. Auxs 1, 2, 3 & 4 are post-fade (for use mainly as effects sends). Aux 5 is pre-EQ and pre-fade. Aux 6 is pre-fade. Aux 5 and 6 are intended for studio cue/stage foldback purposes. A switch is provided to access either Aux 4 or

Aux 5.

13 **PAN** control allows a left/right stereo balance when the channel is routed to L-R outputs. It can also bias the channel's output to odd/even numbered groups by moving it from the zero position. Panning fully left routes to odd-numbered groups only, panning fully right routes to even-numbered groups only.

14 **CLIP** indicator. This LED illuminates whenever the signal approaches the maximum running level. It monitors all stages within the input channel and illuminates at 4dB prior to clipping. This LED also illuminates whenever the **SOLO** switch is depressed, to indicate that SOLO is in use. While indicating SOLO, it ceases to act as a CLIP indicator.

15 **SOLO** button. A pre-fade monitor switch which allows the channel signal to be isolated on the monitor/headphone system (along with any other channels which are in SOLO condition). The signal level can be seen on the RIGHT output bargraph.

16 **CH. ON** button and LED indicator. LED lights when the switch is depressed. In the OFF position most signals from the channel are muted. The exceptions are SOLO operation, CLIP indication and Aux 5 (as it is pre-EQ).

17 **L-R** routing button. When depressed, this routes the post-fader, post-pan signal directly to the left/right outputs, bypassing the subgroups. The signal may be used for final multitrack mixdown in recording studios or for routing of signals to the left/right outputs in live performance applications.

18 **ROUTING BUTTONS.** These allow bussing of the channel signals to any combination of group outputs simultaneously. For use in multitrack recording or subgrouping signals, such as drums, during live sound reinforcement. Exclusive routing is achieved using the PAN control.

19 **FADER.** High quality 100mm channel fader with 10dB gain.

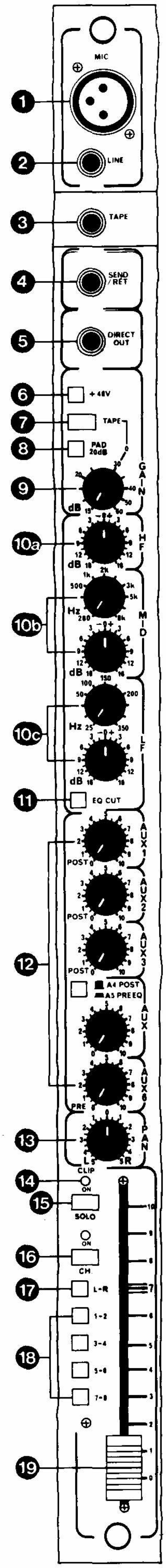


FIG 1

2:2 GROUP & OUTPUT SECTION (Referring to FIG 2)

The 16.8.16 output section is shown. The 16.4.8 differs only in having 4 group channels as opposed to 8.

1 **TAPE OUTPUTS** are 0.25" mono jacks wired Sleeve = Ground, Tip = Signal. These connect to the multitrack recorder inputs. Nominal level (Factory Set) is -10dBV (Min load 5kohm). An optional level of +4dBm can be selected (refer to Chapter 6).

2 **AUX LINE INPUTS** are 0.25" mono jacks wired Sleeve = Ground, Tip = Signal. These can be used as auxiliary returns or extra line inputs. Nos. 9 to 16 have 2-band EQ assigned to them when their input button (12) is depressed. Inputs 1 to 8 are connected to the lower monitors by connecting a jack to the AL input socket. This, unlike the upper monitors (9 to 16) means that to retain normal tape monitoring, the AL input jack must be removed. Input impedance typically 220kohm. Nominal input sensitivity is as the TAPE input sensitivity, -10dBV (Factory Set). +4dBm can be selected (refer to Chapter 6) and will also change the TAPE inputs to +4dBm. Maximum input level (at -10dBV) +4dBV. Maximum input level (at +4dBm) +18dBm.

3 **LED BARGRAPHS** to display output level (or tape monitor level if tape monitoring is selected) of the 4 groups. Group 4 bargraph displays SOLO level when a SOLO button on the console has been depressed. (The SOLO ACTIVE LED informs the operator when the RIGHT bargraph is displaying SOLO level, and not right output level). The bargraphs are 12 segment, 2 colour with typical VU meter ballistics. 0VU = ref. level -10dBV/+4dBm (To select optional output level of +4dBm, refer to Chapter 6).

4 **AL EQ.** This 2-band EQ is automatically in circuit when the switch 5 is released for the tape monitor controls to be used as an auxiliary input. This EQ is available only on AUX INPUTS 9 to 16. The HF shelving control gives 12dB of cut and boost at 12kHz. The LF shelving control gives 12dB of cut and boost at 60Hz. Excessive cut or boost may require input level adjustment at source to avoid low level or clipping.

5 **TAPE/AL INPUT** button (on TAPE MON 9 to 16). When depressed, (for TAPE), this brings the TAPE return signal (FIG 1, 3) to the monitor section. Its level can also be indicated on the associated channel bargraph by depressing the meter select button 19. The AUX LINE nominal input sensitivity is as the tape monitor (9 to 16) sensitivity (Factory Set -10dBV, +4dBm option). Aux line level may therefore need setting and adjusting with the source output level control.

6 **FADER REV** (on TAPE MON 5 to 8). This reverses the functions of the top TAPE MON level control and the group fader. The primary purpose for this is so a fader can be used on an auxiliary line input, making it almost as good as one of the 16 standard input channels. This button is also effective when the AL INPUT is not selected, so faders can be used for tape monitor (9 to 16) level if required.

7 The **MONITOR** level control sends its source (TAPE or AUX LINE input) to the left and right master outputs, and so to the monitors (headphones/amplifier system). The **MONITOR** level

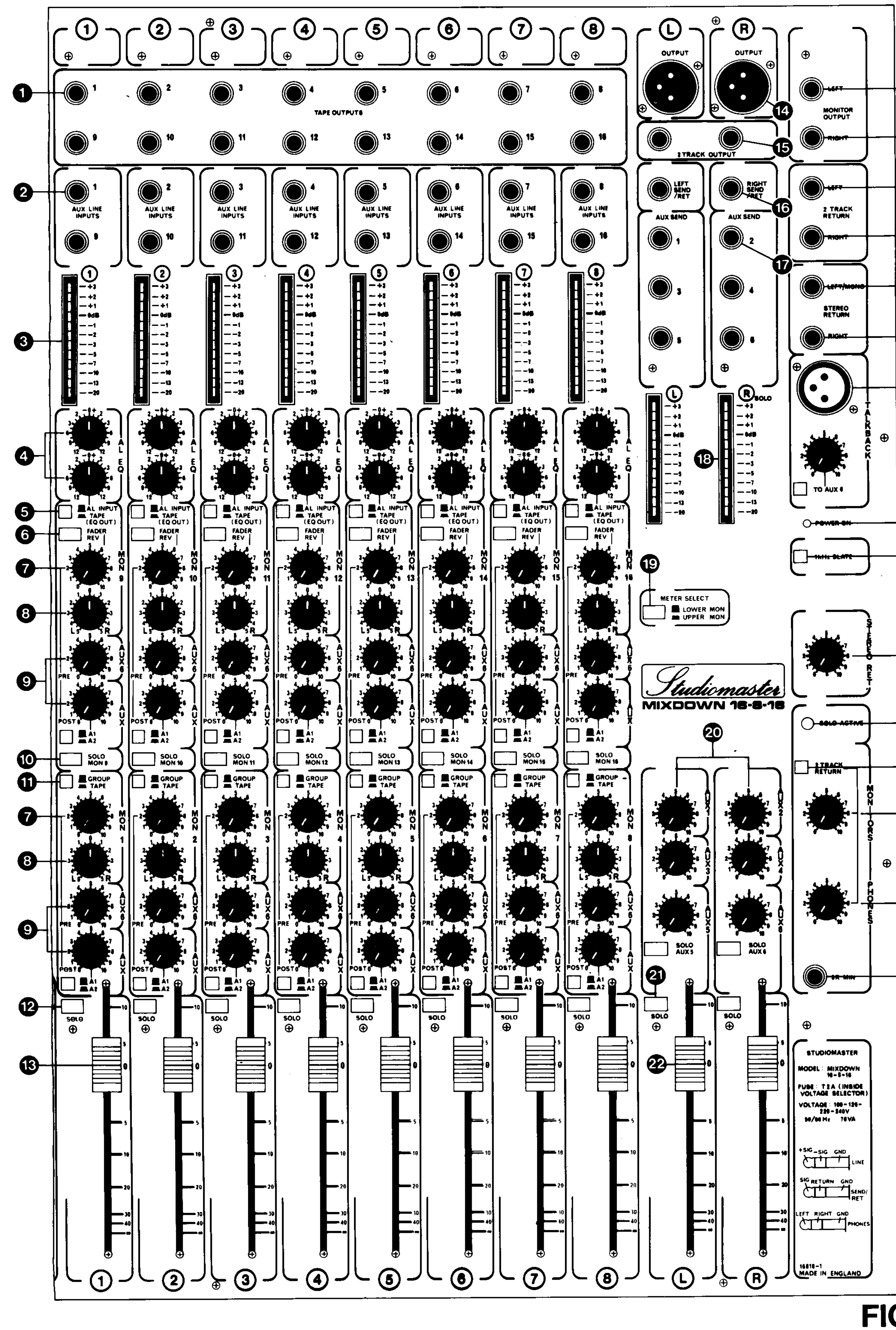

TAPE OUTPUTS
AUX LINE INPUTS
OUTPUT
MONITOR OUTPUT
LEFT
RIGHT
2 TRACK OUTPUT
LEFT SEND /RET
RIGHT SEND /RET
AUX SEND
2 TRACK RETURN
STEREO RETURN
TALKBACK
TO AUX 6
POWER ON
METER SELECT
LOWER MON
UPPER MON
Studiomaster
MIXDOWN 16-8-16
SOLO ACTIVE
2 TRACK RETURN
MONITORS
PHONES
STEREO RET
SOLO AUX 5
SOLO AUX 6
SOLO
STUDIOMASTER
MODEL: MIXDOWN 16-8-16
FUSE: T2A (INSIDE VOLTAGE SELECTOR)
MADE IN ENGLAND

FIG 2

control on the monitors 9 to 16 may be transposed with the group fader by depressing FADER REV (6).
NOTE: The left/right faders must be raised to allow the signal to reach the headphones/monitors.

8 **PAN** control determines left/right balance of the subgroup/tape return/aux line input bussed to the monitor output and master outputs.

9 **AUXILIARIES** 3 auxiliaries are available: Aux 1, Aux 2 and Aux 6. Aux 6 is mainly for foldback, but Auxs 1 & 2 are for effects. Aux 1 or 2 is selected by a switch below the bottom aux control.

10 **SOLO MON** routes the monitor signal source (TAPE MON or AUX LINE input 9 to 16) to the monitor system. This button is pre-monitor level.

11 **GROUP/TAPE** button (on TAPE MON 1 to 8). This assigns the monitor controls to either a tape return input, or a subgroup output.

12 **SOLO** button (on GROUPS 1 to 8). Routes the pre-fade subgroup signal to the monitor system.

13 The **FADER** on subgroups is a 100mm control provided with dB calibrations for accurate relative signal indication. This determines the final signal level available at the subgroup output jack as well as to all post fader busses. As mentioned earlier, by using the FADER REV button, this fader can be used for the AUX LINE input.

14 **LEFT/RIGHT OUTPUTS.** Balanced Male XLRs wired Pin 1 = Ground, Pin 2 = Out-phase, Pin 3 = In-phase. Nominal output level (Factory Set) is +4dBm unbalanced. An option of +4dBm balanced is available (refer to Chapter 6).

15 **LEFT/RIGHT OUTPUTS.** Unbalanced 0.25" mono jacks wired Sleeve = Ground, Tip = Signal. Nominal level -10dBV. Minimum load 5kohms.

16 **SEND/RETURN** 0.25" stereo jacks wired Sleeve = Ground, Ring = Return, Tip = Send. Nominal level +4dBm.

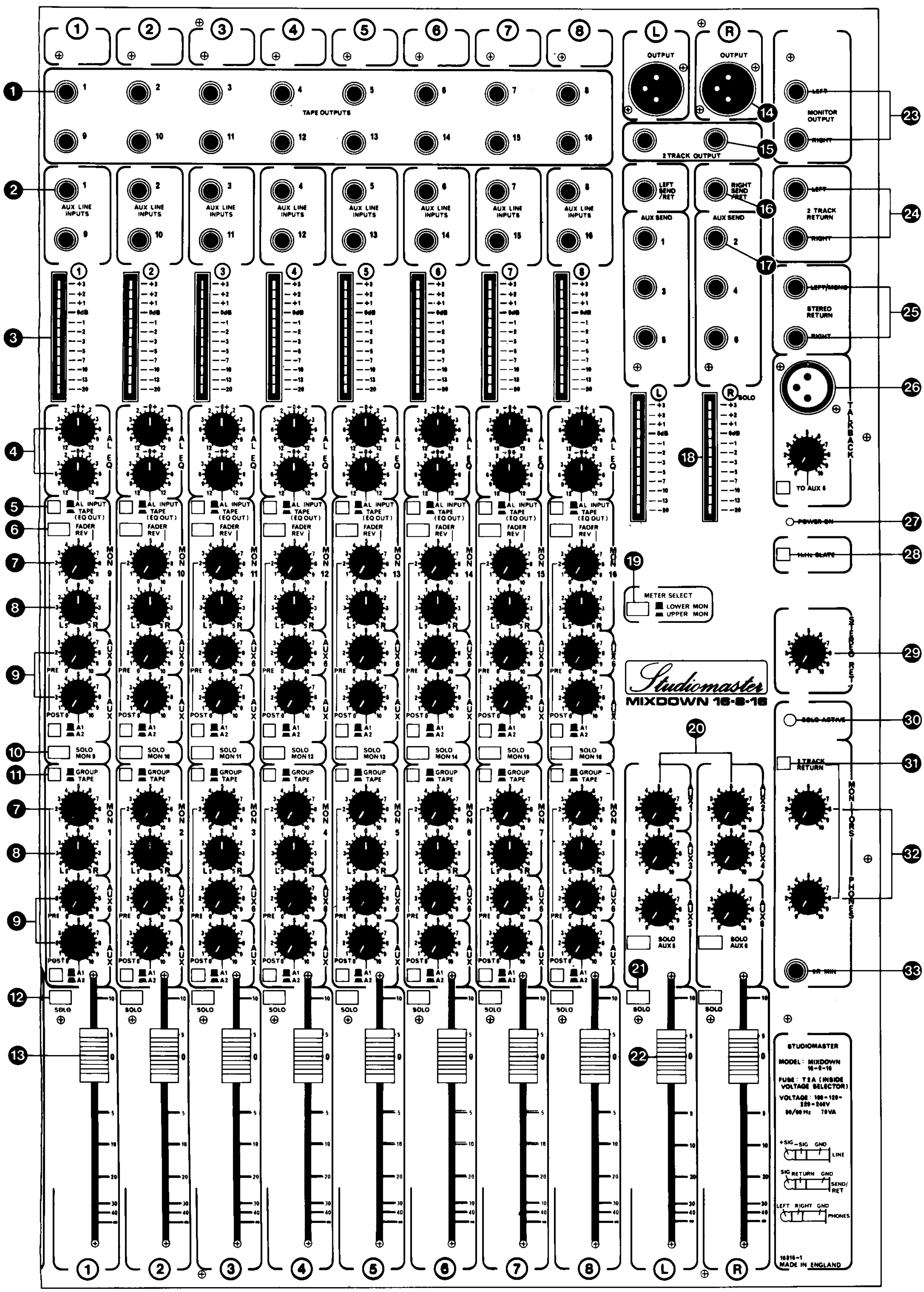
17 **AUX SENDS** 0.25" mono jacks wired Sleeve = Ground, Tip = Signal.

18 **LED BARGRAPHS** displaying left and right output level. The bargraphs are 12 segment, 2 colour with typical VU meter ballistics. 0VU = ref. level +4dBm XLR/-10dBV jack.

19 **METER SELECT** allows the bargraphs to indicate the upper monitor source signal level - TAPE MON or AUX LINE input 9 to 16.

20 **AUX MASTER SENDS.** These control the level of the auxiliary mix sent to the effects and foldback system. As Aux 5 and Aux 6 are generally intended for foldback, they have a useful SOLO button on the master sends for the engineer to check foldback mixes for level and sound quality. Output impedance less than 50ohm. Maximum load 600ohm. Maximum output level +18dBm (600ohm).

21 **SOLO** button. This is pre-fade and allows monitoring of the left/right master signals.



TAPE OUTPUTS
AUX LINE INPUTS
OUTPUT
2 TRACK OUTPUT
MONITOR OUTPUT
LEFT
RIGHT
2 TRACK RETURN
LEFT/MONO
STEREO RETURN
LEFT SEND /RET
RIGHT SEND /RET
AUX SEND
TALKBACK
TO AUX 6
POWER ON
1kHz SLATE
AL EQ
AL INPUT
TAPE (EQ OUT)
FADER REV
PRE
POST
A1
A2
GROUP
TAPE
SOLO
METER SELECT
LOWER MON
UPPER MON
Studiomaster
MIXDOWN 16-8-16
STEREO RET.
SOLO ACTIVE
2 TRACK RETURN
MONITORS
PHONES
SOLO AUX 5
SOLO AUX 6
STUDIOMASTER
MODEL: MIXDOWN 16-8-16
FUSE: T2A (INSIDE VOLTAGE SELECTOR)
VOLTAGE: 100-120-220-240V 50/60Hz 70VA
LINE
SEND/RET
PHONES
MADE IN ENGLAND

FIG 2

22 **MASTER FADERS,** dB calibrated for accurate relative signal level indications. These control the final signal bussed to the left and right XLR or jack outputs.

23 **MONITOR OUTPUT** 0.25" mono jacks wired Sleeve = Ground, Tip = Signal. Output impedance less than 50ohm. Maximum load 1kohm. Maximum output level +19dBm.

24 **2-TRACK RETURN** 0.25" mono jacks wired Sleeve = Ground, Tip = Signal. Input impedance 5kohm. Maximum input level infinite.

25 **STEREO RETURN** 0.25" mono jacks wired Sleeve = Ground, Tip = Signal. Input impedance 10kohm. Maximum input level infinite.

26 **TALKBACK.** 3 pin female XLR wired Pin 1 = Ground, Pin 2 = Ground, and Pin 3 = Signal. Minimum input impedance 47kohm, gain range 0 to 40dB. The volume control determines the level of the talkback mic signal to be mixed onto the Aux 6 buss. The **TO AUX 6** button is a momentary push-to-talk switch which while depressed mixes the talkback voice with the Aux 6 signal. This allows communications to the stage monitoring system during live sound reinforcement or to the foldback system in the studio during recording. It may also be used as part of an intercom system for on or off stage direction communications during complex performances

27 **POWER ON** indicator. Lights when AC power has been switched on at the rear of the mixing console (or at the external power supply if used).

28 **1kHz SLATE.** In operation, it applies a 1kHz signal to all group busses for aligning the meters and lining up recording equipment. By using the MON controls on the groups, this signal may be sent to the left/right outputs, thus the tone may be used for lining up recording equipment, slating tapes and system checking in live applications.

29 **STEREO RETURN.** The gain range is from 0 to a maximum of 24dB, catering for a very wide range of inputs.

30 **SOLO ACTIVE** flashing LED warns that a SOLO button is depressed somewhere on the console and that the RIGHT bargraph is not displaying RIGHT output level.

31 **2-TRACK RETURN** button routes the 2-track return onto the monitors. This overrides all other monitor/SOLO signals.

32 **PHONES** and **MONITOR** volume controls. MONITOR volume control determines the overall level of any signal bussed via the left/right output and on to the monitor amplifier and speakers. The left/right faders must be raised so that the signal reaches the monitor volume control. The PHONES volume control is identical in operation to the MONITOR but provides independent level control for the headphone output.

33 **PHONES.** For stereo headphones wired Sleeve = Ground, Ring = Right, Tip = Left. The PHONES amplifier is designed to cater for all impedances from 600ohm to a minimum of 8ohm.

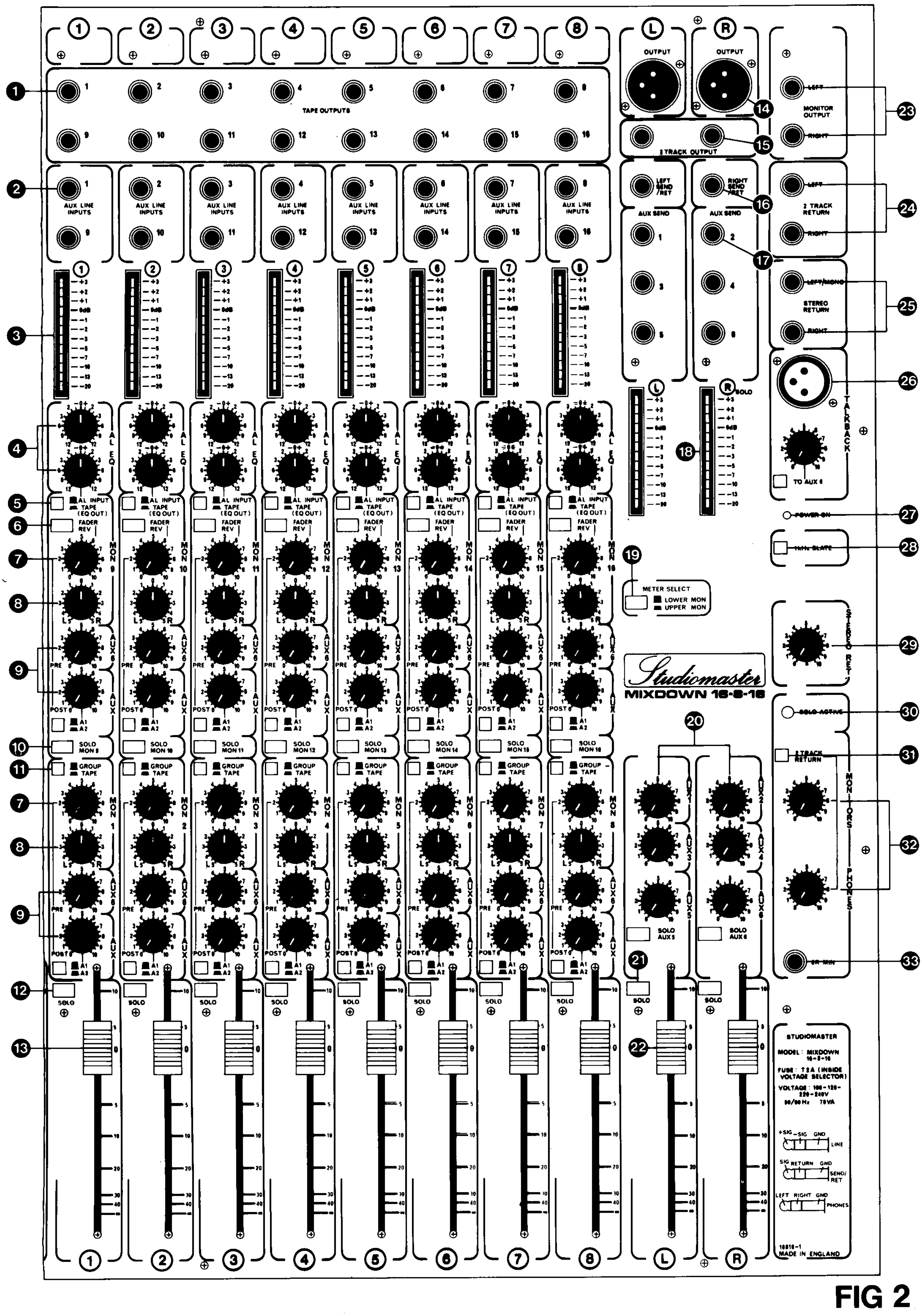

TAPE OUTPUTS
AUX LINE INPUTS
OUTPUT
OUTPUT
MONITOR OUTPUT
LEFT
RIGHT
2 TRACK OUTPUT
LEFT SEND /RET
RIGHT SEND /RET
AUX SEND
AUX SEND
2 TRACK RETURN
STEREO RETURN
LEFT/MONO
TALKBACK
TO AUX 6
POWER ON
METER SELECT
LOWER MON
UPPER MON
Studiomaster
MIXDOWN 16-8-16
STEREO RET
SOLO ACTIVE
2 TRACK RETURN
MONITORS
PHONES
INPUT
TAPE
(EQ OUT)
FADER REV
PRE
POST
SOLO
GROUP
TAPE
SOLO AUX 5
SOLO AUX 6
STUDIOMASTER
MODEL: MIXDOWN 16-8-16
MADE IN ENGLAND

FIG 2

3 Connections

This chapter shows connections to and from the MIXDOWN console (a 16.8.16 has been shown), and the tables explain the conditions under which the conditions may be safely made, without damaging equipment.

3:1 - INITIAL CONNECTIONS TO THE CONSOLE

As these are preliminary connections, the console should be OFF. If the conditions do not state that the console should be OFF, then the conditions stated AT LEAST must be observed.

The numbers refer to FIG 3:

	CONNECTION	CONDITIONS
1	Outputs of multitrack tape recorder to TAPE inputs	Multitrack tape recorder OFF
2	TAPE (subgroup) OUTPUTS to multitrack tape recorder inputs	Multitrack tape recorder OFF, subgroup faders down (infinity)
3	Effect unit output to LINE input	Effect unit (or other source) OFF
4	Effect unit outputs to AUX LINE INPUTS	Effect units (or other sources) OFF
5	XLR LEFT/RIGHT outputs to either +4dBm 2-track tape recorder, or P.A. inputs (not shown)	2-track tape recorder OFF, or P.A. system OFF. LEFT and RIGHT output faders down (infinity)
6	2-TRACK OUTPUTS to -10dBV 2-track tape recorder inputs	2-track tape recorder OFF, LEFT and RIGHT output faders down (infinity)
7	LEFT and RIGHT SEND/RETS to graphic equalisers, or Noise Reduction units (not shown)	Graphic equaliser/Noise Reduction OFF, left and right faders down (infinity)
8	AUX 1 - 4 SEND outputs to effect units inputs	Effects OFF, AUX 1 to 4 master send controls turned to 0
9	AUX 5 & 6 SEND outputs to performers' monitor system inputs	Headphone/stage amplifiers OFF, AUX 5 & 6 master send controls turned to 0
10	MONITOR OUTPUT to monitor amplifier input	Monitor amplifier OFF, MONITOR level turned to 0
11	2-track tape recorder outputs to 2-TRACK RETURN inputs	2-track tape recorder OFF, "2-TRACK RETURN" button released
12	CD player, or stereo effect output (not shown) to STEREO RETURN inputs	CD player, or stereo effect, OFF
13	Microphone to TALKBACK input	Do not press "TO AUX 6" button while connecting the microphone
	Stereo headphones to 8ohm min. PHONES output	HEADPHONE level turned to 0

Now, before turning on the console, make sure the 1kHz SLATE is OFF (button UP).

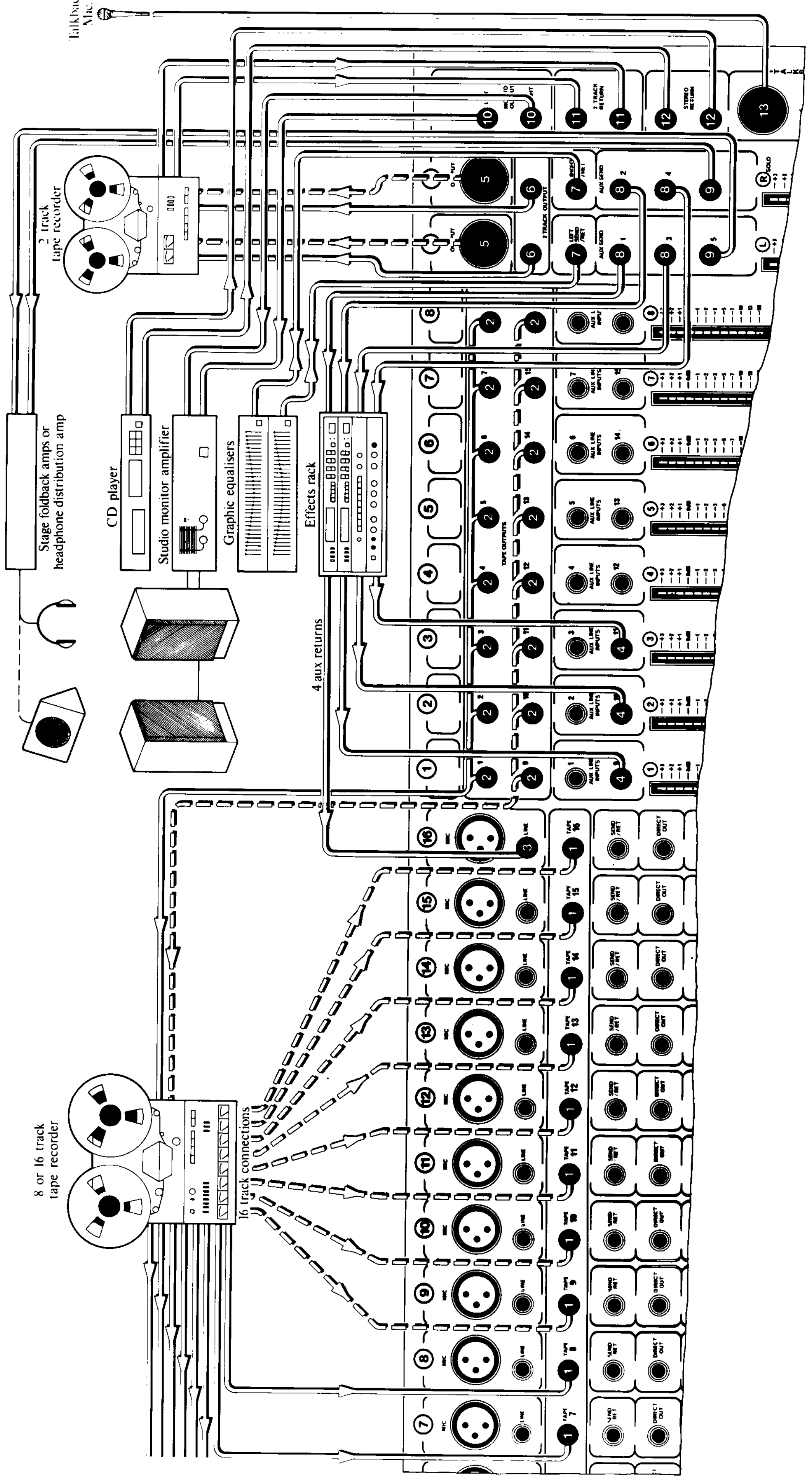

Talkback Mic
2 track tape recorder
Stage foldback amps or headphone distribution amp
CD player
Studio monitor amplifier
Graphic equalisers
Effects rack
4 aux returns
8 or 16 track tape recorder
16 track connections

FIG 3

3:2 - TYPICAL CONNECTIONS FOR A RECORDING SYSTEM

This shows a typical recording system. Most of the connections have already been made and their conditions explained in 3:1. New connections are explained below.

The numbers refer to FIG 4:

	CONNECTION	CONDITIONS
1	MIC input	Input channel GAIN turned to 15
2	LINE input	Input channel GAIN turned to 15
3	Noise Gate to SEND/RET	Input channel fader down to 0, AUXS 5 & 6 master sends (studio cues) turned to 0
4	DIRECT OUT to sidechain of Noise Gate	Input channel fader down to 0, "sidechain listen" OFF on Noise Gate
5	DIRECT OUT to input of auto-panner	Input channel fader down to 0, auto-panner OFF
6	Drum machine output to AUX LINE INPUT	Drum machine OFF, MONO control turned to 0
7	Auto-panner output to STEREO RETURN input	Auto-panner OFF, STEREO RETURN level turned to 0

MIXDOWN consoles have been designed with sufficient inputs and switching capabilities to allow nearly all connections to remain in place throughout a recording session. TAPE OUTPUTS and TAPE inputs can be switched in and out of place when required, so tape returns can be left connected, without removing LINE inputs.

The AUX LINE INPUTS allow input channels to be kept for vocal, acoustic instrument and multitrack return processing during 2-track mixdown, without filling them with effects returns or synthesised sounds which would probably not need the extensive EQ and effects facilities offered by an input channel. The upper AUX LINE INPUTS (5 to 8 on 16.4.8, or 9 to 16 on 16.8.16) do however offer basic EQ, and all AUX LINE INPUTS can have effects added via AUX 1 or 2.

The AUX LINE INPUTS are for recording sounds at the 2-track mixdown stage only, as they are not routable to subgroups. The upper AUX LINE INPUTS (5 to 8 on 16.4.8, or 9 to 16 on 16.8.16) can be left connected during multitrack recording processes, but the lower ones need to be disconnected when tape monitoring.

A Noise Gate, such as the Studiomaster IDP1 or Studiofex SF800, is connected to the SEND/RET jack of an input channel with a bass guitar. A DIRECT OUT, from an input channel with a bass drum, is used to trigger the Noise Gate through its sidechain.

A DIRECT OUT from a channel with an acoustic guitar has been connected to the input of an auto-panner which is brought back to the console via the STEREO RETURN.

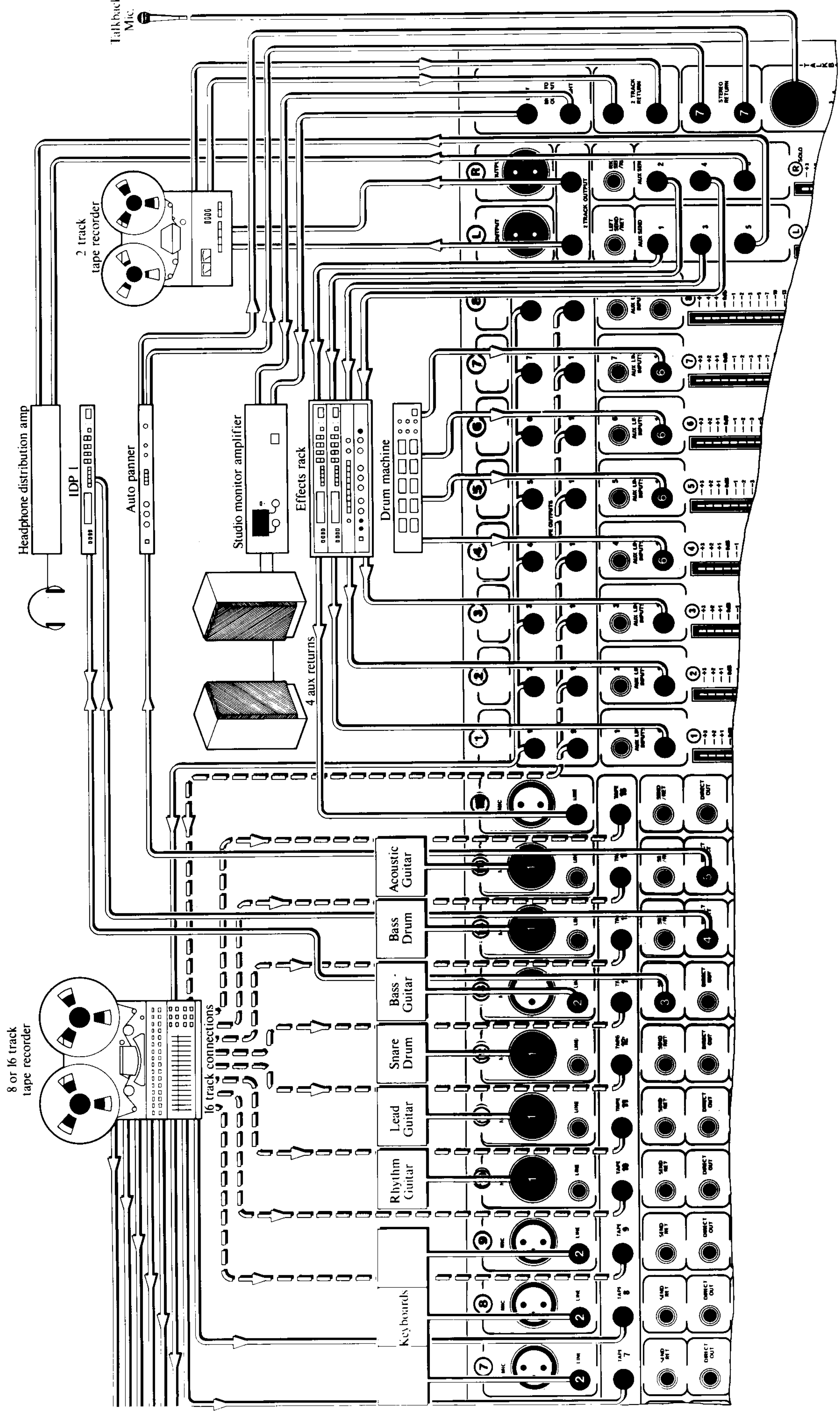

Talkback Mic
2 track tape recorder
Headphone distribution amp
IDP 1
Auto panner
Studio monitor amplifier
Effects rack
Drum machine
4 aux returns
Acoustic Guitar
Bass Drum
Bass Guitar
Snare Drum
Lead Guitar
Rhythm Guitar
Keyboards
8 or 16 track tape recorder
16 track connections

FIG 4

3:3 - TYPICAL CONNECTIONS FOR A LIVE SYSTEM

This shows an example of a live system. Most of the connections have already been made and their conditions explained in 3:1. New connections are explained below.

The numbers refer to FIG 5:

	CONNECTION	CONDITIONS
1	MIC input	Input channel GAIN turned to 15
2	LINE input	Input channel GAIN turned to 15
3	Outputs of multitrack tape recorder to TAPE inputs	Multitrack recorder OFF
4	TAPE (subgroup) OUTPUT to LINE input	Subgroup fader down (infinity), input channel GAIN turned to 15
5	Noise Gate to SEND/RET	Input channel fader down to 0, AUXS 5 & 6 master sends (foldback) turned to 0
6	DIRECT OUT to sidechain of Noise Gate	Input channel fader down to 0, "sidechain listen" OFF on Noise Gate
7	DIRECT OUT to input of auto-panner	Input channel fader down to 0, auto-panner OFF
8	Outputs of drum machine to AUX LINE INPUTS	Drum machine OFF, MON control turned to 0
9	TAPE (subgroup) OUTPUTS to multitrack tape recorder inputs	Multitrack tape recorder OFF, subgroup faders down (infinity)
10	Auto-panner output to STEREO RETURN input	Auto-panner OFF, STEREO RETURN level turned to 0

The diagram shows a 16.8.16 being used for live work. Instrument inputs are MIC, either from microphones, or from line level equipment through direct injection boxes. Subgroup outputs 1 to 4 have been used for submixing, making AUX LINE INPUTS 1 to 4 unavailable. Subgroup 1 has been connected to an input channel for effects to be added. Subgroups 2 to 4 would be routed to the LEFT and RIGHT outputs by their MONitor and PAN controls.

A live session recording is made by the 4-track tape recorder connected to subgroups 5 to 8. Outputs from the 4-track tape recorder have been connected to TAPE inputs 5 to 8.

It may be preferable in a system to not use subgroups for submixing - particularly when a band may be using many synthesised instruments which require little or no processing by the console. All the AUX LINE INPUTS being available for such instruments would be desirable. Indeed, multitrack session recordings may not be required; an adequate recording of the performance could be made from the LEFT and RIGHT output jack sockets, or from the post-fade DIRECT OUTS on input channels.

A Noise Gate, such as the Studiomaster IDP1 or Studiofex SF800, is connected to the SEND/RET jack of an input channel with a bass guitar. A DIRECT OUT, from an input channel with a bass drum, is used to trigger the Noise Gate through its sidechain.

A DIRECT OUT from a channel with an acoustic guitar has been connected to the input of an auto-panner which is brought back to the console via the STEREO RETURN.

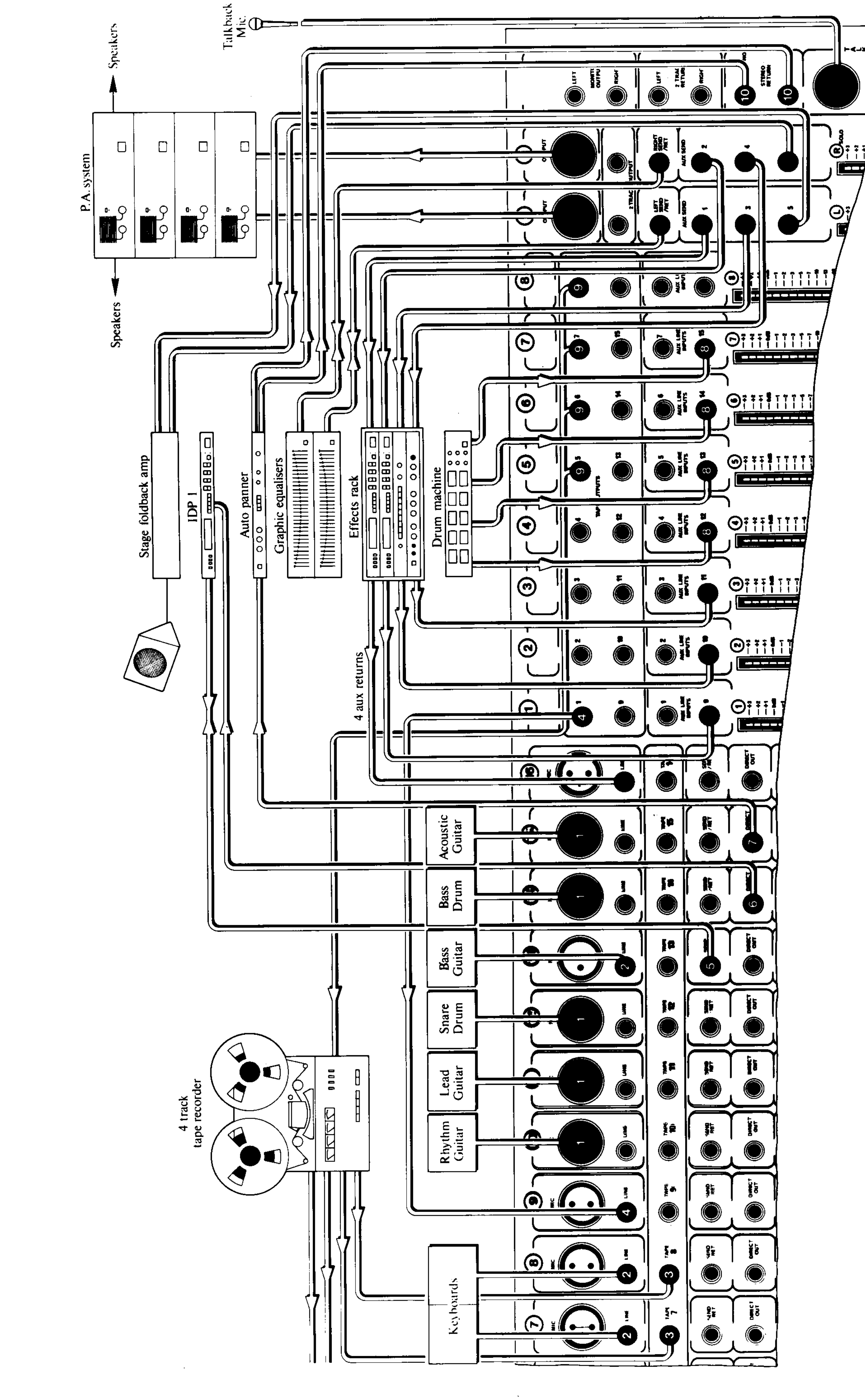

Speakers
P.A. system
Speakers
Talkback Mic
Stage foldback amp
IDP 1
Auto panner
Graphic equalisers
Effects rack
Drum machine
4 aux returns
Acoustic Guitar
Bass Drum
Bass Guitar
Snare Drum
Lead Guitar
Rhythm Guitar
4 track tape recorder
Keyboards

FIG 5

4 Getting to know Mixdown

This chapter assumes that all initial connections to the console have been made as outlined in 3:1.

Note: In this chapter switch positions are described as "released" or "depressed". Released is the "up" position, and depressed is the "down" position.

4:1 FIRST THINGS FIRST

Set all console controls to their minimum, or zero position: all levels fully anticlockwise, pans to centre, faders to zero (on inputs) or infinity (on outputs). Make sure all buttons are set OUT (released), except EQ CUT buttons which must be set IN (depressed). Switch on the console.

4:2 THE INPUT CHANNEL CONNECTORS (FIG 1: 1-5)

THE 16.8.16

On each input channel there are five sockets: a female XLR labelled MIC, a 0.25" stereo jack labelled LINE, a 0.25" mono jack labelled TAPE, a 0.25" stereo jack labelled SEND/RET and a 0.25" mono jack labelled DIRECT OUT. The MIC socket is designed for low level signals, typically generated by microphones. The LINE socket is for higher level (or "line level") signals produced by drum machines, keyboards, some electric guitars, tape players etc.. Both the MIC and LINE inputs are electronically balanced to eliminate ground buzz problems and deliver superb transient response. The LINE input can be used unbalanced by using a 0.25" mono jack. The TAPE socket is for returns from the multitrack tape recorder. The SEND/RET and DIRECT OUT jacks have special purposes explained in 4:9 and 4:10.

THE 16.4.8

On each input channel there are four sockets: a female XLR labelled MIC, a 0.25" mono jack labelled either LINE or TAPE, a 0.25" stereo jack labelled SEND/RET and a 0.25" mono jack labelled DIRECT OUT. The MIC socket is designed for low level signals, typically generated by microphones. It is electronically balanced to eliminate ground buzz problems and deliver superb transient response. The LINE socket (on inputs 9 to 16) is for higher level (or "line level") signals produced by drum machines, keyboards, some electric guitars, tape players etc.. The TAPE socket (on inputs 1 to 8) is primarily for returns from the multitrack tape recorder. It can be used as a LINE input if no tape recorder is used. The SEND/RET and DIRECT OUT jacks have special purposes explained in 4:9 and 4:10.

For the purpose of setting up the console, connect a keyboard or tape recorder to the LINE input of channel 1 on a 16.8.16, or channel 9 of a 16.4.8 so that a continuous signal is fed into the console. (Refer to Chapter 2 for wiring of jack plugs.)

At this stage a signal has entered the MIXDOWN but cannot be heard.

4:3 HOW TO GET THE INPUT SOUND THROUGH TO THE MONITOR SPEAKERS

1. **The CH. ON button (FIG 1: 16).** Depress this button (adjacent green LED will light) and the input channel is now ON.

2. **The LINE button (FIG 1: 7b).** on channel 9 must be depressed to select the LINE input rather than the MIC input. This is unnecessary on a 16.8.16 as the MIC and LINE inputs are

electronically linked, but make sure the TAPE button (FIG 1: 7) is released. **NEVER** connect sources to both MIC and LINE inputs simultaneously on a 16.8.16 - this may result in damage to the source or the console.

3 **The GAIN control (FIG 1: 9).** Adjust this control until it reads about 20 on its scale. The signal within the input channel has its level determined by this control.

4 **The CLIP indicator (FIG 1: 14).** Check that this LED is not lit. This will only light when the signal at any point in the channel is too high. It illuminates just before "clipping" occurs, to warn that the channel is near its limit. If a signal clips, then it will be distorting. Should it light, take either of the following steps:
a Reduce the GAIN control level until the light goes off.
b Reduce the output level of the input source (keyboard etc.).

5 **The PAD button (FIG 1: 8).** This is used when turning the input GAIN control anti-clockwise does not lower the incoming signal to a usable level ie the CLIP LED stays ON and the sound is distorting. It gives an attenuation of 20dB on the MIC input and (on the 16.8.16 only) the LINE input.

6 **The ROUTING buttons and PAN control (FIG 1: 18 & 13).** Adjust the PAN control fully to the left. Depress the routing button labelled 1-2. The PAN control and routing buttons work like this: having pressed routing button 1-2 and panning fully left, the input source has been bussed exclusively to subgroup 1. Pan fully right and the signal is now routed exclusively to subgroup 2. By panning between left and right, the signal is swept between subgroups 1 and 2, or with pan centred, routed to both equally. All odd numbered subgroups (and the LEFT output) are exclusively routed to by panning LEFT. All even numbered subgroups (and the RIGHT output) are routed to by panning RIGHT. Of course, if all routing buttons are depressed and the PAN control is centred, all subgroups will have the signal from the input source. (**NOTE:** At this point, there will still be no sound as no faders have been raised).

7 **The input channel and subgroup FADERS (FIG 1: 19 & FIG 2: 13).** The signal going through the input channel, PAN and routing buttons is finally controlled by the input fader and, for the time being, is best set at 7 on the scale. This presents the incoming signal to subgroup 1 and its level is now passed directly to the subgroup 1 fader. (Make sure the GROUP/TAPE button (FIG 2: 11) is released - GROUP selected.) Raising this fader will now bring the signal onto bargraph 1, but not to the headphones/monitors. (**DO NOT** allow the signal to go beyond the first red LED on the bargraph, lower the subgroup fader rather than the input fader should this occur.)

8 **The MONitor and PAN controls (FIG 2: 7 & 8).** To now bring the signal from the subgroup into the headphones/monitors, the MONitor control on the subgroup must be turned clockwise. This sends the sound to the monitors. The PAN control positions the signal between the left and right monitors (just like the input channel PAN control.

9 **The LEFT and RIGHT (or MASTER) faders (FIG 2: 22)** now need to

be raised. Finally, turn up the headPHONES or MONITOR level controls. The signal has now reached the headphones/monitors.

4:4 THE EQUALISATION SECTION ON THE INPUT CHANNELS (FIG 1: 10)

Equalisation is, put simply, like the tone controls on hi-fi systems, but on mixing consoles is more flexible and powerful. The function of equalisation is to increase or decrease selected frequencies to either achieve a particular sound, or eliminate an unpleasant sound characteristic. The EQ (short for equalisation) on STUDIOMASTER MIXDOWN consoles is rather special as it provides two types of adjustment. Refer back to the input channel now heard in the headphones/monitors. The EQ, at the moment, should be flat, or out of circuit, as EQ CUT (**FIG 1: 11**) was depressed at the start of this chapter. To set the EQ roughly flat, with the EQ in circuit, set to zero (centre) the three knobs calibrated in dB, and turn fully to the right the two knobs with frequency calibrations. With the EQ set like this, pushing EQ CUT in and out should result in little or no difference in the sound. However, with the EQ in circuit, turn up (clockwise) the HF (treble) control. Press EQ CUT in and out and compare the sounds. Notice the difference? High, or treble, frequencies have been boosted. The EQ network covers the whole range of audible frequencies from treble to bass.

What makes this network special is that on mid and bass, not only can frequencies be cut and boosted, but the centre frequency to cut or boost can be chosen by adjusting the controls with frequency calibrations. Experiment with the effect created by moving these controls. This type of network is called "SEMI-PARAMETRIC". The electronic characteristics and frequencies chosen by STUDIOMASTER designers are unique and are developed to SOUND good. Its main purpose is to correct deficiencies in a signal's frequency response, but used creatively it can also enhance sounds. It is an EQ network that when mastered can make mediocre sounds good, and good sounds excellent. Practise is recommended.

4:5 THE SOLO BUTTON

On input channels (FIG 1: 15).

The SOLO button is used when two or more signals are present in the console (all inputs are initially set up following the procedures outlined in 4:3 and 4:4). Depress a SOLO button on an input channel carrying a signal. Notice (while listening to the headphones/monitors) that it has excluded all other signals, hence the term "SOLO". Releasing the button brings back the other signals. Four things will have occurred while the SOLO button was depressed:

- a the signal from the SOLOed channel is isolated on the headphones/monitors,
- b the CLIP LED on the SOLOed channel will be on continuously, indicating that SOLO is selected,
- c the level of the SOLOed signal is shown on the RIGHT bargraph (**FIG 2: 18**),
- d the SOLO ACTIVE LED (**FIG 2: 30**) flashed.

Here is an example of where SOLOing is needed:

A whole series of microphones, keyboards, guitars, drums etc. has been set up and from somewhere a rogue hum or some other type of interference is being emitted. By depressing and releasing each of the input channel SOLO buttons in turn (thereby isolating each sound source on the headphones/monitors) the troublesome instrument will eventually be located. The SOLO ACTIVE LED warns that a SOLO button is depressed and also that the RIGHT output bargraph is no longer displaying the RIGHT output level. Now the problem can be corrected, the SOLO button released and normal monitoring resumed.

Other SOLO buttons (FIG 2: 10, 12, 20 and 21).
These other SOLO buttons work in the same way as those on the input channels, and serve the same purpose: to examine individual sounds when there are many in the console. These other buttons allow SOLOing of AUX LINE INPUTS, subgroups, AUX 5 & 6 master sends and the LEFT and RIGHT outputs.

4:6 THE CHANNEL ON BUTTON (FIG 1: 16)

Each input has a CH. ON button, and when depressed, an adjacent green LED illuminates. When released, it isolates the channel from the following: ALL subgroups, LEFT & RIGHT, AUXS 1 to 4 and the AUX 6 buss. This allows the input fader setting to be maintained, but the signal can be quickly cut from the main mix. The following will continue to function: CLIP indication, SOLO and AUX 5. AUX 5 continues to operate because it is the signal at the pre-amp, pre-EQ.

4:7 THE L-R ROUTING BUTTON

This routes the input channel signal to the left and right MONITOR outputs as well as the LEFT/RIGHT main outputs; both via the LEFT and RIGHT faders. This button fulfils three functions. The first, to provide a facility which permits direct 2-track recording (the input channel PAN control allows the input signal to be positioned anywhere in the stereo picture), the second to provide the ability to mix direct to stereo for live P.A. mixing when on the road and the third application is during a mixdown (explained in 4:19).

4:8 THE AUXILIARIES

The MIXDOWN consoles have 6 auxiliary sends, 8 or 16 AUXiliary LINE inputs and a stereo auxiliary return. They may be used in several different ways, and how they are finally employed is dependent on individual circumstances. AUX LINE INPUTS and the STEREO RETURN are explained in 4:12 and 4:13.

The most common use of auxiliaries 1 to 4 is for effects sends. (AUXS 5 & 6 and their normal applications are explained later). The master sends **(FIG 2: 17)** are outputs and connect to the inputs of effects units. The effects outputs can be connected to either a LINE jack on an input channel **(FIG 1: 2 or 3b)**, an AUX LINE input **(FIG 2: 2)**, or the stereo return **(FIG 2: 25)**.

A loop has now been created which is sending a group of signals to an effect (reverb, echo etc.), in which they are processed, and then returned into the MIXDOWN for recording.

Each input channel can have 4 effect sends. AUXS 1, 2 and 3 are constantly available and the fourth is selected by releasing the A4/A5 button to allow the fourth AUX control to be assigned to AUX 4. Turning these clockwise sends the channel's signal to the corresponding MASTER send controls **(FIG 2: 20)**, which determine the level of signal (which may have been derived from many input channels) that is sent to the effect.

The subgroups/tape monitors/AUX LINE INPUTS also have AUX 1 and 2 available. A button adjacent to the AUX 1/2 control (FIG 2: 9) selects either AUX 1 or 2. These allow submixes/tape returns/AUX LINE INPUTS to be sent to effects 1 or 2, just like input channels.

One further note about these 4 auxiliary sends: they are "post-fade". This means that their level is affected by the position of the input channel fader. On the subgroups/tape monitors/AUX LINE INPUTS, they are also post-fader, or post MONitor level control.

This example demonstrates why they are POST-fade:

> A mix may have been set, say from inputs 9 to 13, to subgroup 2. The AUX 1 master send has been connected to the input of a reverb, and the output of the reverb is connected to LINE input 16. Turning AUX 1 clockwise on each of the inputs 9 to 13 by the same amount means that the same relative mix levels that were sent to the subgroup by their input faders, are now at the AUX 1 master send (connected to the reverb input). Altering the faders on these input channels to alter the subgroup mix, will also alter the mix sent to the reverb.
>
> The signal FROM the reverb is "wet", and is connected to LINE input 16. "Wet" means with the effect added, as opposed to "dry" with no effect added. Depress the SOLO button on subgroup 2: in the headphones a "dry" mix of inputs 9 to 13 can be heard. Release this SOLO button. Depress the SOLO button on input 16: a "wet" reverb mix of inputs 9 to 13, with the same mix levels as the "dry" version, can be heard.
>
> The mix with reverb can now be routed (if required) from input 16 to subgroup 2 instead of the "dry" version from inputs 9 to 13.

AUXILIARIES 5 & 6

These auxiliaries are normally employed as "foldback" sends. Foldback is where one or more signals are routed to a totally separate Studio Headphone Monitor System or Stage Foldback System - it is the mix for the performers, all other monitoring so far described has been for the operator.

AUX 5 and 6 controls on the input channels are pre-fade and are therefore independent of any channel fader level. AUX 5 is also pre-EQ, which makes it ideal for stage foldback systems which have their own EQ. Like the effects sends (AUXS 1 to 4), the combination of levels on input channels are sent to the master send controls. On AUXS 5 and 6, these have an adjacent SOLO button which is post-level control so that the operator can examine the performers' mix for sound quality and content.

There is also an AUX 6 control on the monitor sections (FIG 2: 9) which is pre-MONitor level control when the monitor sections are being used for tape monitoring or AUX LINE INPUTS. When GROUP is selected, AUX 6 on the lower monitors is post-fader.

As the sends for AUX 6 may be derived from both input and subgroup channels, it is most versatile for foldback purposes when submixes are being used in a live application. Another useful feature of AUX 6, being available on AUX LINE INPUTS is that when effects returns are connected to the AUX LINE INPUTS, "wet" signals can be sent to the foldback system.

AUX 5 can be used to produce an alternative foldback mix for other performers, and can be connected to a foldback system which contains its own EQ facilities. To put effects returns onto AUX 5 foldback requires the effects returns to be connected to input channels.

WARNING: DO **NOT** return, say, AUX 4 into an input and turn up the AUX 4 control on that channel - the console will start to oscillate.

The master send jacks (output) of AUXS 5 & 6 should be connected to the inputs of the performers' Studio Headphone or Stage Foldback systems.

NOTE

Throughout this section it has been recommended that AUXS 1 to 4 are used for effects and AUXS 5 & 6 are used for foldback. This need not be the case, some engineers prefer pre-fade, and even pre-EQ, auxiliaries for effects purposes.

4:9 INPUT CHANNEL SEND/RET (FIG 1: 4)

The SEND/RET jack on each input is often referred to as a "break-jack" or "insert point". This socket accepts a 0.25" stereo phone jack and its function permits channel patching access. An input signal present in the channel may be routed out of the channel (via the SEND/RET jack) to an effects unit, perhaps, then back into the same input channel which may, in the normal manner, be routed onto the subgroups. The tip of the jack allows the signal out (to an effect etc.), the ring (the part immediately above the tip) returns the signal to the console and the sleeve (the longer part of the jack) is for the earth screen. The point at which the signal is diverted in and out of the channel is post-EQ and pre-fader. The EQ section and auxiliaries continue to work as normal. The returned signal is, like a normal input signal, ultimately controlled by the channel fader.

An example for using this facility would be to imagine that an input has a keyboard connected to it which is only playing some of the time. While it is not playing, a "hiss" is present. To stop this hiss reaching the subgroup stage, a noise gate (such as the Studiomaster IDP1 or Studiofex SF800) is connected to the SEND/RET jack. (Refer to Chapter 2 for wiring of jack plugs.)

Another example would be to use the facility for line-out purposes only (a pre-fade "listening-in" point), feeding a monitor amplifier, or using the output to trigger a keyboard or drum machine and so on.

4:10 THE DIRECT OUT (FIG 1: 5)

This mono 0.25" jack socket allows a "tap-off" point for the post-fade signal which can be used as an individual effects send, or as a post-fade trigger signal to the sidechain of an effect unit. It is preset at -10dBV nominal level, so the 16 input channels could go directly to a 16-track tape recorder, while the console is used for "live" sound mixing.

4:11 PHANTOM POWERING

The MIXDOWN has a +48V phantom powering facility. The phantom power buttons (which are slightly recessed to avoid accidental selection) on each input (FIG 1: 6) are best released (+48V switched OFF) when not in use. With the button depressed, 48V is applied to pins 2 and 3 of the MIC input XLR. This is intended for remote powering of condenser/ capacitor microphones, thus eliminating the need for external power supplies. A balanced microphone not requiring phantom power will not be damaged if 48V is ON (there are a few exceptions to this, and if in doubt contact the manufacturer of the microphone). An unbalanced microphone MUST NOT be used - so check before turning the +48V phantom power ON.

NOTE: If the MIXDOWN has an external power supply be sure that 48V phantom power is switched ON at the P.S.U. if its required.

4:12 THE AUX LINE INPUTS

These inputs make use of the redundant upper tape monitors when using the console for "live" work with subgroups. These inputs make use of both upper and lower tape monitors when the console is being used for "live" work without subgroups, or during the final 2-track mixdown.

THE UPPER AUX LINE INPUTS

The upper AUX LINE INPUTS (5 to 8 on 16.4.8, or 9 to 16 on 16.8.16) are selected by releasing the TAPE/AL INPUT button (FIG 2: 5). Their input jacks, (FIG 2: 2) may be left permanently connected. Once selected, the 2-band EQ (FIG 2: 4) comes into circuit. Depressing the FADER REVerse button (FIG 2: 6) transposes the upper MONitor level control with the subgroup fader below (the subgroup now using the upper MONitor level control instead of its fader). Depressing the METER SELECT button (FIG 2: 19) connects all subgroup bargraphs to the upper monitors. Each of these AUX LINE INPUTS now has:

1. a fader for level control
2. 2-band EQ
3. LED bargraph and SOLO monitoring
4. post-fade (or post-MONitor control) AUX 1 or 2 for effects sends
5. pre-fade (or pre-MONitor control) AUX 6 for studio/foldback monitoring
6. PAN control, routed directly to the LEFT and RIGHT outputs

In short, the upper monitor section has been converted to a basic input channel which can be used for aux returns or LINE inputs. This provides 4 extra inputs on a 16.4.8, or 8 extra inputs on a 16.8.16.

THE LOWER AUX LINE INPUTS

The lower AUX LINE INPUTS (1 to 4 on 16.4.8, or 1 to 8 on 16.8.16) are selected when a jack plug has been connected to their input (FIG 2: 2) and the TAPE/GROUP button has been released. The lower AUX LINE INPUTS are simplified versions of the upper ones which have:

1. level control by MONitor level
2. LED bargraph monitoring (with METER SELECT released)
3. post-MONitor control AUX 1 or 2 for effects sends
4. pre-MONitor control AUX 6 for studio/foldback monitoring

5 PAN control, routed directly to the LEFT and RIGHT outputs.

These lower AUX LINE INPUTS can be used in the same way as the upper ones giving a further 4 inputs on a 16.4.8, or 8 inputs on a 16.8.16.

NOTE: As the AUX LINE INPUTS have only a level control, and not a GAIN control, a signal above or below their nominal sensitivity must be adjusted at source ie adjust the output level of the source. The sensitivity is Factory Set at -10dBV, with an option to change to +4dBm. The nominal level of the AUX LINE INPUTS is changed when the TAPE monitor levels are set.

USING THE AUX LINE INPUTS

With a total of 8 (16.4.8) or 16 (16.8.16) extra line inputs, the console is of great use in MIDI based studios. In this application, the multi-track tape recorder is used only for vocals, guitars etc., and the keyboards, drum machines etc. (controlled by a MIDI sequencer) are added at mixdown - straight onto 2-track. The MIXDOWN console provides 16 input channels, 8 or 16 AUX LINE INPUTS and a stereo return (explained in 4:9), making it ideal for this type of recording.

4:13 THE STEREO RETURN

The stereo return is ideal as a return from a stereo effect (such as an auto-panner), or as a line level input for, say, a CD player or cassette tape player. By using the LEFT input of the return, it can be used as a mono line input for normal effects, or as a further AUX LINE input. It is bussed directly to the LEFT and RIGHT outputs.

4:14 TALKBACK

A talkback microphone XLR is provided on the front panel of the MIXDOWN console. Into this a gooseneck studio microphone can be connected. This allows the operator to communicate with the performers through the Studio Headphone Monitor System or Stage Foldback System. It has its own level control and push-to-talk ("TO AUX 6") button. The "TO AUX 6" button does not latch, and while it is depressed, the operator can talk to the performers through AUX 6. On its release, the button "mutes" the microphone signal.

4:15 THE MASTER SECTION

The master output section can be used in two ways:

a The first is as a 2-track recording output. The LEFT and RIGHT XLR sockets (FIG 2: 14) are connected to the inputs of the 2-track tape recorder. It must be understood, however, that output from these XLR sockets is at a high level (+4dBm balanced) and may therefore be unsuitable for the 2-track tape recorder. Should it require a lower level, then the 2-track output jack sockets (FIG 2: 15) provide a -10dBV level (many tape recorders are set for this level). Bear this in mind when making the connections, and check the tape recorder manual if in doubt. If neither +4dBm balanced or -10dBV is suitable, the option of changing the XLR outputs to +4dBm unbalanced is available (refer to Chapter 6).

b The second is as a P.A. output. The LEFT and RIGHT +4dBm balanced XLR sockets are connected to the inputs of a P.A. system. In a live performance, no amplifier is needed at the monitor output (the operator is at the back of the hall, and can hear the performance as the audience does). The operator monitors with the headphones, and uses SOLO buttons throughout the console for individual signal examination.

The master LEFT and RIGHT faders control the signal level to the tape recorder as they do to a P.A. amplification system. The LEFT and RIGHT outputs have SEND/RET jacks **(FIG 2: 16)** for a graphic equaliser to be connected in line. The sound going into the equalisers should have been perfected during the mixing, leaving the equalisers to make up for room acoustics, the presence of a crowd or loudspeaker system peculiarities. The SOLO LEFT and SOLO RIGHT buttons **(FIG 2: 21)** allow the output signals to be checked before the graphic equalisers (it can be heard post-equalisers through the headphones when the SOLO buttons are released).

4:16 THE 2-TRACK RETURN

The 2-TRACK RETURN sockets **(FIG 2: 24)** route the signal to the 2-TRACK RETURN button **(FIG 2: 31)** which, when depressed routes the signal to the MONITOR and PHONES level controls **(FIG 2: 32)**.

NOTE: When depressed, all other monitor or SOLO functions are overidden, so make sure that when it is not in use, the button is released.

4:17 TAPE MONITORING PROCEDURE

The MIXDOWN consoles while being used for studio work, will need to do "off-tape" monitoring. MIXDOWN consoles can monitor more tracks than they can record. The 16.4.8 has 4 subgroups, 8 tape outputs jacks (which share the 4 subgroups: TAPE OUTPUTS 1 and 5 share subgroup 1 etc.), 8 dedicated tape inputs, and 8 tape monitor/AUX LINE INPUT sections. The 16.8.16 has 8 subgroups, 16 tape output jacks (which share the 8 subgroups: TAPE OUTPUTS 1 and 9 share subgroup 1 etc.), 16 dedicated tape inputs, and 16 tape monitor/AUX LINE INPUT sections.

During tape monitoring, subgroup outputs (and hence their controls) are not required, so on each subgroup select TAPE on the GROUP/TAPE button **(FIG 2: 11)**. Also, during tape monitoring, AUX LINE INPUTS are not required so select TAPE on the AL INPUT/TAPE button **(FIG 2: 5)**. During monitoring, AUX LINE INPUTS (if used) 1 to 4 (on 16.4.8) or 1 to 8 (on 16.8.16) must be disconnected. These buttons have now routed the tape input to its respective tape monitor section (TAPE 1 to MON 1 etc...). Its playback level can be seen on bargraph 1, and it can be sent to the headphones/monitors by turning clockwise the MON 1 level control, its stereo position being determined by the PAN control. If, say 8 tracks with a 16.4.8, or 16 tracks with a 16.8.16, have been recorded, then to view the extra playback tracks (5 to 8 or 9 to 16 respectively), depressing the METER SELECT button **(FIG 2: 19)** changes the bargraphs over to display them, rather than tracks 1 to 4 and 1 to 8 respectively.

EXAMPLE (USING A 16.8.16): Drums, bass, rhythm and lead guitar have been previously recorded. These occupy tracks 1, 2 and 3, leaving vocals to go onto track 4. Vocals need to be recorded while the vocalist is listening to the previously recorded tracks as backing. The vocalist must be separated totally from any sound other than his/her own voice, and yet he/she needs to hear the backing tracks. This is achieved by the use of the performers Studio Cue headphones which are fed via the AUX 6 (foldback) controls and a headphone amplifier.

To set up the vocalist's monitor mix. To monitor the three previously recorded tracks, depress the GROUP/TAPE buttons on subgroups 1, 2 and 3 (selecting TAPE). Turn clockwise AUX 6 on these three monitors and on the input channel used for the vocals (so the vocalist can monitor his/her own voice) and finally the overall mix level to the Studio Cue headphones is adjusted by the AUX 6 master control.

To set up the operator's monitor mix. For the operator to monitor the three previously recorded tracks, turn clockwise the MONitor controls on subgroups 1 to 3 (with TAPE already selected). For the operator to listen to the vocalist being recorded, GROUP must be selected on subgroup 4 (to allow recording to take place), then the MONitor control on subgroup 4 turned clockwise to bring it into the headphones/monitor mix.

There are now two separate mixes of the played back tracks. The reason for allowing a different mix for a performer and the operator is that the operator may wish to listen to a more balanced mix of the tracks, whereas the vocalist may want to hear mostly rhythm guitar and not much of the drums, for instance, to sing along to.

4:18 TAPE REMIXING PROCEDURE

Re-mixing (ping-ponging, throwovers etc.) is a process whereby previously recorded backing instruments, for example, need to be mixed together onto one or maybe two tracks. This releases tracks for further recording.

EXAMPLE (USING A 16.4.8 WITH A 4-TRACK TAPE RECORDER): Essentially, the previously recorded tracks, say 1, 2 and 3, are passed through the input channels 1, 2 and 3 respectively, by depressing the TAPE button **(FIG 1: 7a)**. The EQ section and effects can be used if "colouration" of the original track is needed. Finally, they can be mixed onto the remaining track (track 4) by routing and panning as described in 4:3.

Selecting TAPE on any GROUP/TAPE or AL INPUT/TAPE button on the subgroups during remixing does not in any way affect the signal while it is being processed in the input channel. It simply allows the monitor system to hear the track selected, **direct** from the tape. When GROUP is selected on a GROUP/TAPE, the monitor system hears the recording channel's TAPE OUTPUT. In this example, if GROUP is selected on all 4 subgroups, nothing will be heard from TAPE OUTPUTS 1 to 3, but 4 has the remix signal - a mix of the original tracks 1 to 3.

Due to the limited number of tracks on a 4-track recorder, remixes will probably have to be done more often than if an 8-track recorder were used. Remixes should be kept to a minimum. however, as every time the signal passes through a piece of equipment, noise is inevitably added to it.

4:19 2-TRACK MIXDOWN PROCEDURE

This procedure is for putting the multitrack tape recording onto a stereo master tape, or cassette tape, for listening to on normal audio equipment. It is achieved as follows:

> The 2-track tape recorder is connected to the LEFT and RIGHT outputs. Multitrack mixdown to these outputs can be achieved from the tape MONitor and PAN controls, or through input channels (or a combination of both). Through input channels allows addition of effects and EQ if desired. To achieve this, select TAPE on each input channel and route all inputs to L-R and PAN as required to achieve the desired stereo picture. Direct mixdown is achieved by depressing all GROUP/TAPE and AL INPUT/TAPE buttons (to select TAPE) on the subgroups, and setting the mix by adjusting the MONitor and PAN controls on the subgroups, which are bussed to the master output faders.
>
> This is a stage at which the AUX LINE INPUTS become of most use. Input channels are all used for multitrack returns (the multitrack tape recorder having been used only for vocals, guitars and other "acoustic" instruments). Electronic instruments like keyboards and drum machines - which need little or no EQ or effect treatment - are controlled by a sequencer and are connected to AUX LINE INPUTS. They are then recorded straight to the 2-track tape recorder (level and stereo image set by MONitor and PAN controls). This type of setup is what the MIXDOWN consoles were primarily designed for, as it is typical of the growing number of MIDI based studios.

Like remixes, sound quality may be adversely affected during the mixdown procedure. Successively adding EQ through remixes will induce noise and may even cause distortion. Careful thought must always be used when adding EQ to the original track. Bear in mind also noise induced by the tape recorder and effects units. To achieve the best results, keep levels throughout the console as near as possible to the optimum recording level during every process. It is therefore obvious that the first tracks must be of the highest quality in order that subsequent processes can be of a similar standard.

5 Step by step use of Mixdown

This chapter assumes connections have been made as explained in Chapter 3, and also that Chapter 4 has been read and understood.

5:1 MULTITRACK TAPE RECORDING

The recording process may be broken down into 5 main stages:

1 The first process is to **PREPARE** the console for recording.

2 **INITIAL MULTITRACK TAPE RECORDING.** This is recording direct from microphones and line inputs onto the tape recorder.

3 **MULTITRACK PLAYBACK.** Listening to what has been recorded.

4 **OVERDUBBING.** Building up the tracks by remixing, adding tracks and listening to what has already been recorded.

5 **THE STEREO MIXDOWN.** Combining all recorded tracks, effects, etc. to form the final stereo recording.

5:2 PREPARING THE CONSOLE

Before beginning the recording session, compatibility between different components of the system - and also a reference level from which adjustments can be made - must be achieved.

CALIBRATING THE MIXDOWN TO A MULTITRACK TAPE RECORDER

1 By using the console's built in 1kHz SLATE oscillator, the subgroup faders may be calibrated to give the optimum recording level on the multitrack tape recorder.

2 The TAPE outputs of the MIXDOWN are factory set at -10dBV. Check in the handbook of the tape recorder that it is compatible with this level. If it is not, then the console may have its outputs and return levels changed from -10dBV to +4dBm by simply re-positioning links on its printed circuit boards. Chapter 6 explains this procedure. We recommend that a Studiomaster dealer performs the modification as the front panel must be removed and this will make the warranty void if an inexperienced person does it.

3 Depress the 1kHz SLATE button on the console **(FIG 2: 28).** Bring up all the subgroup faders until a zero level is registered on both the console's and the tape recorder's meters. If there is a serious difference between the meter readings of the console and the tape recorder (ie it is impossible to get both of them to read zero at the same time) then this suggests the output level links are incorrectly positioned. Refer to Chapter 6 for adjustment.

4 The playback level should also be checked. This is done by recording a section of the 1kHz tone. Then, by depressing the TAPE buttons on the input channels (with TAPE recorder outputs connected) and playing back the recorded tone, compare the levels of MIXDOWN and tape recorder meters again. A serious difference between readings once again indicates that links may be incorrectly positioned - refer to Chapter 6.

5 The subgroup faders are now at their optimum position for recording and should remain here as an initial reference position.

SETTING UP THE INPUT CHANNELS

1 Firstly, set all the input channel faders to the 0 position (fully down), all auxiliary send controls to 0, make sure all buttons with the exception of EQ CUT are OUT, and make sure the PAN controls are at their centre (0) position.

2 Select whether the input is to be MIC or LINE by releasing or depressing the LINE button (16.4.8 only). On a 16.8.16, only connect to a MIC or LINE input, never to both on the same channel, as this can cause damage to the console. If +48V phantom power is required, depress the +48V button on the input channel. If the MIXDOWN is an external power supply model, make sure +48V has been switched on at the power supply. Phantom Power is explained in 4:11.

3 To set up a sound, use the monitor system by routing to the LEFT & RIGHT (L-R) outputs, so that the sound can be heard at the PHONES or MONITOR outputs and viewed on the LEFT & RIGHT bargraphs. As several inputs are set up, the operator may wish to release the L-R button on certain channels to isolate individual sounds. Alternatively, several inputs may remain routed to the LEFT & RIGHT outputs and individual monitoring achieved by depressing SOLO buttons (explained in 4:5), then the SOLO level would be viewed on the RIGHT bargraph and the SOLO system would override the main monitor mix in the MONITOR or PHONES outputs.

4 Adjust the input GAIN control so that on peaks, a good registration is attained on the SOLO bargraph (illuminating the first red LED, while at the same time there is no indication on the channel's CLIP LED). The aim is to get the GAIN as high as possible on peaks, so that level boosting by faders later on in the recording process (which will encourage noise) is unnecessary. If minimum GAIN is still too high to achieve this, then depress the PAD button to reduce the incoming signal level.

5 If the EQ requires adjustment, release the EQ CUT button to bring the EQ into circuit. While listening in the headphones, adjust the controls to give the tonal quality desired. Careful manipulation of the controls is essential and it is good practise to EQ CUT in and out to compare adjustments against a "flat" response. While adjusting EQ (particularly when boosting), watch the bargraphs and CLIP indicator. GAIN, or an offending EQ control should be re-adjusted if the level rises too high.

6 Repeat this for all the inputs. During recording, effects will be added and adjustments will probably be necessary, but these settings provide a good basis.

SETTING UP THE AUXILIARIES

1 Assuming that the actual connections between console and effects have been made (as in Chapter 3), the auxiliaries can be adjusted to put effects on the input signals and produce a studio cue mix.

2 On inputs that require effect 1, turn clockwise the AUX 1

controls by varying amounts to create the required mix at the master send control which is turned clockwise until a good signal is present at the effect. The same can be repeated for the other effects.

3 The effects may be returned into the console at either AUX LINE INPUTS or LINE input channels. As explained in 4:8, the best place to return an auxiliary is a LINE input channel. However, these may all be needed for TAPE inputs at a later stage, so AUX LINE INPUTS are the next best. The "upper" AUX LINE INPUTS have SOLO buttons and may be used throughout the multitrack tape recording to monitor the "wet" signals while recording is taking place.

4 Auxiliaries 5 & 6 are for studio cue - so the performers can hear themselves. AUX 5 can only be used on input channels not already using AUX 4 for effects. AUX 5 is pre-EQ on the input channels and is for connection to performer monitor systems which have their own EQ etc.. Like the effects sends, turn clockwise the AUX 5 and/or AUX 6 controls on the inputs the performers want to hear by different amounts to create a mix at the master send controls. This can be checked by the operator by depressing the SOLO buttons adjacent to the master send controls.

5 The performers may also be allowed to hear themselves "wet", by turning clockwise AUX 6 on the AUX LINE INPUT or LINE input channel that the effect is returned to. Make sure that if, say, AUX 1 is returned to a channel that the AUX 1 control on that channel is turned to 0.

6 At this stage, effects should not be recorded, so do not route effects returns to any of the subgroups.

7 It is also a good idea at this stage to check that the talkback system works. Connect a microphone to the TALKBACK input XLR. The volume of the operator's voice sent to the performers' studio cue mix (AUX 6) is controlled by the TALKBACK level control, and is bussed onto AUX 6 while the "TO AUX 6" button is held down.

5:3 INITIAL MULTITRACK TAPE RECORDING

This is the first recording stage. It cannot be emphasised enough how important it is that input levels (determined by GAIN controls) are optimal at this stage. If further processes are to be of a high standard, then this stage is critical.

SETTING UP

Route the inputs to the desired subgroups. Each input may be routed to one or more subgroups. Remember to keep one track (usually the last one: 4, 8 or 16) free for remixes later. This is especially important when working with 4-track tape where the limited number of tracks means that remixes (as explained in 4:18) are virtually inevitable. A track may also be required for recording a MIDI sync. signal onto for triggering MIDI synthesisers, drum machines and samplers (the use of this is explained in 5:9).

Raise the input channel faders to create the desired mix of signals at the subgroups. Watch the bargraphs on the subgroups and check that the level on peaks does not pass the first red LED. If it does, then reduce input channel fader levels. The subgroup bargraph levels should not pass the first red LED on peaks with the input channels routed to them anyway, as a nominal fader level has already been set using the 1kHz SLATE oscillator. If they do light more than the red LED, then check that none of the input channels are clipping, slightly reduce input fader levels and slightly reduce the subgroup fader in question. If the subgroup overload is due to an input clipping, then adjust that input's GAIN control before moving any faders.

Monitoring generally should be done through the main LEFT & RIGHT monitors, leaving the SOLO function for individual channel checking. Remember that the LEFT & RIGHT monitor mix can either be derived from the L-R buttons on the inputs, or from the monitor section on the subgroups.

For future line-up, a section of the 1kHz tone should be recorded at the beginning of the tape (the procedure is described in 5:2). This piece of recorded tone is made use of in 5:6.

FADER LEVELS

The excellent audio specifications of the MIXDOWN consoles are measured with optimum control settings. To make use of the console's capabilities, take care when adjusting GAIN controls and faders at every stage.

The subgroup faders should be set around the 0dB mark, and the input faders ideally around No.7. From these positions, the signal level from the channel can be increased, and conversely, the channel may be smoothly faded out. By operating the faders at these positions, the resolution of the faders is such that fine adjustments can be made accurately.

If however the input fader is as low as No.3, then the pre-fade headroom is reduced so far that to attain any reasonable volume from the channel the GAIN is turned up excessively (which will cause the channel to overload and distort). Another example of poorly positioned faders is if an input routed to a subgroup has its fader level below No.7 while the subgroup fader is above 0dB. This will adversely affect the signal-to-noise ratio as the unduly attenuated input signal is being compensated for by a high subgroup fader level. It is far better to run the subgroup fader at 0dB, and make up gain at the input fader.

READY TO RECORD

Having decided routing, set the track(s) on the tape recorder which are to be recorded to RECORD-PAUSE. Count the performers in and release the PAUSE button(s) to commence recording. Remember to check subgroup levels which clip as explained above throughout the recording. Fader levels on both subgroups and inputs may well need adjusting throughout the performance. The operator's ears are the final judge of the "right mix", so attention must be paid to the monitor system, ensuring that the guitarist is not drowning out the vocalist etc..

USING THE AUXILIARIES DURING RECORDING

Some performers like to hear themselves playing with effects added, (exactly as they will ultimately be recorded at the stereo mixdown). This is especially true for vocalists who like Ambience added, or guitarists who need Delay to produce a rhythmic sound. However, it is often more desirable not to record the effect onto multitrack tape as this limits flexibility at the final stereo mixdown stage. To allow the performers to hear effects, the effects must be returned to the MIXDOWN at either a LINE input channel or AUX LINE INPUT in such a way as to allow the effect to become part of the AUX 5 and/or AUX 6 performer mix(es), but not routed to any subgroup being recorded, ie turn clockwise AUX 5 and/or AUX 6 controls, but leave subgroup routing buttons released.

5:4 PLAYING BACK THE MULTITRACK TAPE RECORDING

Now that the first tracks have been recorded, it is necessary to listen to what has been produced. This is achieved by depressing the GROUP/TAPE or AL INPUT/TAPE buttons on the subgroups in question (assuming that multitrack tape recorder outputs are already connected to corresponding TAPE inputs).

This routes the playback into the monitor section on the subgroups. By turning clockwise the MONitor control, this signal is sent to the LEFT & RIGHT monitors, the PAN control being used to position the signal in the stereo picture between LEFT & RIGHT. Turning clockwise the AUX 6 controls on the subgroups will route the playback to the performers.

Another use for AUX 6, is that in later stages when more tracks are being recorded, previous tracks are being played back so the performers can synchronise themselves. The sound they need to hear is sent to them via AUX 6 on the monitor sections. This is explained in the following section.

5:5 OVERDUBBING

Overdubbing is a process of editing the multitrack tape recording by re-recording portions of tracks and correcting mistakes made in the first "take".

Tracks that are already on tape are listened to as explained in 5:4. Set up monitor mixes to the LEFT & RIGHT outputs using MONitor and PAN on the subgroups and a mix to the performers via AUX 6.

The performer's AUX 6 mix can be derived from two sources; the input channels or the subgroups.

1 **From input channels only:** the musician will always hear himself, but will be unable to hear what he has already recorded, which he would need to do, in order to "drop in" to his original performance during the overdub.

2 **From subgroup channel monitors only:** when the studio cue is derived from these, the operator decides whether the performer hears himself "live", or the previous recording. This is done by either routing the playback to AUX 6 on the monitor section (depressing TAPE on the performer's input channel), or by just leaving the performer routed to the subgroup without TAPE selected. He will not be able to hear both which would not be a problem for most instruments, but a vocalist may be unable to make his voice match the previous recording under these conditions.

The switching between sources, however, can be achieved automatically by means of an in-line noise reduction unit or certain types of tape recorder which, when switched to STOP, FF, REW or RECORD switch the sync. output to line input and which only turn to sync. playback when in the PLAY mode. With this type of system, all the operator needs to do is set a mix from both input and subgroup auxiliaries and depress TAPE on the input channel in question.

The musician will now hear himself live all the time, until the recorder goes into PLAY mode, in which case he will also hear his recorded self to sing/play to.

3 **From both:** in this case the musician is hearing both himself live and his previously recorded performance (only while TAPE is selected) up to the moment of recording again, where most tape recorders will switch automatically from sync. playback to line input. This will result in a slight level increase to the cue mix as the GROUP/TAPE button is released and the subgroup AUX 6 signal is added to the input channel AUX 6 signal (effectively the same signal), instead of the tape monitor AUX 6 signal being mixed with the input channel AUX 6 signal.

5:6 TAPE REMIX

This process (already outlined in 4:18) combines previously recorded tracks onto two tracks as stereo or one, as a "bouncedown" - and will apply most to users recording on 4-track tape with 16.4.8 consoles where the limited number of tracks makes remixes inevitable.

Depress the TAPE button on all the input channels which have multitrack outputs connected to them and which need to be remixed onto one subgroup. Depress routing buttons and PAN exclusively if necessary to the spare subgroup. (At least one track should have been left after the initial multitrack tape recording).

This process can introduce noise if gain or equalisation is added and subsequently upset the re-recorded sound quality. If extra gain or EQ is required, then this factor should be considered. Take care.

Use of GAIN at this stage can largely be avoided if all levels were set correctly during the early stages of recording.

Every recording process and piece of equipment which the signal has to pass through will add noise to the original signal, so remixes which have to pass through the console and the tape recorder more than twice should be kept to a minimum. The input channel during remix can, however, be set in such a way that only minimal gain is added.

HOW TO SET THE INPUT CHANNEL FOR MINIMUM NOISE ON REMIX

1 If no EQ is required, depress the EQ CUT button.

2 Select the remix track, and route to it.

3 Set the channel fader to the No.7 position.

4 In 5:3, it was advised to record some of the SLATE tone at the beginning of the tape. This is where it is used. Play this into the input, depress SOLO and while watching the SOLO bargraph

adjust the input channel GAIN until 0dB is indicated.

5 Release the SOLO button, and FF the tape recorder past the tone. As a guide, the GAIN control should read about 0dB on its outer calibration. This setting will give 0dB gain for remix. **NOTE:** The 0dB calibration on the outer scale refers to LINE and TAPE inputs, and is equivalent to 30dB on the MIC input.

RECORDING

Recording can now begin, and the original tracks will be reduced onto one only. The next step, is to go back to the "Initial Multitrack tape recording" procedure (5:3), and record further "live" tracks, adjacent to this remixed track. The performers studio cue to synchronise themselves to, will now come from the AUX 6 control on the remixed track, together with any "live" sounds from other AUX 6 controls.

5:7 STEREO MIXDOWN - THE FINAL MIX

Like remixing, this procedure was explained earlier (in 4:19) so its purpose should basically be understood.

Once all recordings, overdubs and remixes are completed, it is time to produce the stereo master mix. This process combines all previously recorded tracks with any effects such as flangers, harmonisers and reverbs that are required.

This procedure is very similar to the remix procedure, but effects and equalisation can be added to perfect the mix.

SETTING UP

1 Depress the TAPE buttons on all the input channels with multitrack tape recorder outputs connected to them.

2 Set channel GAIN controls using the line-up tone as described in 5:6. Set all the remix inputs to this "no gain, no EQ" status as a starting point (later on, EQ may be adjusted if necessary).

3 Route these input channels to L-R. The PAN control on the channels can position these inputs in the final stereo picture.

4 For returning auxiliaries, there is a stereo return (**FIG 2: 25**) which for some applications will be adequate for the incoming signal. In most cases further processing of the auxiliary returns will be required. In such cases, the returning signal should be returned to a LINE input channel or an AUX LINE INPUT.

NOTE: If an auxiliary (say AUX 1) is returned into an input channel, then make sure that the AUX 1 control on the channel is turned to 0 otherwise a loop will occur and cause the console to oscillate.

5 Limiters and noise gates can be connected to the SEND/RET points on the input channels for individual track processing.

6 Before finally going to tape, the final mix of tracks from the tape recorder and auxiliary returns can be checked through the monitor system: Set the LEFT & RIGHT faders at 0dB. Play back the

multitrack tape recorder and adjust any effects, EQ settings and levels. Perform a few of these "dry runs" until the "PERFECT MIX" has been achieved.

8 Once all signal levels have been set, recording may commence.

THE STEREO RECORDING

The 2-track tape recorder should already be connected to the LEFT & RIGHT console outputs (which outputs to use - they have different levels - is discussed in 4:15). It is imperative that the correct outputs are connected to the 2-track recorder so that the console and tape recorder are compatible. Also, the tape recorder's outputs need to be connected to the 2-TRACK RETURN jacks **(FIG 2: 24)**.

It is critical that the levels of the LEFT & RIGHT outputs are set correctly. The output levels are, of course, dependent on their fader positions. The faders can be set for the optimum results during the "dry run" procedure explained earlier. The first red LED on the LEFT & RIGHT bargraphs lighting on peaks is the correct level. Alternatively, the multitrack tape may be rewound and the piece of 1kHz tone played back through the system to set the faders. The faders should end up reading about 0dB on their calibration with the bargraph lighting correctly.

If the faders have to be above 0dB, then the overall mix level is too low. In this case, raise by a small amount all the input channel faders. The LEFT & RIGHT output faders should not be used to add gain to the recording signal. **ALWAYS** try to make up any level deficiency with the input channel faders.

When recording commences, use SOLO buttons throughout the console to monitor individual signals at remix input channels, upper AUX LINE INPUTS and auxiliary master sends. To check the recording itself, depress the 2 TRACK RETURN button. This overrides all other signals at the monitors, and routes the 2-track returns into the monitor system.

Check constantly that the LEFT & RIGHT bargraphs never light more than the first red LED on peaks.

5:8 AN ALTERNATIVE MIXDOWN METHOD

The mixdown method outlined above allows gain, EQ etc. to be added to the remix signals. There is another method which allows no gain or equalisation to be added and subsequently it adds no noise:

1 Release the TAPE buttons on all input channels with outputs from the multitrack tape recorder connected to them.

2 Depress the GROUP/TAPE buttons on the subgroups, so that instead of being fed through the input channels, the remix signals go to the subgroup monitor section.

3 From here, the MONitor levels can be turned clockwise and the PAN controls used to create the stereo effect.

4 Effects may be added via the AUX1/AUX 2 control, and also the AUX 6 control as a foldback send is no longer required. These effects may be returned through the STEREO RETURN or a LINE input channel (remember that the input channel can add gain and EQ).

5 SOLOing can be done at the subgroup SOLO buttons and the SOLO buttons of any input channels used for returning effects. The recorded mix can be heard by depressing the 2-TRACK RETURN button, which overrides **ALL** other signals at the MONITOR and PHONES outputs.

6 If it is found that the LEFT & RIGHT bargraphs are reading correctly at peaks, but the LEFT & RIGHT faders are above 0dB, then the overall mix level is too low. Turn up all the MONitor controls a little to rectify this (the MONitor controls act like the input channel faders in 5:7).

7 Constantly check the bargraphs and make sure clipping does not occur.

Using this method on its own is probably quite unlikely, but by using it in conjunction with the other mixdown method (5:7), remix signals which are perfect and need no further processing may be brought to the LEFT & RIGHT outputs by this method while others may be brought via the input channel method.

5:9 AUX LINE MIXDOWN

This type of mixdown is where the Studiomaster MIXDOWN consoles are superior to similar types of mixing console. In MIDI based systems, where some of the tracks to be mixed down come direct from synthesisers, drum machines or samplers, as opposed to being recorded onto multitrack tape, the AUX LINE INPUTS can be used as extra input channels for the mixdown process. A total of 34 tracks could be mixed down on a 16.8.16. At mixdown in this system, the 16 tracks would be played back (15 into the console's TAPE inputs, 1 into the MIDI sequencer), then 17 MIDI instrument outputs and effects returns can be played into the 16 AUX LINE INPUTS and the one remaining LINE input channel, and a stereo effect returned into the STEREO RETURN.

5:10 LIVE PERFORMANCES

The procedure for using the MIXDOWN for live sound systems:

1 The first process is to **PREPARE** the controls on the console to a "working level", and check compatibility with the P.A. system.

2 A **SOUNDCHECK** must then be performed to make adjustments to the original settings in order that the audience will hear a good mix and also that the performers have good foldback sound.

3 **THE PERFORMANCE.**

5:11 PREPARING THE CONSOLE

Before beginning, make sure that the MIXDOWN is compatible with the amplifier system, and set all the controls to a "working level".

CHECKING COMPATIBILITY

Amplifiers usually have a maximum input level of +4dBm. This can be taken from the LEFT & RIGHT outputs of the MIXDOWN console via the male XLRs. These outputs are Factory Set UNBALANCED, but it is likely that the amplifier system will require them BALANCED, and the MIXDOWN consoles have an option for this (explained in Chapter 6). The XLRs are wired Pin 2 - Out-phase, Pin 3 - In-phase, Pin 1 - Ground for balanced use, and Pin 3 - Signal, Pin 1 - Ground for unbalanced use.

SETTING UP

1 Firstly, set all the input channel faders to the 0 position (fully down), all auxiliary send controls to 0, make sure all buttons with the exception of EQ CUT are OUT, and make sure the PAN controls are at the centre position (0).

2 The input may be MIC or LINE. +48V phantom power for condenser microphones may be applied to the MIC inputs by means of a button on each input channel. This button is recessed to avoid accidental selection, as phantom power can damage some microphones (refer to 4:11). For the purposes of setting up, a tape recorder can be connected to a LINE input.

NOTE: Direct Injection (D.I.) boxes may be used for LINE level instruments, which convert the line level signal to balanced MIC level (and sometimes unbalanced MIC level) for connection by XLRs to the MIC inputs. This would almost certainly be the case if the optional Studiomaster multicore kit is fitted to the console. Some D.I. boxes will require the phantom power to be ON, others will not.

3 To set up a sound, depress the SOLO button on the input channel with the tape recorder connected to it. Turn clockwise the GAIN control until, on peaks, the CLIP LED is not lit and the first red LED on the SOLO bargraph is lit. Listen to the signal through the headphones. EQ may be adjusted if necessary. Release the EQ CUT button and make the necessary adjustments. Make sure, on the SOLO bargraph and the CLIP indicator, that EQ adjustments do not affect the channel level. Correct the GAIN, or the offending EQ control, as necessary. Release the SOLO button and route the input channel to L-R.

4 To hear the sound through the P.A., and obtain an initial volume setting for the system, raise the input channel fader to approximately No.7 and raise the LEFT & RIGHT master faders until the desired volume is achieved. Aim to keep the input channel fader around No.7, and make system volume adjustments with the LEFT & RIGHT faders.

5 At this point, the musicians will need to perform in order to mix the different channel sound levels and adjust their EQ settings. If it is found difficult to balance the sound because of feedback or distortion, it can be useful to use headphones and SOLO buttons to listen to individual channels in order to examine the sound quality.

THE AUXILIARIES

MIXDOWN consoles have 6 auxiliary sends. 4 of these are post-fade (for effects), 1 is pre-fade and 1 is pre-EQ (the latter two are for foldback). AUX 5, which is pre-EQ, can only be used on input channels not using the post-fade AUX 4 for effects. As it is pre-EQ, it is ideal in a live system where an on-stage foldback amplifier may have additional EQ facilities. Such an amplifier system would require an "unequalised" signal as an input, which the AUX 5 master send provides.

EFFECTS

Adding effects is done in the same way as for multitrack tape recording. Connect effects inputs to the master send jacks 1 to 4. If AUX 5 is to be used for foldback, depress the A4 POST/A5 PRE EQ button on inputs requiring AUX 5 instead of AUX 4. If, ultimately, all input channels are using AUX 5, then no effect need be connected to the AUX 4 master send.

As an example, AUX 1 is connected to, say, a reverb. On all inputs which reverb is required on, turn clockwise the AUX 1 control. The different AUX 1 settings will provide a mix at the master send control, this is then turned clockwise to send this mix to the reverb. To return this "wet" signal to the LEFT & RIGHT outputs, the output of the reverb may be connected to either a LINE input channel, an AUX LINE INPUT or the STEREO RETURN. The STEREO RETURN and AUX LINE INPUTS are connected direct to the LEFT & RIGHT outputs. Turn clockwise the STEREO RETURN LEVEL control or the MONitor control to put the reverb onto the LEFT & RIGHT outputs. If the reverb is returned to a LINE input channel, then depress that channel's L-R button and raise the fader.

FOLDBACK

AUX 6 is intended for foldback use, and during recording, one foldback send to the performers is normally sufficient. However in a live performance, with ALL performers playing together, another foldback mix may be needed. For this, depress the A4 POST/A5 PRE EQ button above the dual purpose AUX control on all the input channels. AUX 5 is now a pre-fade, pre-EQ foldback send.

Connect the master send jacks of AUX 5 and AUX 6 to the inputs of two on-stage monitor systems. Different foldback mixes can be made by turning clockwise various AUX 5 and AUX 6 controls on the input channels. AUX 6 controls on subgroup channels which are being used for submixing (see next section on "**SUBMIXING**"), can be turned clockwise also to contribute to the overall AUX 6 foldback mix. Adjusting the AUX 6 control on a subgroup mix of, say, all the drums is easier than adjusting AUX 6 controls on all drum microphone input channels.

The operator can listen to AUX 5 and AUX 6 foldback mixes by depressing the SOLO button adjacent to their master send controls. The performers on stage will also let the operator know when the foldback mix(es) is/are correct for them.

Now check that the TALKBACK system works correctly. Connect an unbalanced microphone to the TALKBACK input XLR. While the TO AUX 6 button is depressed, the operator's voice will be sent to performers on stage via the AUX 6 foldback system. The level of the operator's voice is determined by the TALKBACK LEVEL control.

Effects can be added to the foldback mix. Effects returned to LINE input channels and AUX LINE INPUTS can be sent to foldback(s) by turning clockwise the AUX 5 and AUX 6 controls on these channels.

NOTE: If an auxiliary (say AUX 1) is returned into an input channel, then make sure that the AUX 1 control on the channel is turned to 0, otherwise a loop will occur and this will cause the console to oscillate.

THE SUBGROUP CHANNELS: SUBMIXING AND SESSION RECORDING

MIXDOWN mixing consoles have either 4 or 8 subgroup channels which can be used for either submixing or making a recording of the live performance. Session recording will apply mainly to 16.8.16 operators, as more subgroups will be spare.

SUBMIXING

This is usually done when several microphones are used on one group of instruments (such as a drum kit), or where several keyboards are being used together. It allows an overall mix level of the instruments to be adjusted by one fader (their relative mix to each other being set with their input channel faders).

To make the submix, route all the drum input channels to one subgroup using their channel routing and PAN controls. Set their faders at the various levels to give the desired mix. This mix of signals is now at the subgroup fader and its monitor section. To send this on to the LEFT & RIGHT outputs, turn clockwise the MONitor control and use the PAN to position the submix in the stereo picture.

SESSION RECORDING

This is when the live performance is recorded at the mixing console while it is being performed. Once onto multitrack tape, it can be mixed down to stereo at a later time. Such a recording may be done from the subgroup outputs, or from the DIRECT OUTS on input channels, the subgroup method (as explained in 5:1 to 5:3) being by far the best.

This live recording will not record the crowd's reactions (though the vocalist's microphone may pick up a little). A hyper-cardioid microphone could be used (which picks up ONLY sounds directly in front of it) by pointing straight at the middle of the crowd (NOT pointing at any speakers as this would cause feedback) to record them properly. Devoting an input channel to this would provide the ability to fade in applause at the end of songs, and fade the crowd out during the song. With a little imagination, a superb recording capturing the atmosphere of the performance could be achieved.

5:12 THE SOUNDCHECK

At the end of the setting up procedures:

a each input channel has been set for the particular performer's instrument to a starting level,

b submixes (if required) will have been set roughly,

c the multitrack tape recorder will be ready to record (if session recording is to take place),

d there will be an overall volume setting for the P.A. system,

e effects will be roughly set.

Throughout the setting up, each performer singularly, or a few together perhaps, will have played to allow setting up of their channels.

The soundcheck is a "dress rehearsal", and all the performers must now play a song. The main adjustments necessary will mainly be to fader levels and the auxiliary controls (particularly the foldback auxs). EQ on inputs should be about right from setting up.

Feedback or distortion can occur when the soundcheck is performed. To find it, SOLO from one end of the console to the other until the problem channel is found. Feedback is caused by a microphone and a speaker with the same signal through them being too close together - it happens worst when they are facing each other. Once the channel has been isolated, inform the stage (via the talkback system) which performer is causing the difficulty and have the monitor speaker or the microphone moved until the feedback stops. The vocalist or an athletic guitarist may move about on the stage and if in some positions they cause feedback, then they must be told to avoid these positions, or to move the monitor speakers. Reducing the level of the performer's own instrument (by turning down AUX 5 or AUX 6 on his input channel) in his monitor speaker will also help to eliminate the feedback.

With some types of feedback, careful use of the equalisation can be effective for eliminating the howling frequency. Adjust the frequency control on the input channel EQ to the approximate frequency of the feedback, then attenuate it by turning anticlockwise the +/-16dB control until the howling is no longer heard. Another way of eliminating feedback is to patch in a Delay line and use a very short delay. This simply shifts the howl in time, and does not always work. The delay must be very short (10-20ms) so that timing problems do not arise.

Through the headphones, (with no SOLO buttons depressed) the operator will be listening to the same mix as the audience. All the controls and effects throughout the console will have been set to ensure a well balanced sound at the headphones. The acoustics of the room, or peculiarities of the speaker systems may mean that the audience will not be hearing quite the same sound though. Graphic equalisers can be patched into the LEFT & RIGHT output SEND/RET points to correct this.

5:13 THE PERFORMANCE

The soundcheck should have revealed any faults in the system and deficiencies in the sound. All effects are now set up, all input channels and submixes are set, the session recorder (if used) will be ready to record, EQ of the final sound will have been perfected and the performers should have the foldback mixes they require.

During the performance watch for CLIP LEDs on input channels (reduce GAIN and bring up the fader a little so the mix is unaffected by GAIN reduction) and be ready to act if feedback occurs (quickly reduce the feed level to the offending monitor speaker - an AUX 5 or AUX 6 control).

Running adjustments will probably have to be made due to room acoustics changing (with the addition of an audience) and performers playing or singing louder. It will be up to the operator's ears to adjust the mix for a balanced sound throughout the performance.

6 Level Options

The MIXDOWN consoles have on-board links which may be moved to change output levels of the subgroups and the left/right outputs.

FIG 6 shows the printed circuit board layouts and the positions of the OPTION LINKS.

6:1 LEFT/RIGHT OUTPUTS

The XLR outputs of MIXDOWN consoles are at +4dBm UNBALANCED, by moving one link on each PCB, they may be changed to +4dBm BALANCED. The balanced position will almost certainly be required if the console is to be used for live work as most P.A. amplifiers have balanced inputs. The jack outputs will remain at -10dBV unbalanced.

6:2 SUBGROUPS

The TAPE outputs and TAPE inputs of the MIXDOWN are set for use with -10dBV multitrack tape recorders. All THREE links on each subgroup PCB must be moved to allow the MIXDOWN to be compatible with a +4dBm tape recorder. Chapter 5 explains the effect incorrect link settings will have on recording levels.

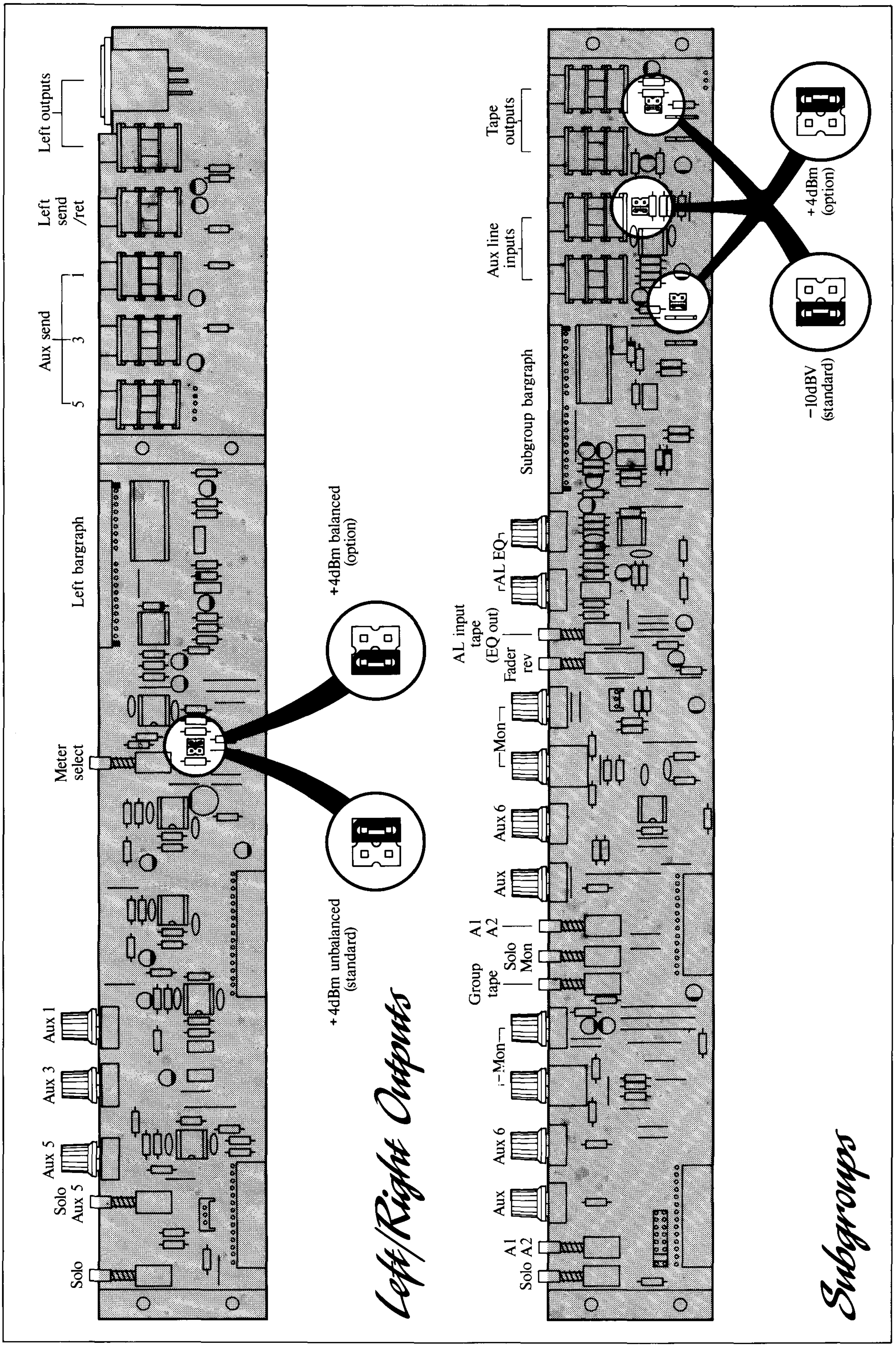

Left outputs
Left send /ret
Aux send
1
3
5
Left bargraph
Meter select
Aux 1
Aux 3
Aux 5
Solo Aux 5
Solo
+4dBm balanced (option)
+4dBm unbalanced (standard)
Left/Right Outputs
Tape outputs
Aux line inputs
Subgroup bargraph
AL EQ
AL input tape (EQ out)
Fader rev
Mon
Aux 6
Aux
A1 A2
Solo Mon
Group tape
Mon
Aux 6
Aux
A1
Solo A2
+4dBm (option)
−10dBV (standard)
Subgroups

FIG 6

7 Expansion

STUDIOMASTER MIXDOWN consoles can have extra input channels added, allowing almost the same flexibility as modular consoles for building larger consoles as requirements change.

7:1 FITTING PROCEDURE

1 Check the contents of the expander kit. The kit comprises 1 4-channel expander, 9 hex-head set screws and shakeproof washers (C), and a packet of numbered self-adhesive labels.

2 Remove the right endcheek from the console via the THREE posi screws (A).

3 Remove the front panels from the mixing console and the expander. There are EIGHT on the expander, TEN on the 16.4.8 and TWELVE on the 16.8.16. The front panel of the console cannot be completely removed because of the Power Loom. Either unplug the power loom FROM THE POWER SUPPLY BOARD, or prop up the front panel at an angle to allow the loom to stay attached, and to allow enough room for access to screw the two bases together.

4 Offer up the two bases, and fit the NINE hex-head screws and shakeproof washers (C). SIX fit from the "console-side", and THREE fit from the "expander-side". Don't forget the shakeproof washers.

5 Rest the two front panels back on their bases in such a way that the two looms with free ends from both the console and the expander can be passed through the rectangular holes in the joined chassis, with the excess neatly folded in the expander base. Make sure that the looms are not crossed over or twisted.

6 Now lower the front panels into place being careful not to trap any of the looms inside. Watch out for the Power Loom also.

7 Re-fit all the screws (B) removed from the front panels.

8 On the right-hand side of the expander are two more free looms, ready for the next expander. Make sure that these are neatly folded away.

9 Re-fit the right endcheek with the THREE posi screws (A).

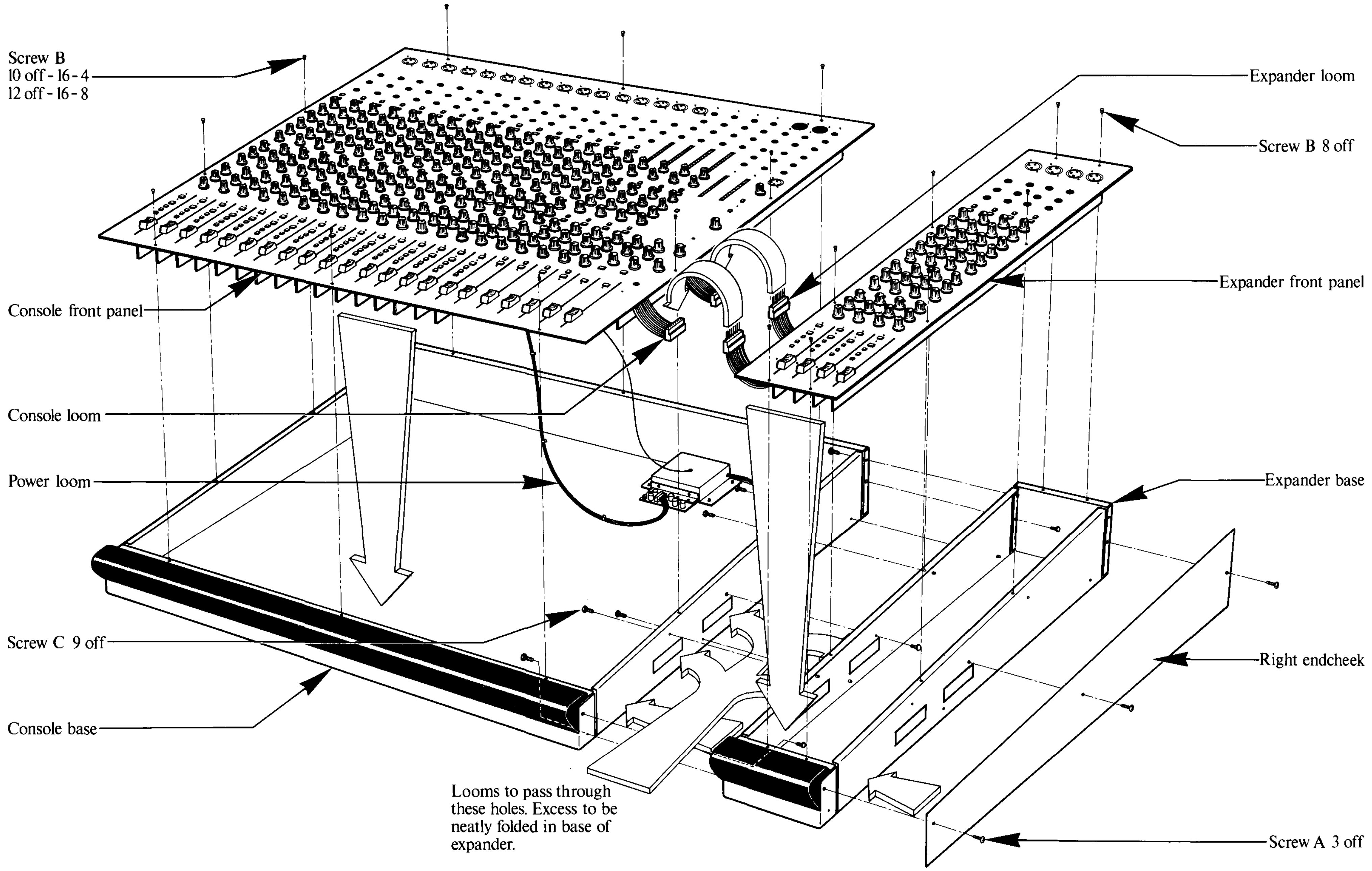
10 Fit the labels over the printed numbers on the expander.

When fitting a further expander, the first expander is effectively the right-hand end of the console, so the console front panel need not be removed again. A maximum of 4 expanders may be fitted to make maximum console sizes 32.4.8 and 32.8.16.

7:2 MORE THAN 24 INPUTS

The MIXDOWN with its original power supply (internal or external EP4) can supply up to 24 inputs. Further inputs require the EP3 rack mounting power supply.

If the MIXDOWN already has the EP4 power supply, then it can be easily replaced with the EP3. Internally powered models will require an interface board which a Studiomaster dealer or the Studiomaster Service Department can supply and fit.



EXPANDING THE MIXDOWN

FIG 7

8 Specifications

TOTAL EQUIVALENT INPUT NOISE (DIN Audio)

10kohm source TAPE input :	-95dB
150ohm source MIC input :	-129dB
10kohm source LINE input :	-89dB

COMMON MODE REJECTION RATIO

MIC maximum Gain @ 100Hz :	76dB
MIC Maximum Gain @ 1kHz :	78dB

TOTAL HARMONIC DISTORTION & NOISE

MIC Gain 50dB, Master
Output 0dBm into 600ohm : Less than 0.05% @ 1kHz

FREQUENCY RESPONSE (+0-1dB)

MIC Gain 50dB, Master
Output 0dBm into 600ohm : 30Hz to 20kHz

INPUT IMPEDANCE

MIC Input :	approx 2kohm
LINE Input :	greater than 50kohm
TAPE Input :	greater than 10kohm

GAIN RANGE

MIC Input :	+15 to +60dB
LINE Input :	-15 to +30dB
TAPE Input :	-15 to +30dB

PAD

20dB attenuation on MIC and LINE inputs.

SIGNAL TO NOISE RATIO

<u>MIC Gain 50dB, 150ohm source</u>

MIC Input to L/R Output :	76dB (ref +4dBm)
MIC Input to Subgroup (-10) :	76dB (ref -10dBV)
MIC Input to Subgroup (+4) :	76dB (ref +4dBm)

<u>LINE Gain 0dB, 10kohm source</u>

LINE Input to L/R Output :	85dB (ref +4dBm)
LINE Input to Subgroup (-10) :	86dB (ref -10dBV)
LINE Input to Subgroup (+4) :	86dB (ref +4dBm)

<u>TAPE Gain 0dB</u>

TAPE Input to L/R Output :	87dB (ref +4dBm)
TAPE Input to Subgroup (-10) :	89dB (ref -10dBV)
TAPE Input to Subgroup (+4) :	89dB (ref +4dBm)

CROSSTALK

Measured @ 1kHz

Input to Adjacent Input :	Below Noise
Left Output to Right Output (separated by PAN control) :	Typically -72dB
Subgroup to Subgroup (2-3) (separated by routing switch) :	Typically -88dB
Subgroup to Subgroup (1-2) (separated by PAN control) :	Typically -72dB

NOMINAL OUTPUT NOISE (DIN Audio)

Faders at "0"

L/R Outputs (+4dBm unbal.) :	-86dBm (16 monitors, no routing)
L/R Outputs (+4dBm unbal.) :	-80dBm (16 monitors, 16 inputs routed)
Subgroups (-10dBV) :	-108dBV (No routing)
Subgroups (-10dBV) :	-99dBV (16 inputs routed)
Subgroups (+4dBm) :	-95dBm (No routing)
Subgroups (+4dBm) :	-85dBm (16 inputs routed)

MAXIMUM GAIN

MIC Input to Subgroup or L/R Output	80dB
LINE Input to Subgroup or L/R Output	50dB

NOMINAL OUTPUT LEVEL

<u>L/R Output</u>
+4dBm (1.23Vrms) Balanced, or +4dBm Unbalanced
Factory Set at +4dBm Unbalanced (internally switchable)
<u>2-Track Output</u>
-10dBV (300mVrms)
<u>Subgroup</u>
+4dBm (1.23Vrms), or -10dBV (300mVrms)
Factory Set at -10dBV (internally switchable)
For Level Options refer to Chapter 6.

MAXIMUM OUTPUT

L/R Output (+4dBm unbalanced) into 600ohm min	+20dBm (8Vrms)
Subgroup (-10dBV) - 5kohm min	+6dBV (2Vrms)
Subgroup (+4dBm) - 600ohm min	+20dBm (8Vrms)
2-Track Output - 5kohm min	+6dBV (2Vrms)

HEADPHONE OUTPUT

Maximum Level Before Clip

Into 8ohm	1.2Vrms (180mW)
Into 600ohm	7.5Vrms (90mW)

STEREO RETURN

Maximum Gain	28dB
1kHz Maximum Attenuation	75dB
Input Impedance	10kohm min.

TAPE MONITOR

Sensitivity -10dBV or +4dBm for 0VU ref.
Factory Set at -10dBV (internally switchable with subgroup levels)

EQUALISATION

INPUT	Treble (HF)	+/-16dB at 12kHz SHELVING
	Midrange (MF)	+/-16dB at 280Hz to 8kHz PEAK/DIP
	Bass (LF)	+/-16dB at 25Hz to 350Hz SHELVING
AUX LINE INPUT	Treble (HF)	+/-12dB at 12kHz SHELVING
	Bass (LF)	+/-12dB at 60Hz SHELVING

PEAK/CLIP

Dynamic Operation, indicates 4dB prior to clip

METERS

VU response 12 segment, 2 colour LED bargraphs with "0" ref. level

FADERS

100mm Sealed Carbon Track

DIMENSIONS (WxHxD)

16.4.8	780mm x 130mm x 640mm
16.8.16	910mm x 130mm x 640mm
4-input expander	165mm x 130mm x 640mm

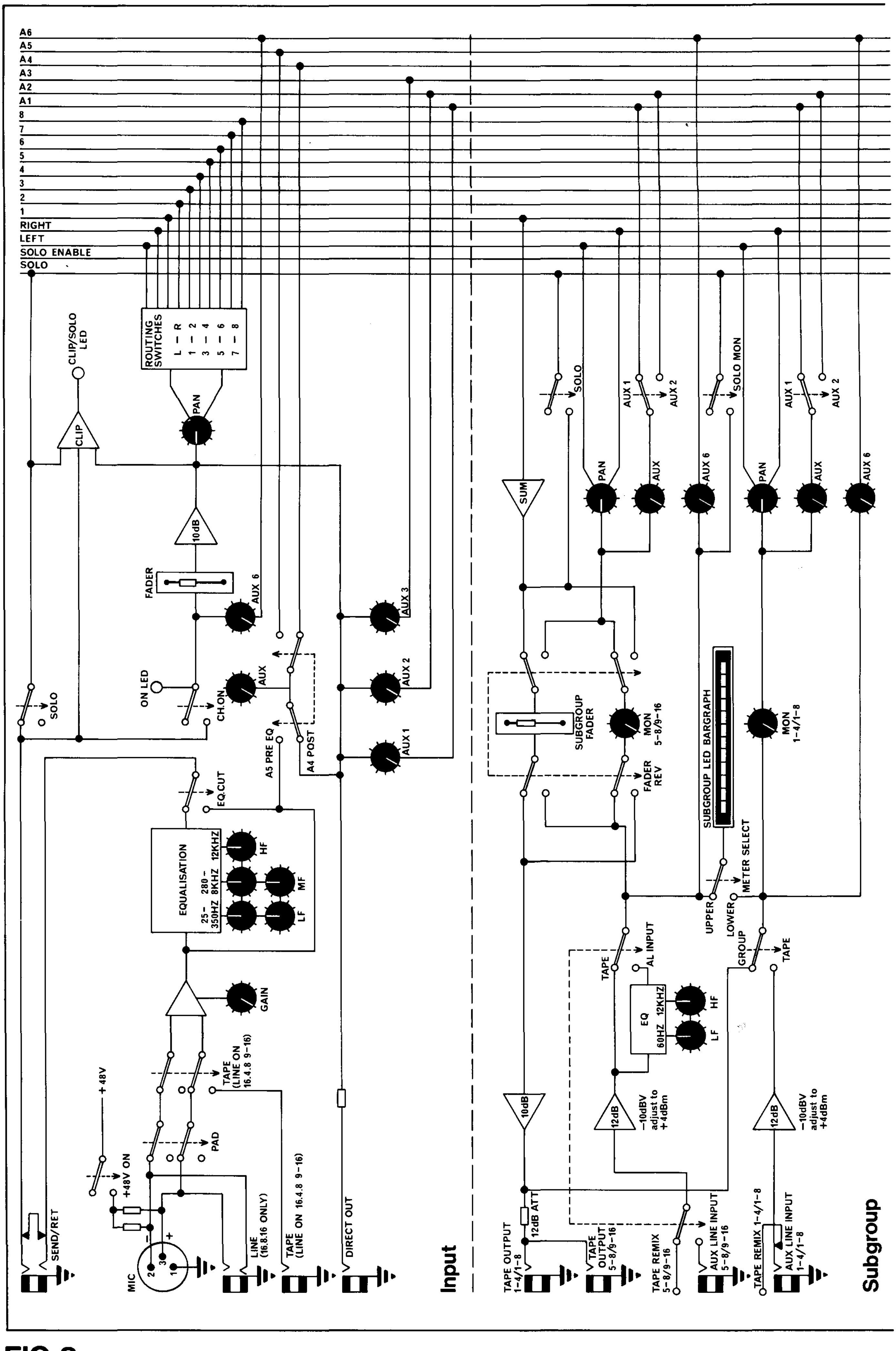

A6
A5
A4
A3
A2
A1
8
7
6
5
4
3
2
1
RIGHT
LEFT
SOLO ENABLE
SOLO
ROUTING SWITCHES
L – R
1 – 2
3 – 4
5 – 6
7 – 8
CLIP/SOLO LED
CLIP
PAN
10dB
FADER
AUX 6
ON LED
CH.ON
AUX
AUX 3
AUX 2
AUX 1
A5 PRE EQ
A4 POST
EQ.CUT
EQUALISATION
25- 350HZ
280- 8KHZ
12KHZ
HF
MF
LF
GAIN
+48V
+48V ON
TAPE (LINE ON 16.4.8 9–16)
PAD
MIC
SEND/RET
LINE (16.8.16 ONLY)
TAPE (LINE ON 16.4.8 9–16)
DIRECT OUT
Input
SUM
SOLO MON
SUBGROUP FADER
MON 5–8/9–16
FADER REV
SUBGROUP LED BARGRAPH
METER SELECT
UPPER
LOWER
MON 1–4/1–8
TAPE
AL INPUT
GROUP
EQ
60HZ
12KHZ
12dB
–10dBV adjust to +4dBm
12dB ATT
TAPE OUTPUT 1–4/1–8
TAPE OUTPUT 5–8/9–16
TAPE REMIX 5–8/9–16
AUX LINE INPUT 5–8/9–16
TAPE REMIX 1–4/1–8
AUX LINE INPUT 1–4/1–8
Subgroup

FIG 8a

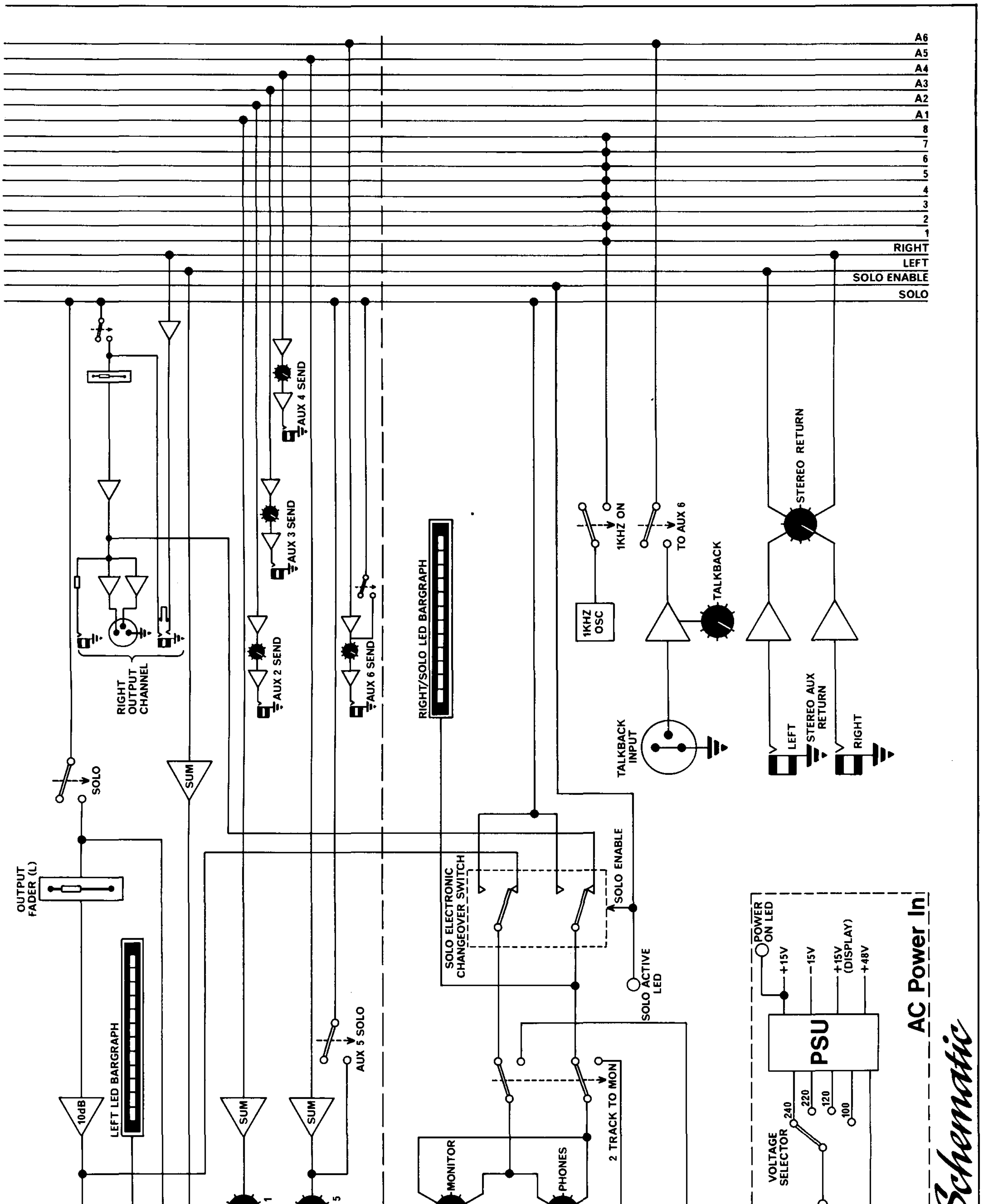

Mixdown Schematic
A6
A5
A4
A3
A2
A1
8
7
6
5
4
3
2
1
RIGHT
LEFT
SOLO ENABLE
SOLO
2 TRACK OUTPUT (L)
12dB
OUTPUT (L)
SEND/RET (L)
OUTPUT FADER (L)
SOLO
SUM
LEFT LED BARGRAPH
10dB
RIGHT OUTPUT CHANNEL
AUX 1 SEND
AUX 1
AUX 5 SEND
AUX 5
AUX 5 SOLO
AUX 2 SEND
AUX 3 SEND
AUX 4 SEND
AUX 6 SEND
Output
RIGHT/SOLO LED BARGRAPH
SOLO ELECTRONIC CHANGEOVER SWITCH
SOLO ENABLE
SOLO ACTIVE LED
2 TRACK TO MON
MONITOR
PHONES
LEFT
MONITOR OUTPUT (600R MIN)
RIGHT
PHONES OUTPUT (8R MIN)
2 TRACK RETURN
Monitor
1KHZ OSC
1KHZ ON
TO AUX 6
TALKBACK
TALKBACK INPUT
STEREO RETURN
STEREO AUX RETURN
POWER ON LED
+15V
-15V
+15V (DISPLAY)
+48V
PSU
240
220
120
100
VOLTAGE SELECTOR
FUSE
T2A
POWER ON
CHASSIS
L
E
N
AC Power In

FIG 8b