SEMI - PROFESSIONAL MIXING DESK

PART ONE

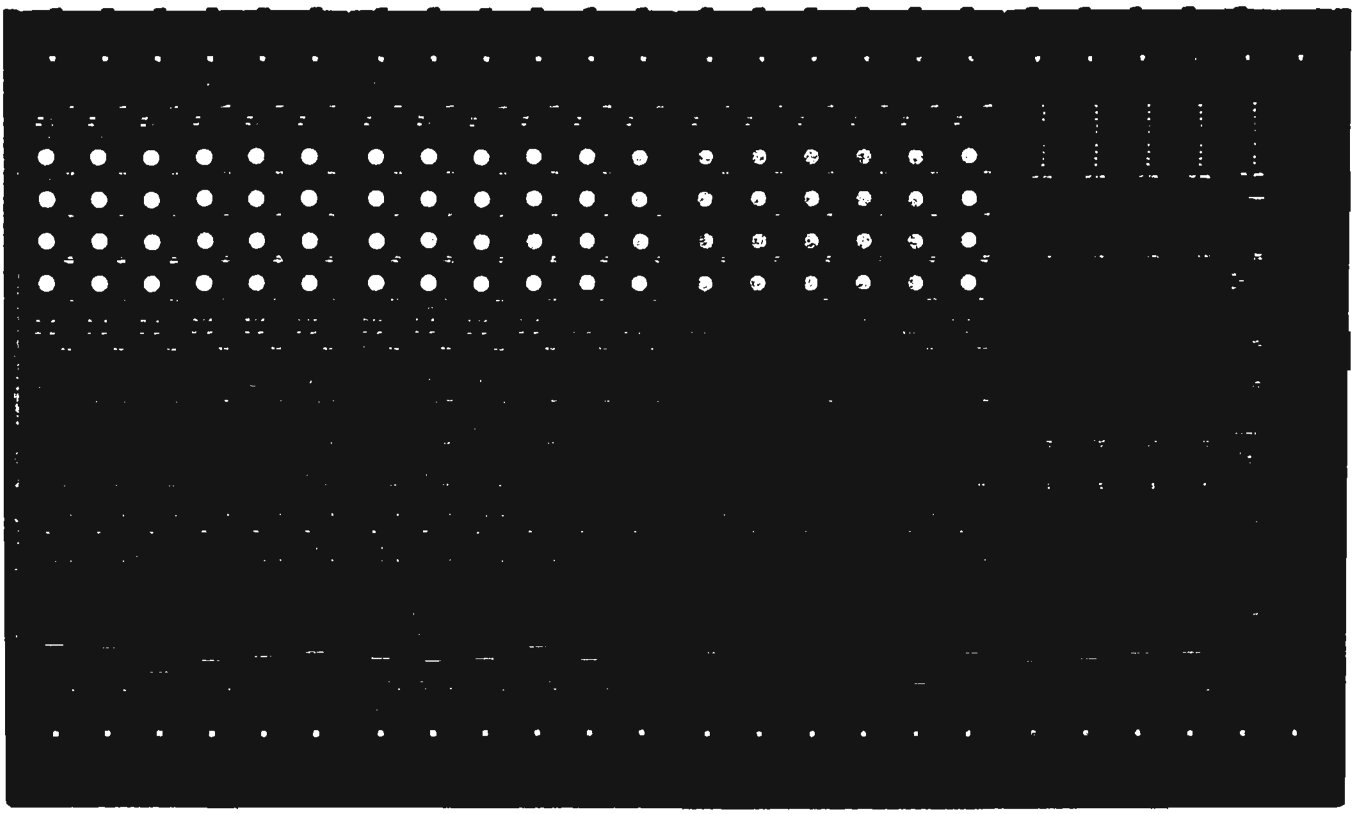
TIM ORR

THE mixer described in this article is constructed from a range of input and output modules all of which plug into a common bus. Even the mixer chassis is modular. It comes in 6 module sections which can be bolted together. The mixer can be assembled with up to 18 inputs (you could even have 24 inputs if you are that keen) and 4 output channels. For example, if you wanted a 6 into 2 mixer then all you would need is 6 input modules, 2 output modules, 1 auxiliary module (optional) and three blanking panels. The power supply for the system is contained in an external box.

INPUT MODULE

The circuit diagram of the input module is shown in Fig. 1. IC1 forms an unbalanced low noise preamplifier with switched and variable gain. The system is designed to run with a signal level of 0dBm (0·775Vr.m.s. or 2·2Vpp) and it is the job of input stage to amplify/attenuate the input signal to this level. For a 0dBm signal level at IC1 pin 1, the microphone signal level can vary between −56 to −16dBm and the line level signal from −24 to +11dBm. Therefore the preamplifier can accept input levels from −56dBm to +11dBm for a 0dBm output. Also the maximum level at IC1 pin 1 before clipping is +18dBm and so there is 18dB of headroom when operating at a signal level of 0dBm.

The tone control has a conventional treble and bass circuit plus a parametric section. The parametric equaliser is constructed from a variable-frequency state-variable bandpass filter which can be used to provide feedforeward (lift) or feedback (cut) around an amplifier section (IC2b). Fig. 2 shows the tone control frequency responses. The equaliser can be bypassed by using the FLAT/EQ switch.



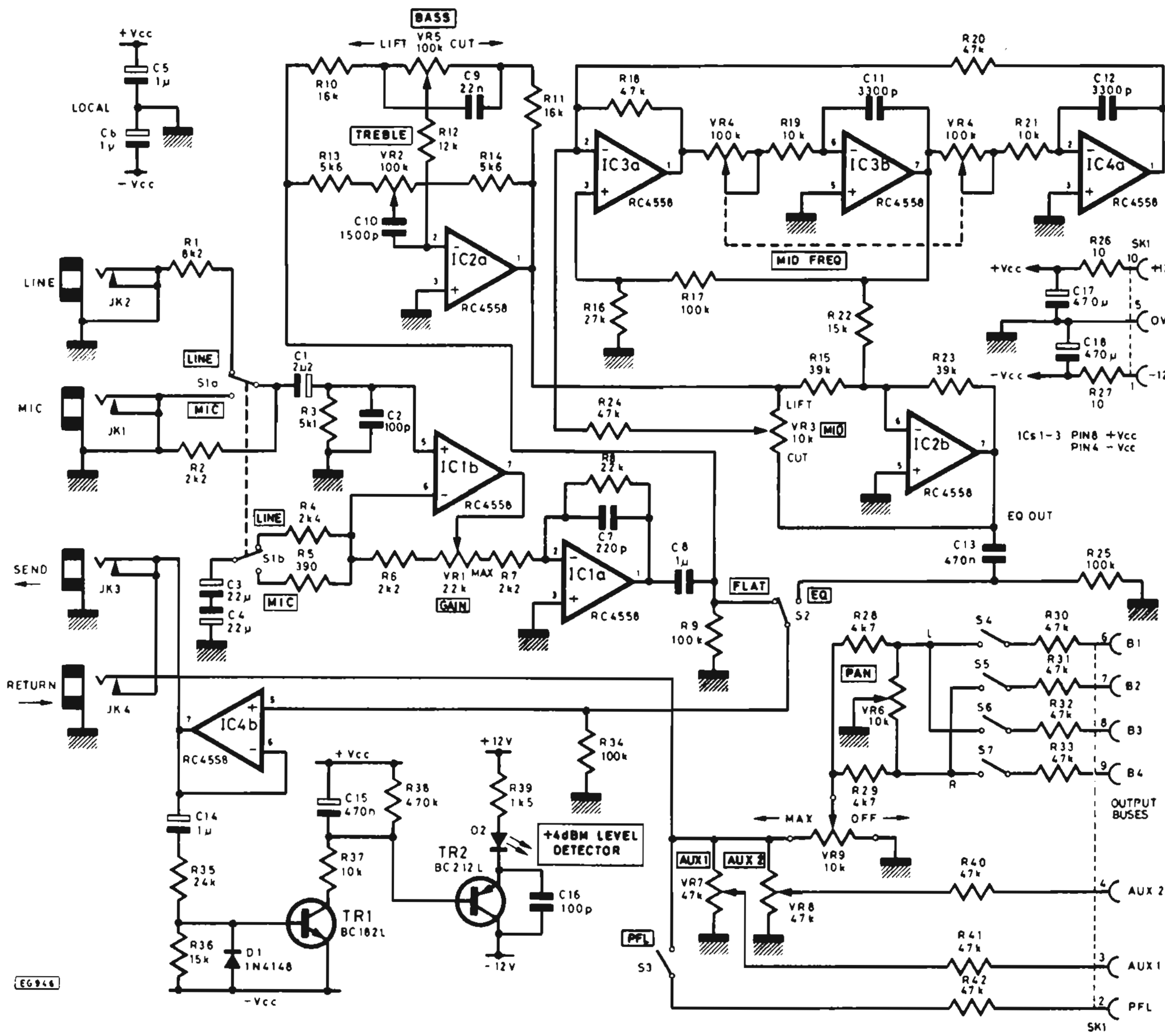


Fig. 1. Circuit diagram of the Input Module

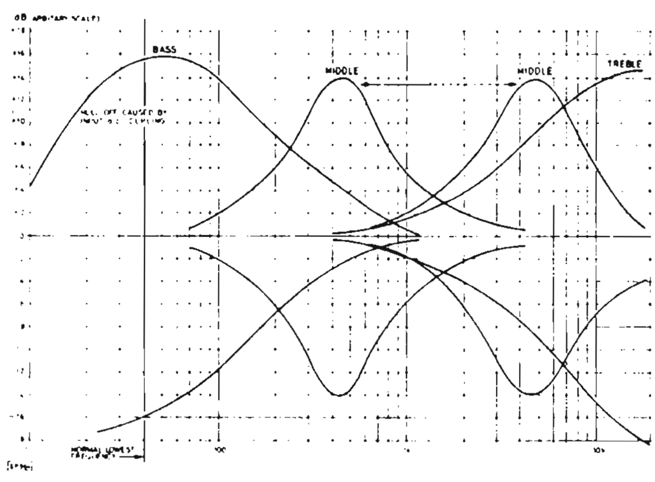


Fig. 2. Tone control frequency responses

The input module can be used to send the amplified signal to an external effects unit via the SEND jack. This signal can then be re-inserted via the RETURN jack. The RETURN jack has a break contact, but the SEND jack has not. Therefore you can use the SEND output to drive foldback monitors without interrupting the signal path. PPM signal level monitoring per input channel is very expensive and so a simple peak level detector has been used. This device lights up a l.e.d. when the signal level exceeds +4dBm. When this l.e.d. turns on it produces a very dirty current which can cause an annoying background noise. However by *not* dumping this current down the ground rail (the current travels from one supply rail to the other) this effect can be avoided. In fact throughout the mixer all currents that are dirty have *not* been dumped into the ground rail.

The output signal from the input module can be sent to up to 7 different ouputs. Before the channel fader it can be switched to the PFL (pre-fade-listen) bus. This route has a fixed unity gain and it enables the mixer operator to monitor the channel signal level on the PFL PPM (this is on the AUX channel) and also to listen to the signal on its own, on either headphones or monitor speakers. Also there are two aux-

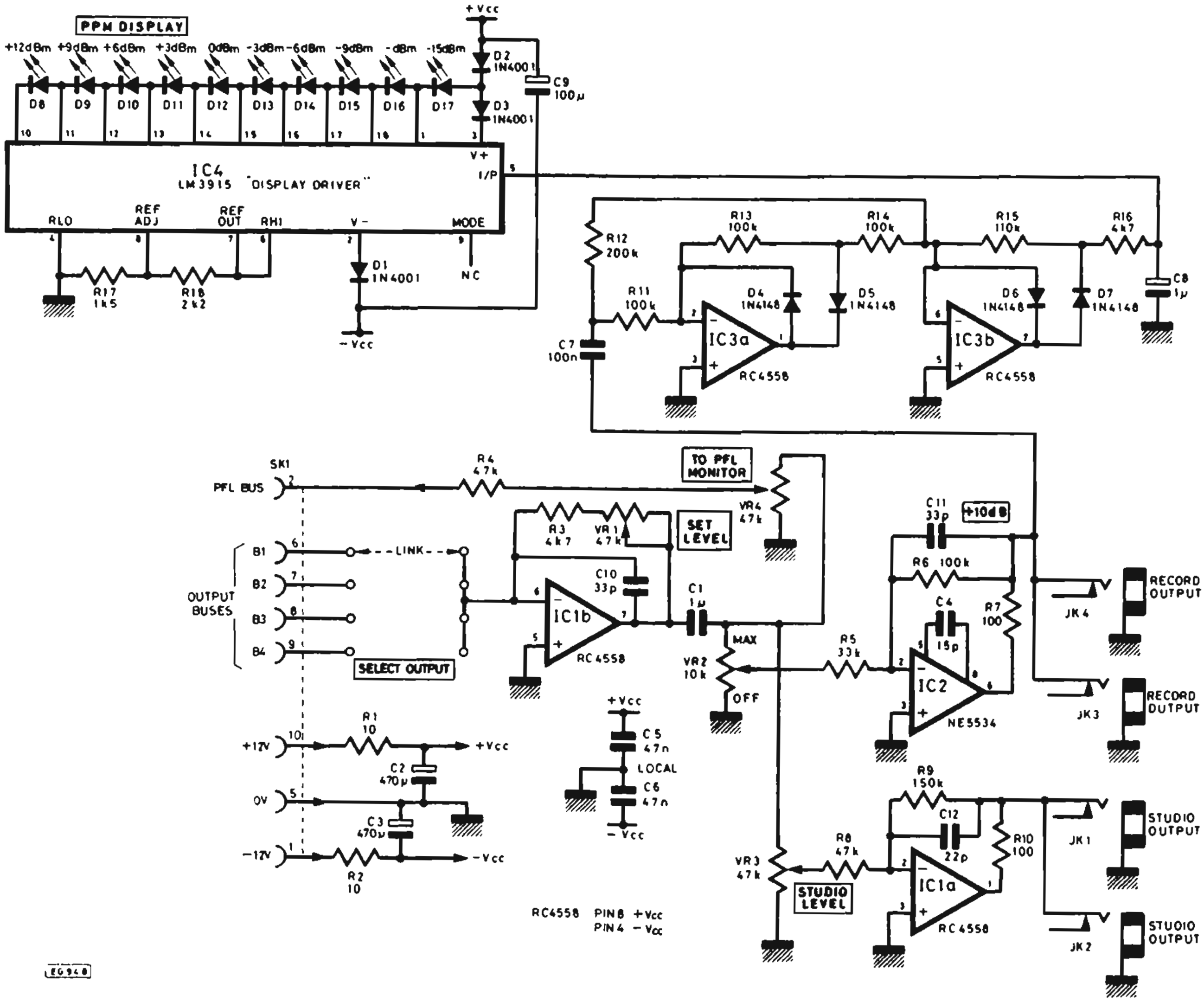


Fig. 4. Circuit diagram of the Output Channel. Note D1 to D3 should be 1N4002

iliary buses (AUX1 and AUX2) which can be used to produce mixes separate to the 4 main outputs. After the channel fader the signal is split up by a pan pot and can then be sent (via selector switches) to the 4 output channel buses. Outputs 1 and 3 are left hand channels and 2 and 4 are right hand channels. The block diagram of the input, output and auxiliary channels are shown in Figs. 6, 7 and 8.

POWER SUPPLY

The p.s.u. circuit is shown in Fig. 3. The power supply can deliver up to ±1 amp at ±12V. It is mounted externally to the mixer to avoid mains hum problems. An RC filter in the circuit (R2, R3 & C3, C4) smoothes out the unregulated rail so that the regulators are presented with a very small ripple (about 100mVpp at full load). An 18 into 4 mixer consumes about 500mA from each rail. This current increases when the PPM displays, peak l.e.d.s and headphone amplifier are on. Make certain that both regulators are insulated from the metal work and that all mains wiring is covered with rubber sleeving.

OUTPUT CHANNEL

The output channel which consists of three virtual earth amplifiers and a PPM circuit is shown in Fig. 4. Both the 'Record' and the 'Studio' output stages have +10dBs of gain. The 'Record' output uses a high performance op-am which is capable of driving a +18dBm 20kHz sinewave into 600 ohms without anything nasty happening! This is the best output to use; the PPM circuit monitors the signal level at this output. The PPM (peak programme meter) consists of a precision full wave rectifier, a peak level detector and a National Semiconductor logarithmic bar graph driver, IC4

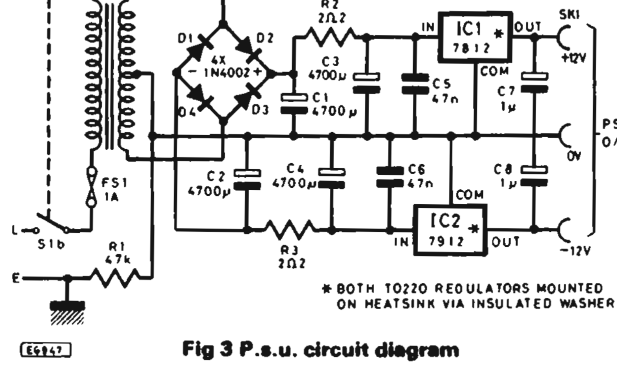


Fig 3 P.s.u. circuit diagram

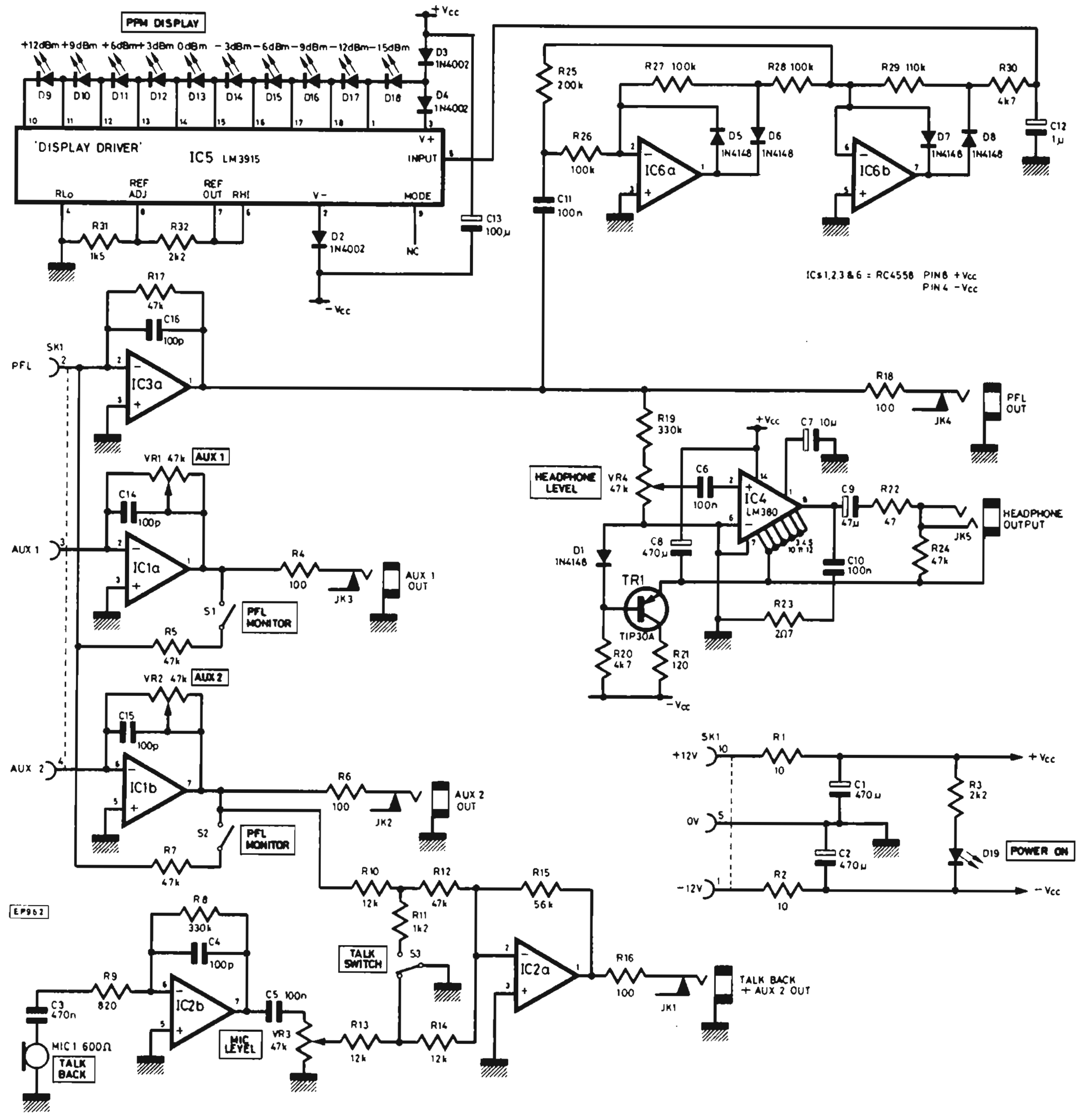


Fig. 5. Circuit diagram of the Auxiliary Channel

The display has been designed to consume very little current. IC4 is run in its dot mode, so that only one output is on at any time. However the current being sunk into any display output also lights up all the l.e.d.s below it. In this way the dot mode is transformed into a bar graph. The display current is only 10mA and is constant even though 10 l.e.d.s may be on. Note that none of the l.e.d. current is dumped into the ground rail.

It is important that the l.e.d.s protrude through the panel to the same height, otherwise the PPM display looks rather nasty. A small metal or wooden jig should be constructed so that all the l.e.d.s can be bent to exactly the same length.

AUX CHANNEL

Both the auxiliary channels in Fig. 5 are simple virtual earth amplifiers. AUX2 is available as a direct signal and as a mix with the mixer talk back signal. When the talk switch is pressed the AUX2 level is attenuated and the talk back microphone signal is enabled.

The PFL amplifier has unity gain (fixed) so that the signal level in any input channel can be monitored on the PFL PPM unit. A small power amplifier (IC4) provides a headphone monitor for the PFL signal. Note that a synthetic ground rail (TR1) has been produced so that the large headphone current is not dumped down the real ground rail.

COMPONENTS . . .

INPUT CHANNEL

Resistors

R1	8k2
R2, R6, R7	2k2 (3 off)
R3	5k1
R4	2k4
R5	390
R8	22k
R9, R17, R25, R34	100k (4 off)
R10, R11	16k (2 off)
R12	12k
R13, R14	5k6 (2 off)
R15, R23	39k (2 off)
R16	27k
R18, R20, R24, R30, R31, R32, R33, R40, R41, R42	47k (10 off)
R19, R21, R37	10k (3 off)
R22, R36	75k (2 off)
R26, R27	10 (2 off)
R28, R29	4k7 (2 off)
R35	24k
R38	470k
R39	1k5

All resistors ¼ W metal film

Potentiometers

VR1	22k lin
VR2, VR5	100k lin (2 off)
VR3, VR6	10k lin (2 off)
VR4	100k reverse log dual pot
VR7, VR8	47k log pot with p.c. bracket (2 off)
VR9	10k log slider ALPS 191M10kA

Capacitors

C1	2µ2 16V tant
C2, C16	100p ceramic (2 off)
C3, C4	22µ 10V tant (2 off)
C5, C6, C14	1µ 35V elect (3 off)
C7	220p ceramic
C8	1µ B32560
C9	22n B32560
C10	1n5 B32560
C11, C12	3n3 B32560 (2 off)
C13	470n B32560
C15	470n 35V elect
C17, C18	470µ 16V elect (2 off)

Semiconductors

D1	1N4148
D2	0·2in red l.e.d.
TR1	BC182L
TR2	BC212L
IC1–IC4	RC4558 (4 off)

Miscellaneous

SK1–SK4	¼in mono jack socket plus shorting tip pin (4 off)
S1–S7	Push switches p.c.b. mounting d.p.d.t. (7 off)
	Knobs ¼in (8 off)
	Caps with line, black (4 off)
	Caps with line, black (4 off)
	p.c.b.
	10 way Molex 0·156in p.c. socket
	8 pin d.i.l. sockets (4 off)

OUTPUT CHANNEL

Resistors

R1, R2	10 (2 off)
R3, R16	4k7 (2 off)
R4, R8	47k (2 off)
R5	33k
R6, R11, R13, R14	100k (4 off)
R7, R10	100 (2 off)
R9	150k
R12	200k
R15	110k
R17	1k5
R18	2k2

All resistors ¼ W metal film

Potentiometers

VR1, VR3, VR4	47k log pot with a p.c. bracket (3 off)
VR2	10k log slider 191 M10kA

Capacitors

C1	1µ B32560
C2, C3	470µ 16V elect (2 off)
C4	15p ceramic
C5, C6	47n 35V ceramic (2 off)
C7	100n B32560
C8	1µ 16V elect
C9	100µ 40V elect
C10, C11	33p ceramic (2 off)
C12	22p ceramic

Semiconductors

D1, D2, D3	1N4002 (3 off)
D4, D5, D6, D7	1N4148 (4 off)
D8–D17	0·2in l.e.d. (10 off)
IC1, IC3	RC4558 (2 off)
IC2	NE5534
IC4	LM3915

Miscellaneous

SK1–SK4	¼in mono jack socket with shorting tip pin (4 off)
	¼in knobs (3 off)
	Caps with line, black (3 off)
	Slider knob (CS9)
	P.c.b.
	10 way Molex p.c. socket
	8 pin d.i.l. socket (3 off)
	18 pin d.i.l. socket

GENERAL PARTS

Chassis units
Wooden end cheeks
Wooden front pieces
Bus p.c.b. (with 6 × 10 way Molex pins)
Rubber feet
Grommet
Ty-rap base
Ty-rap
3 core lead
3 pin 180° inline plug

AUX CHANNEL

Resistors

R1, R2	10 (2 off)
R3	2k2 1W
R4, R6, R16, R18	100 (4 off)
R5, R7, R12, R17, R24	47k (5 off)
R8, R19	330k (2 off)
R9	820
R10, R13, R14	12k (3 off)
R11	1k2
R15	56k
R20, R30	4k7 (2 off)
R21	120
R22	47
R23	2Ω7
R25	200k
R26, R27, R28	100k (3 off)
R29	110k
R31	1k5
R32	2k2

All resistors ¼W metal film unless otherwise stated

Potentiometers

VR1–VR4	47k log pot with p.c. bracket

Capacitors

C1, C2, C8	470µ 16V elect (3 off)
C3	470n 832580
C4, C14, C15, C16	100p ceramic (4 off)
C5, C6, C10, C11	100n 832580 (4 off)
C7	10µ 16V elect
C9	47µ 16V elect
C12	1µ 16V elect
C13	100µ 40V elect

Semiconductors

D1, D5, D6, D7, D8	1N4148 (5 off)
D2, D3, D4	1N4002 (3 off)
D9–D18	0·2in square red l.e.d. (10 off)
D19	0·2in red l.e.d.
TR1	TIP30A
IC1, IC2, IC3, IC6	RC4558 (4 off)
IC4	LM380
IC5	LM3915

Miscellaneous

SK1–SK4	¼in mono jack socket plus shorting tip jack (4 off)
SK5	¼in stereo jack socket
VR1–VR4	47k log pot plus p.c. bracket (4 off)
S1, S2	Push switch p.c. mounting d.p.d.t.
	¼in knob (4 off)
	Cap with line, black (4 off)
S3	push switch 8125–J81 3/2
	10 way Molex p.c. socket
	p.c.b.
	8 pin d.i.l. socket (4 off)
	14 pin d.i.l. socket
	18 pin d.i.l. socket
	600 Ω microphone

POWER SUPPLY UNIT

Resistors

R1	4k7 ¼W
R2, R3	2k2 2·5W

Capacitors

C1–C4	4700µ 25V elect (4 off)
C5, C6	47n 35V ceramic disc (2 off)
C7, C8	1µ 16V tant (2 off)

Semiconductors

D1–D4	1N4002 (4 off)
IC1	7812 with insulating kit
IC2	7912 with insulating kit

Miscellaneous

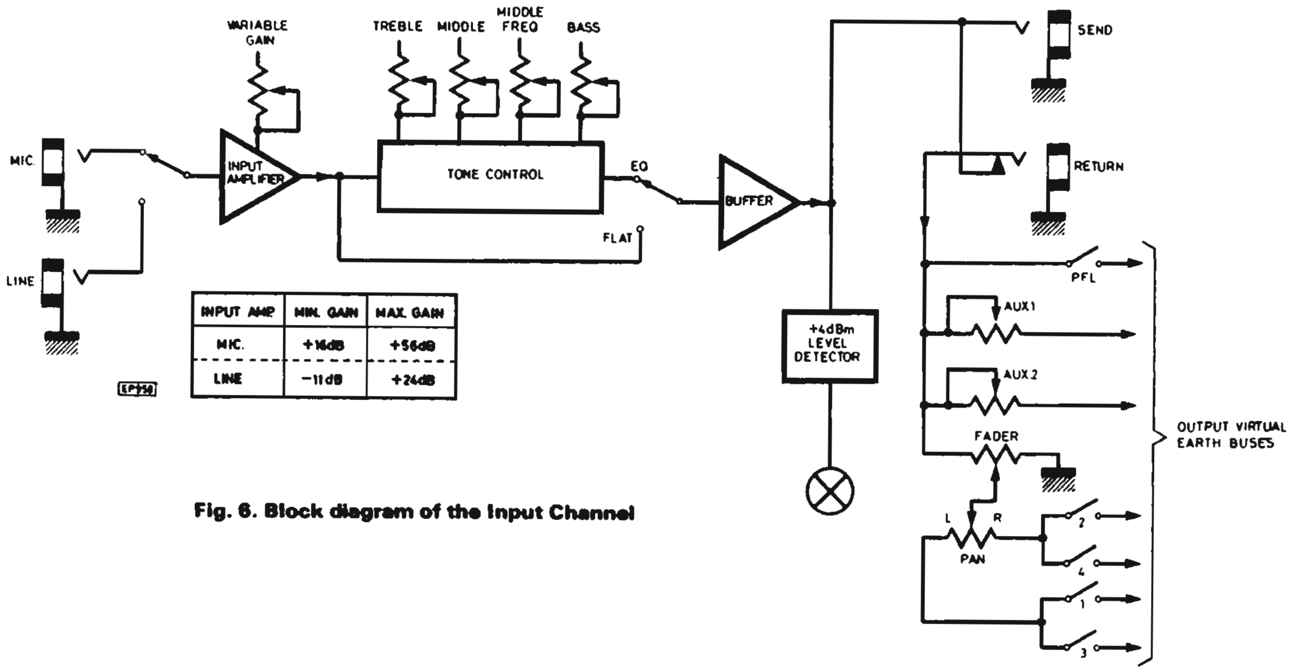
S1	Mains switch d.p.s.t.
T1	15–0–15V torroid at 30VA
FS1	20mm 1A fuse
	20mm fuseholder
	Rubber feet
	Heat sink bracket
	P.c.b.
	Mains grommet
	P.s.u. case
	4 way screw block
	3 pin 180° din socket

Constructor's Note

Complete kits of parts for this project can be obtained from Powertran Electronics, Portway Industrial Estate, Andover, Hants SP10 3WN (0264 64455)

Input channel (including p.c.b., panel and controls)	£19·90
Output channel (including p.c.b., panel and controls)	£18·50
Auxiliary channel (including p.c.b., panel and controls)	£22·50
Blank panel	£3·00
Base unit for up to 6 channels (including wooden front)	£27·50
Pair of dark mahogany end cheeks	£12·50
Power supply (including transformer and cabinet)	£19·50

All prices subject to 15% VAT. No charge is made for carriage



INPUT AMP	MIN. GAIN	MAX. GAIN
MIC.	+16dB	+56dB
LINE	−11dB	+24dB

Fig. 6. Block diagram of the Input Channel

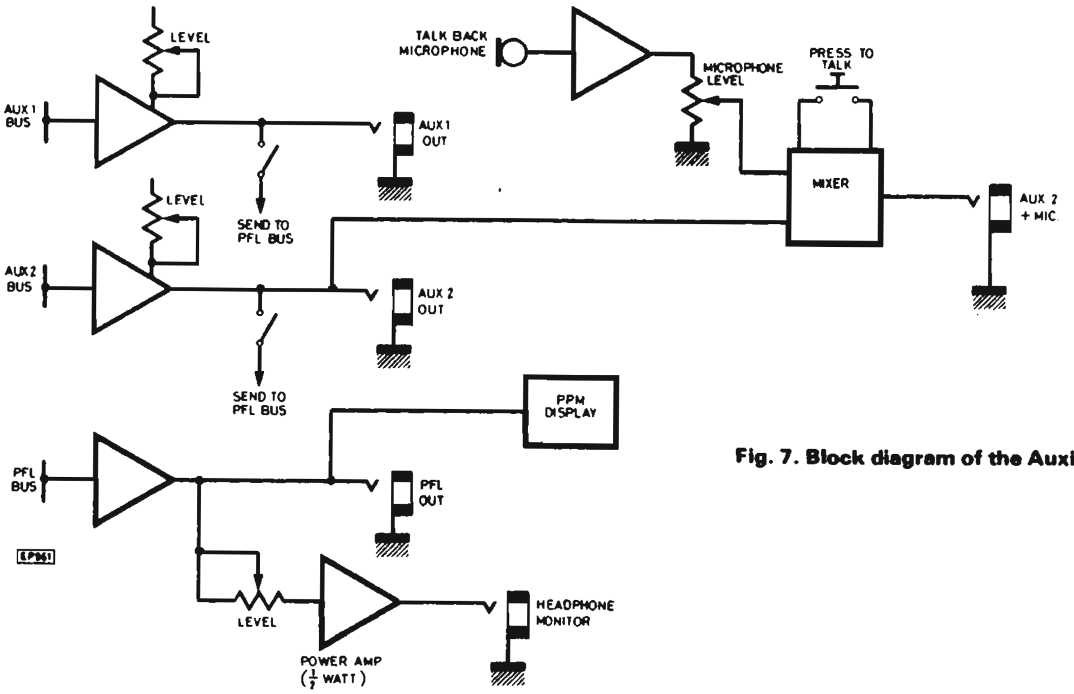


Fig. 7. Block diagram of the Auxiliary Channel

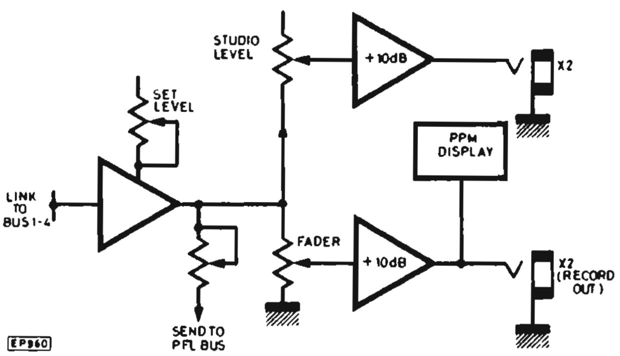


Fig. 8. Block diagram of the Output Channel

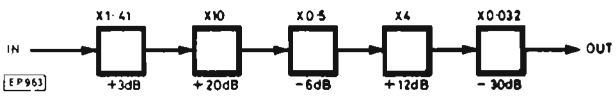
THE dB, THE dBm AND NOISE

The dB (deci-Bell) is always used to describe gains and losses in audio networks. If a signal passes through an audio network with a multiplicative gain of X, then that gain in dBs is $20\log_{10}(X)$. At first sight it may seem that the dB is just a complicated way of describing gain and loss. It is not. If a signal passes through several stages of gain then the total gain is the product of all the multiplicative gains in the system. However if you use dBs to describe the gain then the overall gain is merely the sum of the gains. It is generally easier to add than to multiply. Table 1 illustrates the advantages of using the dB.

The dBm is a logarithmic method of measuring voltage. If a voltage of −10dBm is passed through an amplifier with a

dB	Multiplier	Rule of thumb approximation
+80	x 10,000	10,000
+70	x3,162	3,000
+60	x 1,000	1,000
+50	x316·2	300
+40	x 100	100
+30	x31·62	30
+20	x 10	10
+18	x7·94	8
+12	x3·98	4
+10	x3·16	3
+6	x 1·99	2
+3	x 1·41	1·4
0	x 1·00	1·0
−3	x0·708	0·7
6	x0·501	0·5
10	x0·316	0·3
−12	x0·251	0·25
−18	x0·125	0·125
−20	x0·100	0·100
−30	x0·032	0·03
−40	x0·01	0·01
−50	x0·0032	0·003
−60	x0·001	0·001
−70	x0·00032	0·0003
−80	x0·0001	0·0001

TABLE 1



Voltage gain = 1·41 x 10 x 0·5 x 4 x 0·032
or in dBs
Voltage gain = 3 + 20·6 + 12 − 30 dBs
Typical system

dBm	Vr.m.s.	Vpp
+18	6·16V	17·47V
+12	3·08V	8·76V
+10	2·45V	6·95V
+6	1·55V	4·39V
0	0·775V	2·2V
6	388mV	1·10V
−12	197mV	553mV
−18	97·6mV	277mV
−20	77·5mV	220mV
30	24·5mV	69·5mV
−40	7·75mV	22mV
−50	2·45mV	6·95mV
60	775μV	2·2mV
−70	245μV	695μV
−80	77·5μV	220μV
90	24·5μV	69·5μV
−100	7·75μV	22μV
−110	2·45μV	6·95μV
−120	775nV	2·2μV
−130	245nV	695nV

Signal levels and the dBm.

TABLE 2

gain of +12dB, then the output voltage from the amplifier is +12−10=+2dBm. 0dBm is defined as the voltage that dissipates 1mW of power into a 600 ohm load.

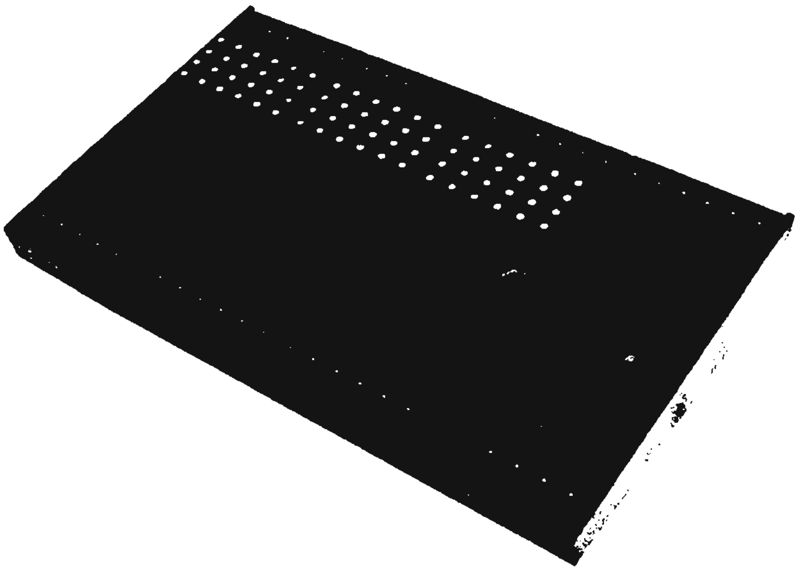
Power $= \frac{V^2}{R}$ where V is measured in Vr.m.s. and R in ohms.

Therefore, if R=600 ohms and power=0·001watt, then $V=\sqrt{600 \times 0{\cdot}001}=0{\cdot}775$Vr.m.s. Table 2 shows a chart of commonly used signal levels in audio work. Note that 0dBm is 0·775Vr.m.s. or 2·2Vpp. Studio level or line level is typically 0dBm to +6dBm. This level is large enough to avoid noise problems and small enough to allow 14 to 20dBs of headroom in mixers and other equipment.

Noise is always a problem in mixers. The signal from a low impedance microphone is usually quite small (maybe 20mV) and so a low noise amplifier is needed if a respectable signal to noise ratio is to be obtained. A mixer might be using 6 microphone channels which will result in 6 lots of noise being fed to the output channels. Noise is a random phenomena and so the 6 noise voltages will not add up linearly, but add up as the square root of the sum of their squares! If the 6 noise voltages are A,B,C,D,E & F then their combined voltage is

$$=\sqrt{(A^2 + B^2 + C^2 + D^2 + E^2 + F^2)}$$

This means that the largest noise voltage predominates. The internally generated noise of a preamplifier is usually referred to as the equivalent input noise. This is the theoretical input noise seen at the input of the amplifier. This noise is multiplied by the gain of the amplifier in just the same way as the input signal is amplified. The equivalent input noise (Ein) can be specified in dBm (this is commonly used in mixer specifications) or in μVr.m.s. or in nV/$\sqrt{Hz}$. The noise of an amplifier is measured by band limiting it to the audio bandwidth and then reading it with a true r.m.s. a.c. voltmeter. The Raytheon op-amp (RC4558) that was used in the mixer has a specified input noise of 10nV/$\sqrt{Hz}$. If we multiply this by the square root of the audio bandwidth we obtain the equivalent input noise in μVr.m.s. Therefore Ein=10nV x $\sqrt{20{,}000}$=10nVx141=1·41μVr.m.s. The measured noise from the input stage (microphone mode, input shorted to ground) was 1·46μV, which is amazingly close to the theoretical value! 1·46μVr.m.s. is equivalent to a signal level of −114·5dBm. Once we know the value of Ein it is easy to calculate the signal to noise ratio for a given input signal level. If we are using a microphone that delivers a signal level of −40dBm and the input noise is −114·5dBm, then the signal to noise ratio is



114·5−40=74·5dB. We could work this out the long way, just to show the power of the dB. The microphone signal level of −40dBm is 7·75mVr.m.s. The noise level (Ein) is 1·46μVr.m.s. Therefore the signal to noise ratio is 20log $\frac{(7{\cdot}75mV)}{1{\cdot}46\mu V}$=20log (5308·2)=74·49dB. Impossible to calculate without a calculator!

NEXT MONTH: Construction

SEMI-PROFESSIONAL MIXING DESK

Part Two

Tim Orr

THE p.c.b. designs for the input, output and auxiliary channels are shown in Figs. 1, 2 and 3 respectively with their component layouts shown in Figs. 4, 5 and 6. Note that the p.c.b.s are shown in two sections and should be joined along the X-X axis making sure that the tracks are aligned properly. Each p.c.b. should be assembled by first fitting the wire licks then the resistors, i.c. sockets, semiconductors and capacitors.

Take care that the jack sockets and potentiometers fit flush with the p.c.b. as these parts have to mate up with the metal work. The push switches can now be inserted but they should not be soldered. After the module panel has been fitted to the p.c.b. and the potentiometer retaining nuts tightened the push switches should be adjusted so that they do not interfere with the metal work. When the switches have been adjusted they can be soldered into position.

The slider switches should be mounted last on the input and output channels along with the mounting socket SK1. On the auxiliary channel the microphone and pre-fade listen jack socket wires should be twisted before being soldered to the p.c.b. Tinned copper wire can be used for wiring the talk back switch.

After all the channel boards have been soldered carefully check each one for any solder splashes or incorrectly placed components.

The p.c.b. design for the bus board is shown in Fig. 7, the thick tracks are for power supplies and the thin tracks for signals. The number of bus boards required depends upon the number of channels incorporated into the mixer, one bus board is required for every six channels used. Ten way pin blocks are used to connect the power supplies and signals to and from the bus board to each channel. After the pin blocks have been soldered onto the board (Fig. 8) trim off the excess pin lengths on the underside. The bus board can now be fitted into the case using spacers to prevent the pins being shorted to the chassis.

Make certain that the 4 signal tracks are facing the front of the mixer. If more than one bus board is used they should be joined together using 24s.w.g. tinned copper wire covered with rubber sleeving. The mains supply lead should be connected to the bus board as shown in Fig. 9.

TESTING

The input channels are relatively simple circuits; only 4 op amps are used to perform the amplification and tone control functions. Inject a 1Vpp sinewave into the line input, select LINE operation. The output signal seen at the SEND jack can be varied from 0·3Vpp to 18Vpp (with FLAT selected) by rotating the GAIN control pot. Now select the MIC mode and inject a 10mVpp signal into the MIC jack. The signal seen on the send jack should be variable over a range of 63mV to 6·3Vpp. In all cases the output signal should be free from distortion and clipping. Note that the LEVEL DETECTOR l.e.d. should come on and stay on when the SEND level exceeds +4dBm (3·5Vpp). The noise performance of the input amplifier can be measured, but only if you have the use of the equipment shown in Fig. 10. The procedure is as follows. Remove all inputs and select MIC operation and maximum gain. Also select FLAT operation and measure the noise voltage at the SEND jack. The theoretical input noise is 1·46μVr.m.s. which when multiplied by the MIC gain of 56dB results in an output noise voltage of 0·9mV r.m.s. If the input noise is significantly bigger than this then check that the gain is actually 56dB. Wrong resistor values may give you a high preamplifier gain, and hence more apparent noise. Also IC1 may be more noisy than other devices. It is not uncommon to select the input op amp for low noise operation. If noise is a problem then check for dry joints or other microphonic faults. When using the MIC input at full gain you will hear the preamplifier noise, this is not a fault. The important parameter in all audio equipment is the signal to noise ratio and not the absolute noise level. If the microphone input signal level is 1·46mV r.m.s. then the signal to noise ratio will be 60dB, which is not very much worse than most semi-professional tape recorders. A microphone signal level of 1·46mV r.m.s. is quite a small signal level. In cases like this the best advice is to move the microphone nearer to the object being recorded!

TONE CONTROLS

The tone controls can be tested either with test equipment or by listening to pre-recorded music through them. Inject a sinewave source into the LINE input, set S2 to EQ and monitor the signal at the SEND jack. The frequency response can be plotted out by varying the sinewave frequency and recording the gain changes. These responses should conform to those shown in Fig. 2 last month. However, no one would want laboriously to plot the frequency responses of 18 tone control units using this method! The best method for determining a circuit's frequency response is to inject a swept sinewave and to monitor the output waveform on an oscilloscope. However, if you do not have access to this equipment, then a listening test is quite adequate. Note that the TONE CONTROL section actually provides gain and so it is possible to amplify the system noise. If any of the controls

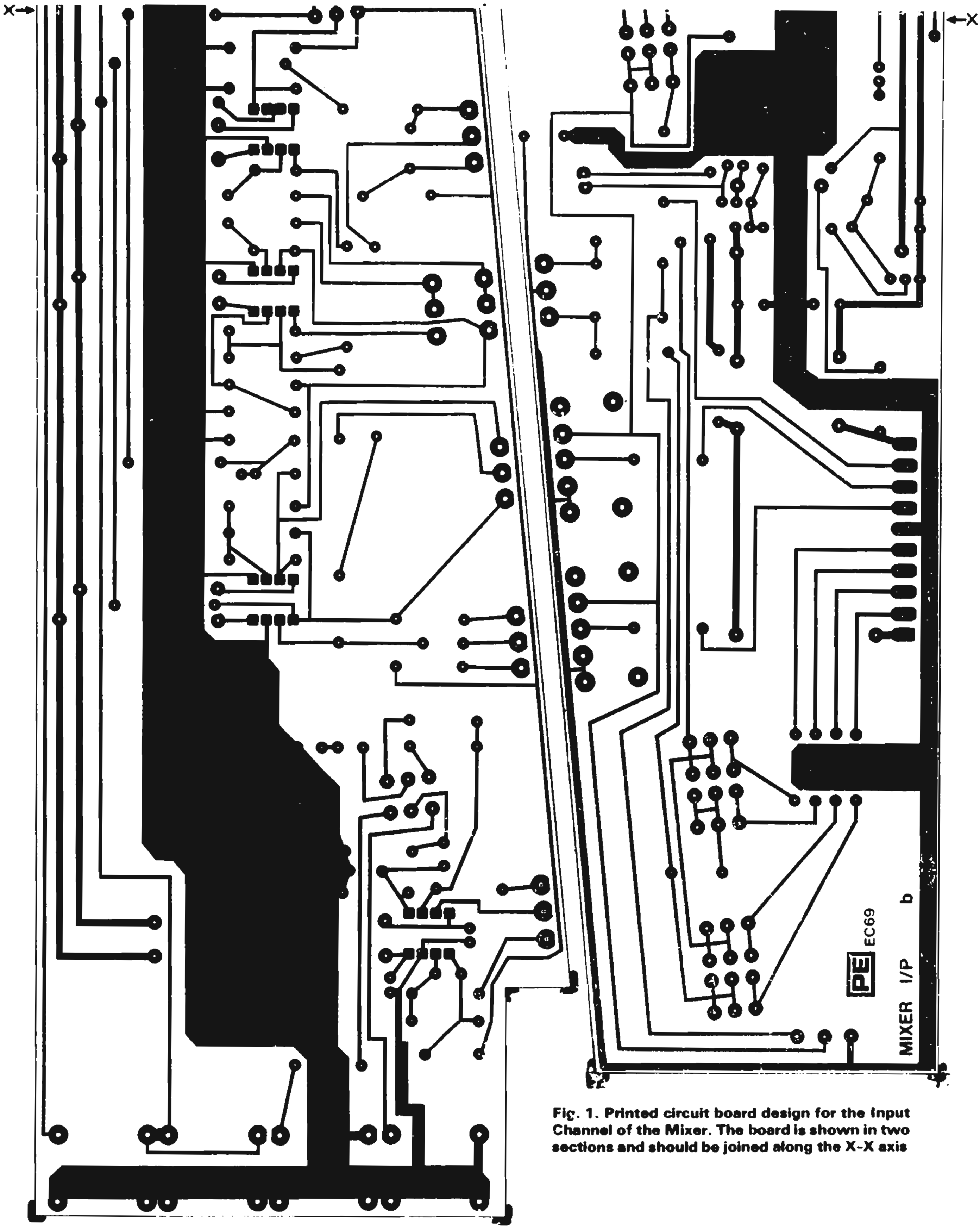


Fig. 1. Printed circuit board design for the Input Channel of the Mixer. The board is shown in two sections and should be joined along the X-X axis

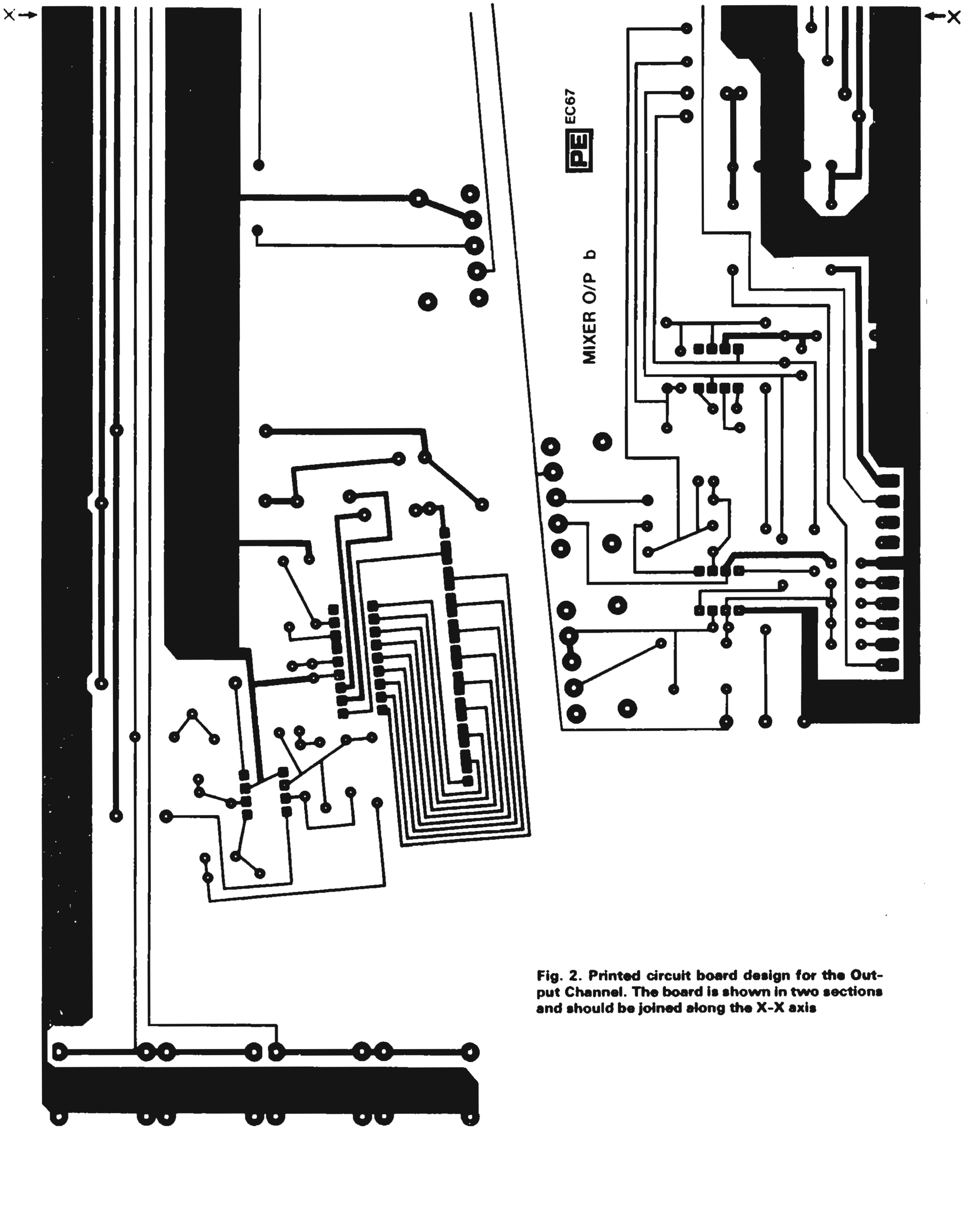


Fig. 2. Printed circuit board design for the Output Channel. The board is shown in two sections and should be joined along the X-X axis

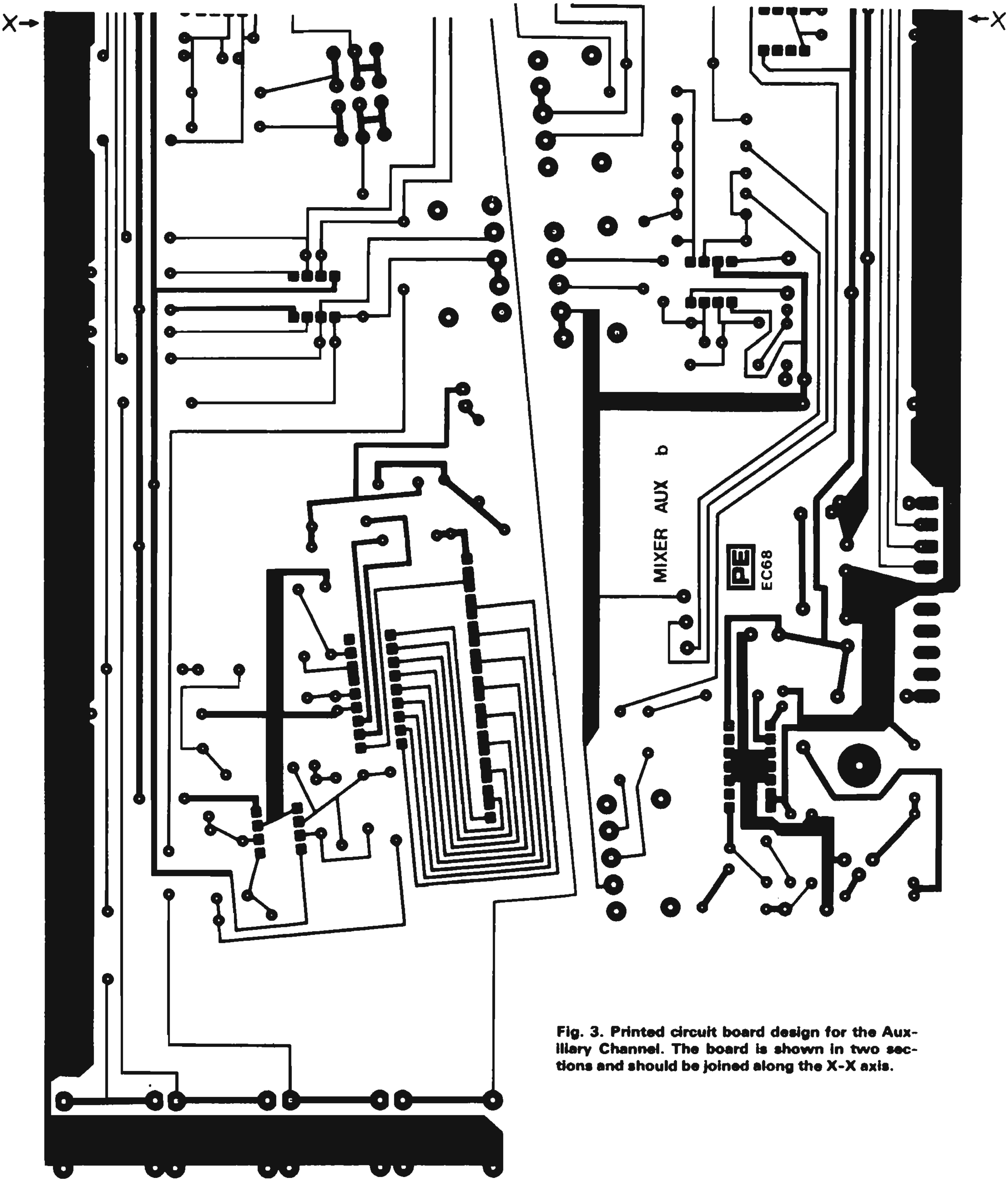


Fig. 3. Printed circuit board design for the Auxiliary Channel. The board is shown in two sections and should be joined along the X-X axis.

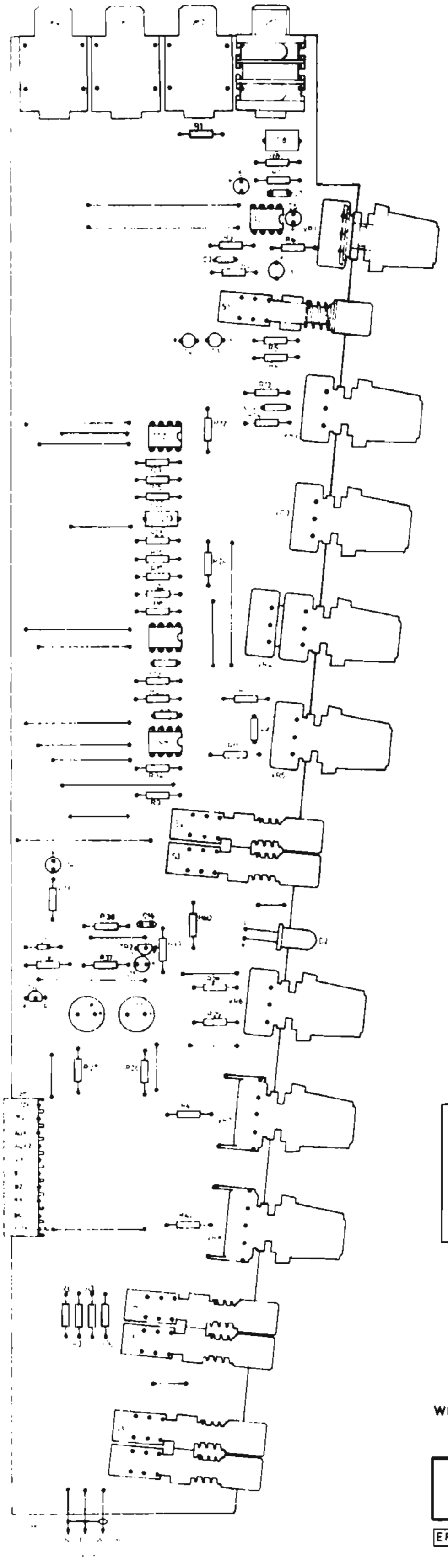


Fig. 4. Component layout for the Input Channel

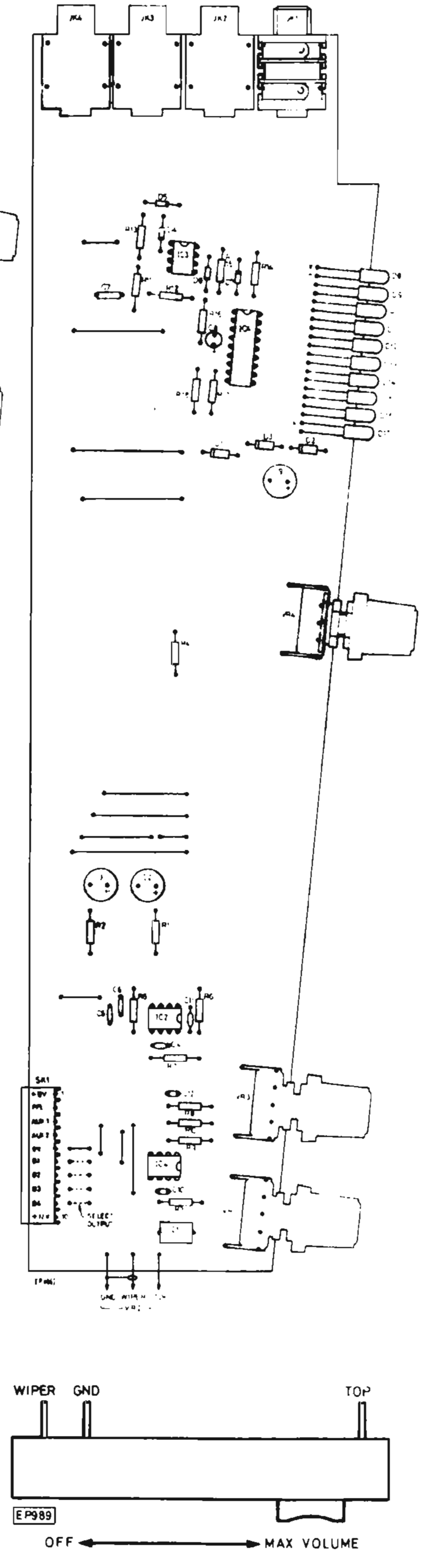


Fig. 5. Component layout for the Output Channel

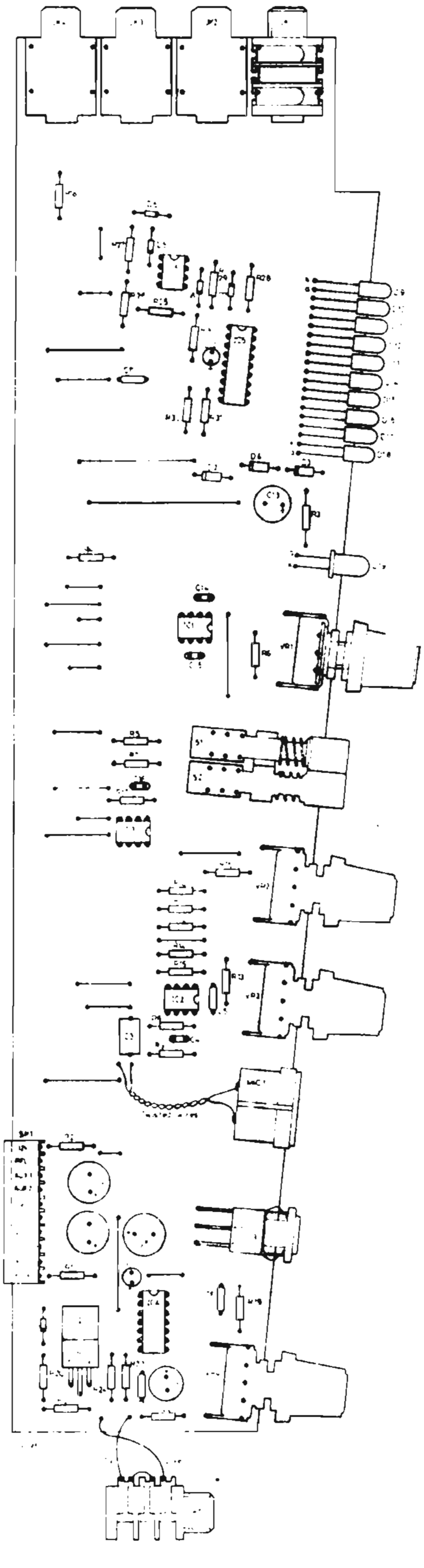


Fig. 6. Component layout for the Auxiliary Channel

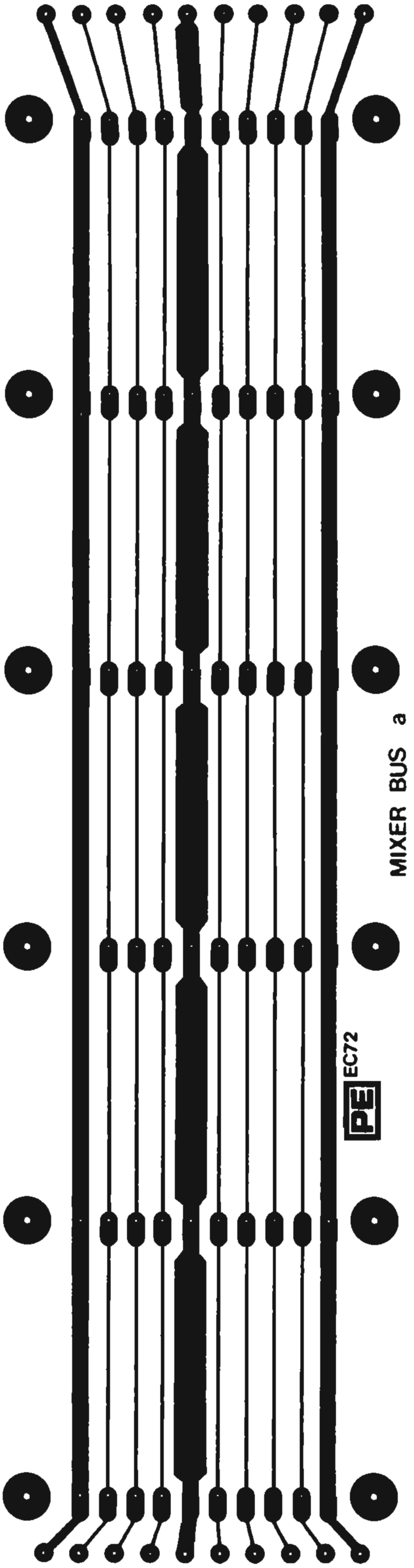


Fig. 7. Bus bar p.c.b. design

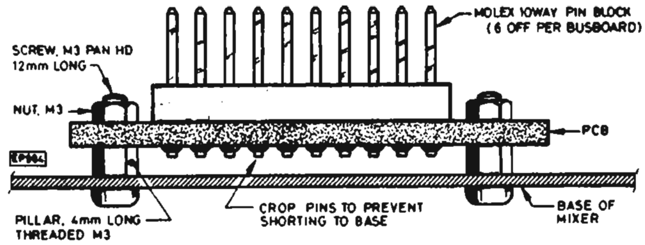


Fig. 8. Mounting details for the bus blocks

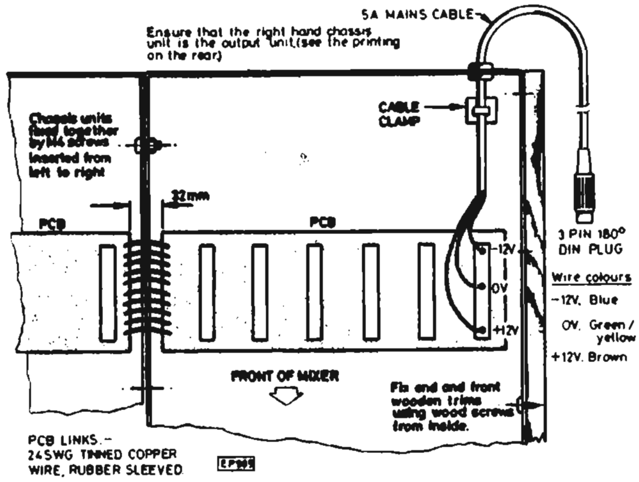
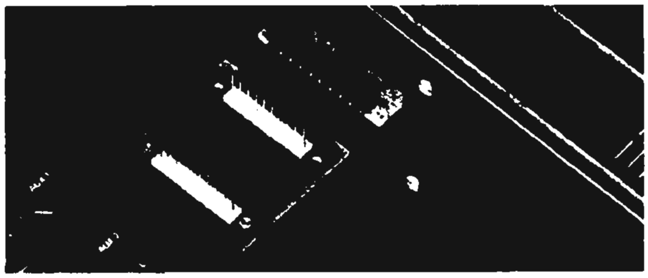


Fig. 9. Wiring and layout for the Bus boards



Bus board link wiring

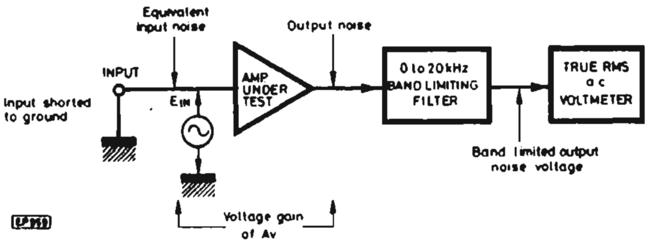
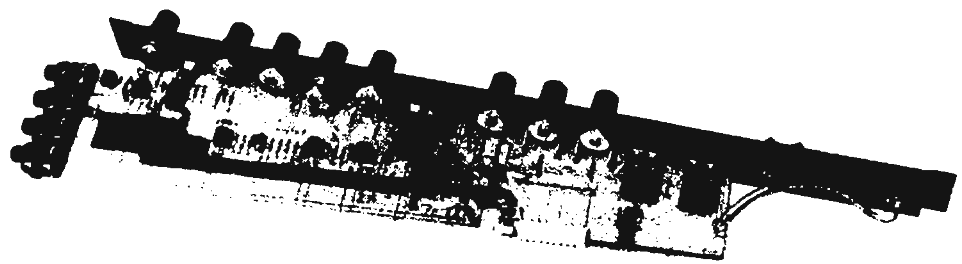


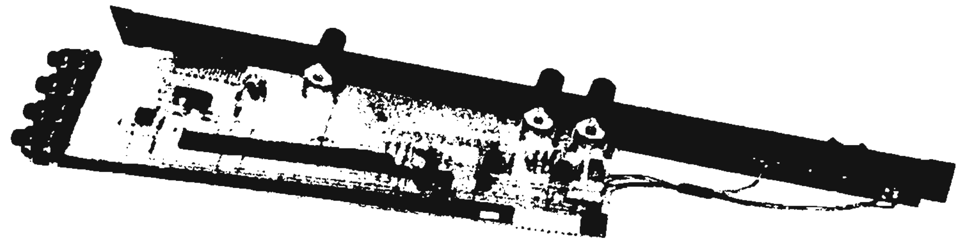
Fig. 10. Arrangement for measuring the equivalent input noise

operate at the wrong frequency or produce a wrong gain change then check the circuit for correct component values. All the pots and switches in the input channel should be almost noiseless and clickless when operated. If this is not the case then check the circuit for correct components or mechanically faulty pots.

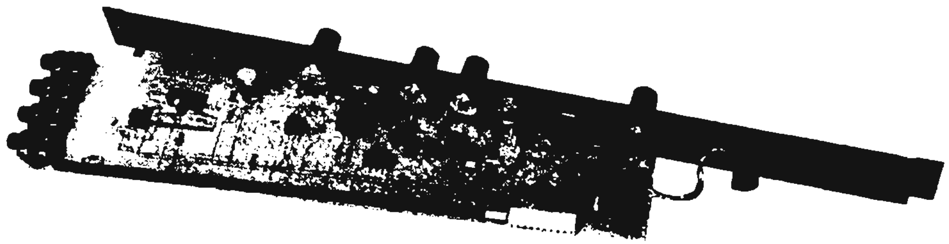
The input channels consume about 15mA from each rail rising to nearly 30mA when the LEVEL l.e.d. comes on. All the op amps should have very little d.c. offset on their outputs. Typically the offset will be ±10mV. Larger offsets may well cause crackle when control pots are rotated. The two integrators in the MIDDLE tone control may have larger offsets but this will not degrade the performance.



Input Channel with the front panel layout shown in (a)



Output Channel with the front panel layout shown in (b)



Auxiliary Channel with the front panel layout shown in (c)

NEXT MONTH: P.S.U. construction and using the Mixing Desk.

(a) (b) (c)

SEMI-PROFESSIONAL
MIXING DESK
Part Three

THE p.c.b. design for the power supply unit is shown in Fig. 1 with the component layout shown in Fig. 2. Before the components are soldered, the 'L' shaped heatsink should first be fitted to the p.c.b.

All the components should then be soldered into position with mica washers being used on the two i.c.s. Carefully check the orientation of the semiconductors and the electrolytic capacitors.

The case components should be mounted as shown in Fig. 3 and all the mains wiring covered with rubber sleeving. The two output rails should be tested before the power supply is connected to the mixing desk.

PA MIXING—LIVE CONCERTS

Mixing the sound for a live music show differs considerably from mixing in a recording studio. Firstly everyone plays at the same time which means that there are many more mixer channels to be balanced and equalised etc. Secondly the high sound level on stage makes it difficult for the players to hear each other with any degree of sound balance. Fig. 4 shows how all the instrument amplifiers, drums and singers are set-up and fed to the mixer which is normally positioned in the audience where the sound engineer can hear clearly the sound of the PA system. It also shows how a separate sound system is provided for the players on stage, so that they can all hear what each person is playing. This other mix is derived from an auxiliary mixer output which is independent from the main stereo PA mix. The mixer also adds effects to the overall sound, such as reverb and echo, using the other auxiliary outputs as well as taping the stereo mix for the inevitable post-mortem of the concert. Live sound mixing is certainly more liable to problems than studio recording, the most common one being feedback or howl-round between the microphones and the on-stage monitoring systems which are placed close to the players. Highly directional microphones are favoured for this reason. Also, because there is never any second chance with live sound it is essential to set up the system and balance the sound before the audience arrives, so that everything is ready when the players walk on to give their performance.

USING THE MIXING CONSOLE

The mixing desk is a modular system built up from sets of 6 input channel modules, individual output channels and the Auxiliary PFL/Headphone output channel. Its main applications lie in the small four-track recording studio where cassette systems are now available for around £600, and live music mixing with PA systems. However, it is flexible enough to be considered for almost any sound mixing requirement. A simplified description follows to illustrate how the mixer is used in the multitrack studio for recording and mixing down as well as in the live concert situation.

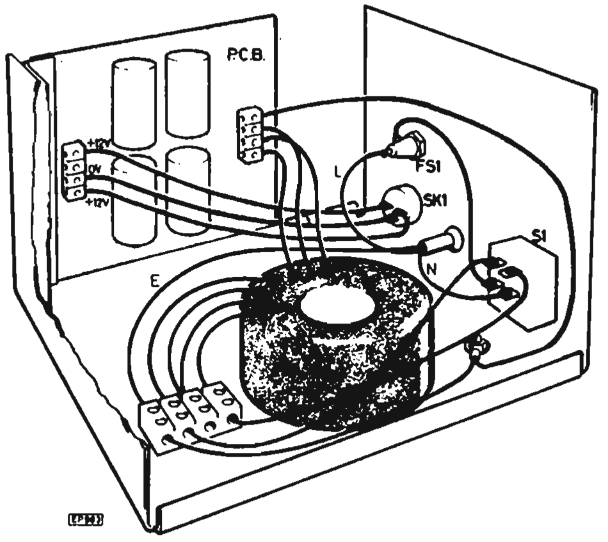


Fig. 3. Wiring diagram for the p.s.u.

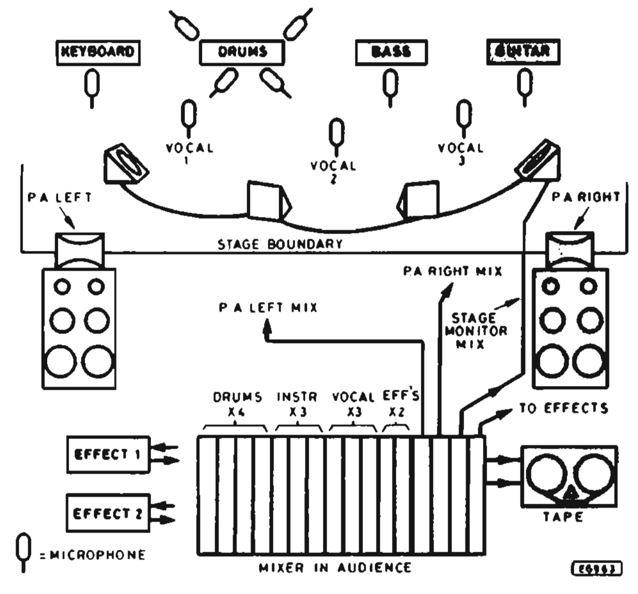


Fig. 4. Live concert mixing

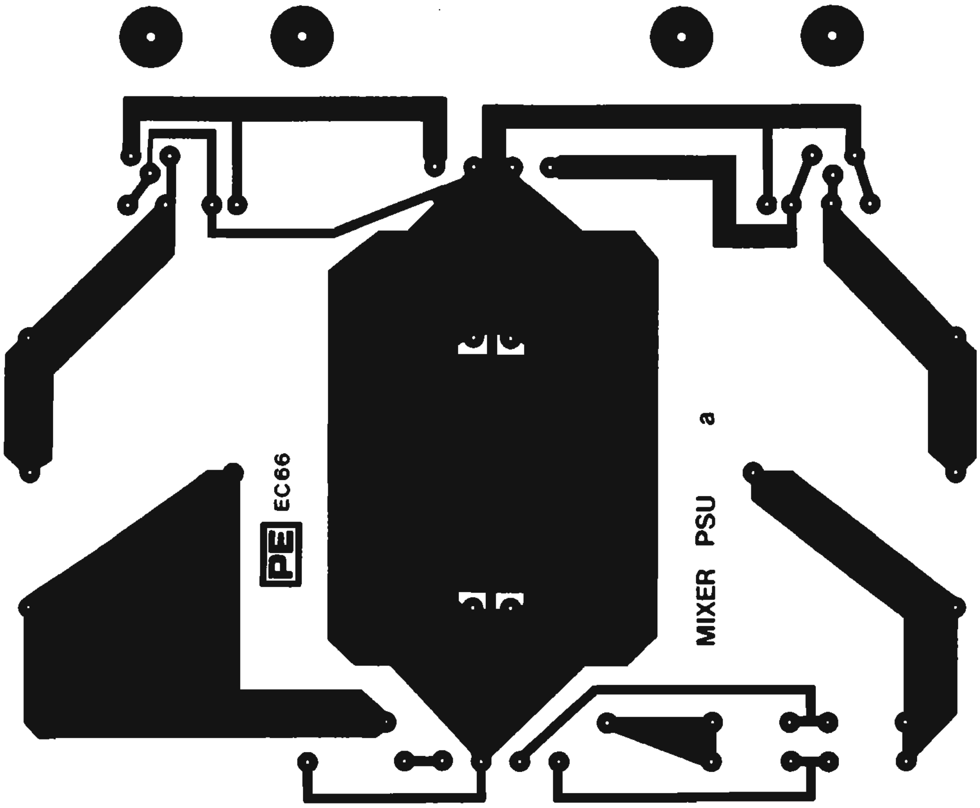


Fig. 1. P.c.b. design for the p.s.u.

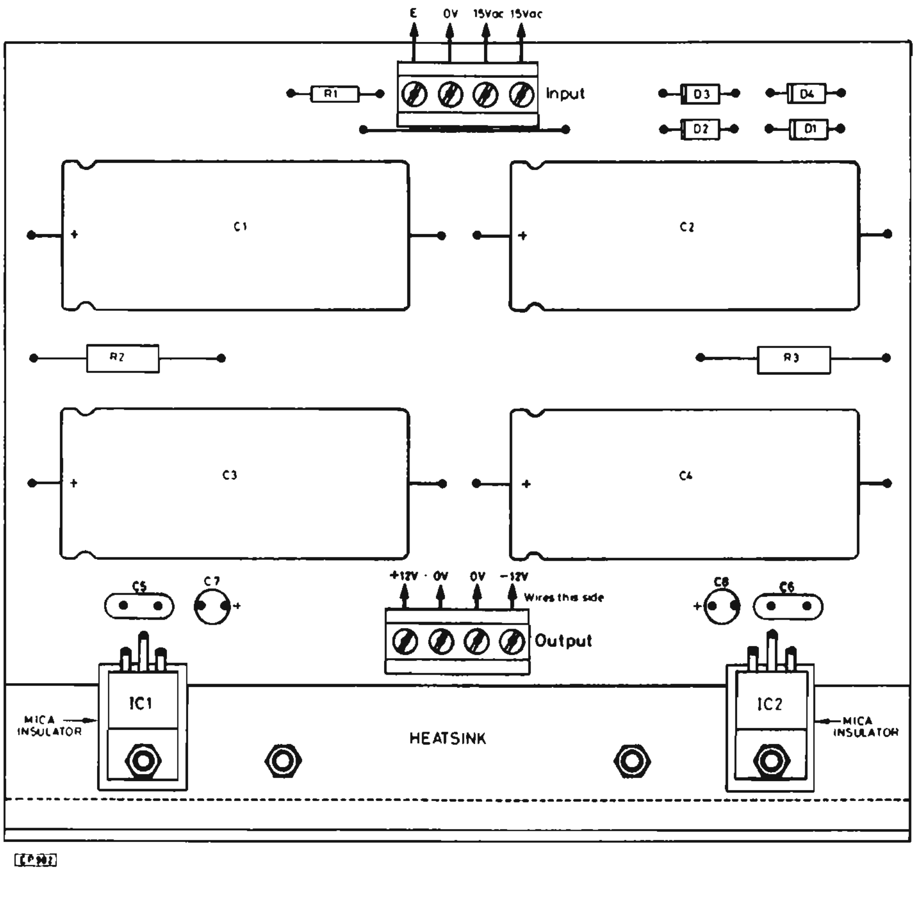


Fig. 2. Component layout

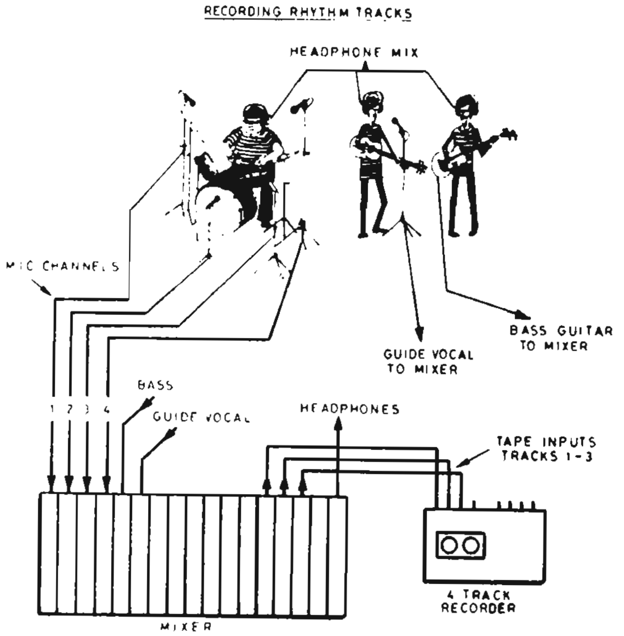


Fig. 5. Recording rhythm tracks

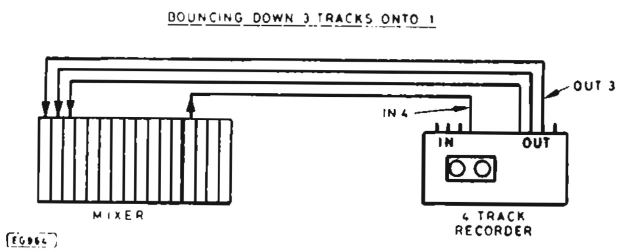


Fig. 6. Bouncing down 3 tracks onto 1

THE MULTITRACK STUDIO

Recording

The first stage of most music recording involves laying down on tape the basic rhythm structure. This usually means drums and bass guitar, along with a rough 'guide' track of vocals to help the rhythm players through the song's arrangement. Fig. 5 shows 4 microphones set up around the drum kit, the bass guitar and the vocal microphone which have all been connected to the mixer input channels. Listening on headphones the recording engineer uses the Pre Fade Listen (PFL) function to examine the sound coming in on each mixer channel, to check that all is well and to add equalisation to any sounds that need it. The drum microphones are all routed to one mixer output and recorded onto one tape track, while the bass and vocals are separately recorded on two other tracks. During this process the players listen to each other using headphones, driven from one of the auxiliary outputs of the mixer which allows them a balanced mix of what is being recorded onto tape. Next comes the addition of the other instrument sounds like guitars, keyboards and vocals, but because of the limited number of tracks available on tape a process known as 'bouncing-down' is used to conserve tape track space. Fig. 6 shows how the mixer may be used to mix three tracks together onto the one remaining track, which then makes room for three new tracks. In this way the recording is built up layer by layer until all the necessary ingredients of the sound are on tape and ready for the final 'mix-down'.

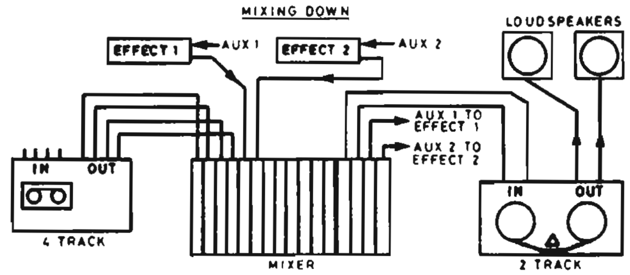


Fig. 7. Mixing down

Mixing Down

In this final stage of the recording process the four tracks of sound are reduced to a two-channel stereo format which necessitates the use of a second tape recorder. Fig. 7 shows how the mixer is used to transfer the sound across from one recorder to the other, and in doing so it allows a number of useful enhancements to be made for the final result. Firstly there is the question of attaining the right balance between the four tracks and the positioning of each track within the stereo sound image. This is known as panning. Final alterations are also made to the equalisation of each track to highlight the overall tone quality, and there is a further opportunity to add special effects such as reverberation, echo, chorus and phasing etc. The auxiliary mixer outputs are used to send any mixture of the sounds to these effects, whos outputs are then brought back on the remaining mixer inpu channels. Finally the end result is reached with a stereo tape that may be played on any domestic sound system. Improvements in technique come with practice, and care is taken to ensure that maximum level goes onto tape wherever possible.